NEAR EAST UNIVERSITY

Faculty of Engineering

Department of Electrical and Electronic Engineering

DONALD DUCKVOICE GENERATOR

Graduatlon Project
EE 400

Student: Omran Hijazi (20033273)

Supervisor: Assoc. Prof. Dr. Adnan Khashman

Nicosia - 2007

ACKNOWLEDGEMENT

First of all, I would like to thank God for guiding me through my study period in Cyprus.

Moreover I want to pay special regards to my parents and my big brother Salim Hijazi who help me to cornplete the electronic devices which I used in my project and supporting me in ali events. They also encouraged me in crises. I shall never forget their sacrifices for my education so I can start my successful life and enjoy it as they are expecting. I anı nothing without theirprayers.

I feel proud to pay my special regards to my project advisor "Assoc. Prof. Dr. Adnan Khashman". He did not disappoint me in any affair. He gave me too much information and did his best of efforts to make me able to complete my project. He has special respect and Devine place in my heart. I anı really thankful to my teacher.

I want to honor ali those persons who have supported and helped me in my project and send to them the best of regards and acknowledgement. Also my special thanks to my all friends who gave me their precious time to complete my project. Also my especial thanks go to Nargiz Abdulina and her family who was supporting me in ali events.

At the end I anı again thankful to all persons who helped and encouraged me to complete my project.

ABSTRACT

üne of the most popular of the Disney cartoon characters, Donald Duck made his debut in the Silly Symphony cartoon "The Wise Little Hen" on June 9, 1934. His fiery temper endeared him to audiences. Donald Duck is an animated cartoon and comic-book character from Walt Disney Productions.

This project will describe how a humarı voice can be converted into Donald duck voice. The operation of the voice frequency signal and it's changes within the electronic devices that are used in the Donald diıck circuit, will also be described.

The problem encountered during the building of this project and their solution will be discussed in this project.

INTRODUCTION

Donald's famous voice, one of the most identifiable voices in ali of animation, was until 1985 performed by voice actor Clarence "Ducky" Naslı. it was Jargely this semi-intelligible speech that would cement Donald's image into audiences' minds and help fuel both Donald's and Nash's rise to stardom. Since 1985, Donald has been voiced by Tony Anselmo, who was trained by Naslı himself for the role[1].

The aim of this project is to design, build and test a Donald duck voice generator. The voice frequency signal, within part of the audio range, that is used for the transmission of speech makes the speaker sound like "Donald Duck".

First chapter of the project present the electroriic components especially the components were used in this project such as resistor, capacitor, diodes, integrated circuits ICs switches, microphones and Ioad speakers. Safety guideline also showed the ways that leads how to use the components in correct way, because if it done in wrong way it will bum or break the components. So that before doing any electrical project this chapter should be taken care.

Second chapter of the project is about the voice frequency. it presents the voice inversion, the transmission signal, where it's used, and the voice frequency types and their operation. Also the method of voice identification and measurements that is used for transmission of speech.

Third chapter of the project is most important one, which explains the hardware project in details, how to built it, how it work, what is the input, voice conversion and output of it. The circuit diagrams, integrated circuit types, their applications, and the schematic diagram of each one will be shown. Forth chapter will show us the solutions of the problems that we had, during the time that we spend to connect the Donald duck circuit.

CHAPTERONE

ELECTRONIC COMPONENTS

1.1 Overview

This chapter presents an introduction to electronic components that are commonly used in hardware projects. Safety guidelines for electronic projects will also be described.

1.2 Introduction to Electronic Components

Electricity is the flow of electrical energy through some conductive material; electronics refers to using changing electrical properties to convey information. Electronic sensors convert some other form of energy (]ight, heat, sound pressure, ete.) into electrical energy.

The main components used in electronics are of two general types: passive (e.g. resistors and capaeitors) and active (e.g. transistors and integrated circuits). The main difference between active and passive components is that active ones require to be powered in some way to make them work. Active components can also be used to amplify signals.

1.3 Resistors

Most of the resistance in circuits is found in components that do specific work, such as bulbs or heating elements, and in devises called resistors. Resistors are devises that provide precise amounts of opposition or resistance to current flow. Resistors are very common in electric circuits. They are used to provide specific resistivity to limit current and to control voltage in a circuit.

In general, a resistor is used to create a known voltage-to-current ratio in an electric circuit. If the current in a circuit is known, then aresistor can be used to create a known potential difference proportional to that current. Conversely, if the potential difference between two points in a circuit is known, a resistor can be used to create a known current proportional to that difference.

1.3.1 The ideal Resistor

The SI unit of electrical resistance is the ohm (.O). *A* component hasa resistance of 1 n ifa voltage of 1 volt across the component results in a current of 1 ampere, or amp, which is equivalent to a flow of one coulomb of electrical charge (appföxifuately 6.241506 x 10^{18} electrons) per second. The multiples kiloohm (1 k.Q = 1000 .O) and megaohm (1 M.O = 10^6 .O) are also commonly used[2].

In an ideal resistor, the resistance remains constant regardless of the applied voltage current through the device or the rate of change of the current. While real resistors cannot attain this goal, they are designed to present little variation in electrical resistance when subjected to these changes, or to changing temperature and other environmental factors.

1.3.2 Nonideal Characteristics

A resistor has a maximum working voltage and current above which the resistance may change (drastically, in some cases) or the resistor may be physically damaged (overheat or bum up, for instance). Although some resistors have specified voltage and current ratings, most are rated with a maximum power which is determined by the physical size. Common power ratings for carbon composition and metal-film resistors are 1/8 watt, ~ watt, and Yz watt. Metal-film and carbon film resistors are more stable than carbon resistors against temperature changes and age. Larger resistors are able to dissipate more heat because of their larger surface area. Wire-wound and resistors embedded in sand (ceramic) are used when a high power rating is required.

Furthermore, all real resistors also introduce some inductance and a small amount of capacitance, which change the dynamic behavior of the resistor from the ideal.

1.3.3 Types of Resistors

Resistors come in a variety of values and types. Themost common type is the fixed resistor. Fixed resistors have single values of resistance, which remain constant. There are also variable resistors that can be adjusted to vary or change the amount of resistance in a circuit.

The value of resistance of resistors is given in ohms. Resistors can have values from less than one ohm up to many millions of ohms.

1. Fixed Resistors: The most common fixed resistor is the composition type. The resistance element is made of graphite, or some other form of carbon, and alloy materials. These resistors generally have resistance values that range from 0.1 Ω to 22 M Ω.

Another kind of fixed resistor is the wire wound type. The resistance element is usually made of nickel-chromium wire wound on a ceramic rod. These resistors generally have resistance values that range from 1 Ω to 100 k Ω

2. Variab]e Resistors:

Variable resistors are used to adjust the amount of resistance in a circuit. A variable resistor consists of a sliding contact arın that makes contact with a stationary resistance element. As the sliding arın moves across the element, its point of contact on the element changes, effectively changing the length of the element. The rating ofa variable resistor is its resistance at its highest setting.

Variable resistors are also called rheostats or potentiometers. The resistance elements of rheostats are usually wire wound. They are most often used to control very high currents, such as in motors and lamps. Potentiometers generally have composition elements. They are used as control devices in radios, amplifiers, televisions, and electrical instruments.

1.3.4 Rating Tolerances

The actual resistance ofa.resistor may be.greate:r of1ess than its indicated rating. The possible range of variance from .the indicatedirating is called its tolerance, Common tolerances for composition resistors are ±5, ±10, and ± 20 percent. Wire wound resistors usually have a tolerance of ±5 percent.

1.3.5 Resistor Rating Color Code

Composition resistors are color cöded to indicate resistance values or ratings. The color code consists of various color harids that indicate the resistance values of resistors in ohms as well as the tolerance rating. The resistor rating color code Table. l below is used to identify the resistance rating of resistors.

Table 1.1: Resistor Rating Color Code Table

Color	1st Band	2nd Band	3rd Band	4th Band
Black	0	0	1	1
Brown	1	1	10	
Red	2	2	100	
Orange	3	3	1,000	
Yellow	4	4	10,000	
Green	5	5	100,000	
Blue	6	6	1,000,000	
Violet	7	7	10,000,000	
Gray	8	8	100,000,000	
White	9	9	1,000,000,000	
Gold			0.1	5%
Silver			0.01	10%
None				20%

Composition resistors generally have four color bands. The color code is read as follows:

First, look up the number values of the first two bands on the table and combine the two numbers.

Then multiply this two digit number by the value of the 3rd band, the multiplier band.

The resulting number is the resistance value of the resistor in ohms.

The fourth band is the tolerance band. If the 4th band is gold, the resistor is guaranteed to be within 5% of the rated value. If the 4th band is silver, it is guaranteed to be within

10%. If there is no 4th band, the resistor is guaranteed to be within 20% of the rated value.

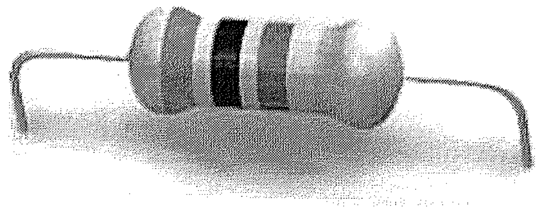

Figure 1.1: Resistor Rating Colo.r Çode

For example, the color code ofthe above resistor in Figllfel.lisfead as follows:

... The 1st band is brown. The first band is always the band closest.to thesenduof the resistor. From the table you can see that the number value of brown in the 1st band column is 1.

- The 2nd band is black. The number value ofblack in the 2nd band column is 0.

... Combining the two numbers gives you 10.

... The 3rd band is red. This is the multiplier band. The multiplier value ofred is 100.

- Multiplying the combined digit of 10 by the multiplier gives us 1,000.

Therefore, the above resister is rated at 1,000 ohms, which can be written as 1 k n.
4th, or tolerance, band of the resister is silver. Therefore, the resistor is guaranteed to have a resistance value within 10% of Ik .Q

1.3.6 Resistors in Series Circuits

A series circuit is a circuit in which the current has only one patlı. In a series circuit, all of the current passes through each of the components in the circuit. The equivalent resistance ofa series circuit is the sum of all the resistances in the circuit. Therefore, to calculate the total resistance ofa series circuit, use the following formula:

$$R_T = R_1 + R_2 + R_3 \ldots$$

Where RT is the total Of equivalent resistance in the circuit, and R_1 through R_3 ··· are the resistance ratings of the individual resistors Of components in the circuit,

1.3. Resistors in Parallel Circuits

A parallel circuit is a circuit in which components are arranged so that the patlı for the current is divided.

Placing the resistors in parallel always decreases the total Of equivalent resistance of the circuit. This is true because connecting resistors in pafallelis equivalent to placing them side by side, increasing the total area available for the 'flôw <of current and thereby,reducing resistance. To calculate the total resistance ofa parallel circuit, use the following formula:

$$RT = 1 \div (1/R_1 + 1/R_I + 1/R3 \ldots)$$

Where RT is the total resistance in the circuit, and R_1 through R_3 · · · are the resistance ratings of the individual resistors Of components in the circuit.

1.4 Capacitors

Capacitor - A device that stores electrical charges and can be used to maintain voltage levels in power lines and improve electrical-system efficiency. and consisting of two Of more conducting plates separated from one another by a dielectric nonconductor, such as glass, mica, plastic, Of dry air, used for storing an electric charge; condenser.

In Si units, a capacitor has a capacitance of one farad when one coulomb of charge causes a potential difference of one volt across the plates. Since the farad is a vefy large unit, values of capacitors are usually expressed in microfarads (μF), nanofarads (nF), Of picofarads (pF).

1.4.1 Capacitors in Series and Parallel

Combined capacitance of capacitors connected in series:

$$\frac{1}{C}=\frac{1}{C_1}+\frac{1}{C_2}+\frac{1}{C3}+$$

Combined capacitance of capacitors connecıed.in'parallel:

$$C=C_1+C_2+C_3+......$$

Two or more capacitors are rarely deliberately connected iriseries in real circuits, but it can be useful to connect capacitors in parallel to obtain a very large capacitance, for examp'^ to smooth a power supply.

1.4.2 Capacitor Hazards and Safety

Capacitors may retain a charge long after power is removed from a circuit; this charge can cause shocks (sometimes fatal) or damage to connected equipment. For example, even a seemingly innocuous device such as a disposable camera flash unit powered by a 1.5 volt AA battery contains a capacitor which may be charged to over 300 volts. This is easily capable of delivering an extremely painful, and possibly lethal shock.

Care must be taken to ensure that any large or high-voltage capacitor is properly discharged before servicing the containing equipment. For safety purposes, all large capacitors should be discharged before handling. For board-level capacitors, this is done by placing a bleeder resistor across the terminals, whose resistance is large enough that the leakage current will not affect the circuit, but sınalfenôugh to discharge the capacitor shortly after power is removed. High-voltage capac:itôts should be stored with the terminals shorted, since temporarily discharged capacitors can develop potentially dangerous voltages when the terminals are left open-circuited.

1.4.3 Capacitor Number Code

A number code is often used on small capacitors where printing is difficult:

- The 1st number is the 1st digit,
- The 2nd number is the 2nd digit,
- The 3rd number is the number of zeros to give the capacitance in pF.
- Ignore any letters - they just indicate tolerance and voltage rating.

For example: 102 means 1000pF = 1nF (not 102pF!)

For example: 472J means 4700pF= 4.7nF (J means 5% tolerance).

1.4.4 Capacitor Color Code

A colour code was used on polyester capacitors for many years. It is now obsolete, but of course there are many still around. The colours should be read like the resistor code, the top three colour bands giving the value in pF. Ignore the 4th band (tolerance) and 5th band (voltage rating) [3].

For example: Brown, black, orange means 10000pF = 10nF = 0.01μF.

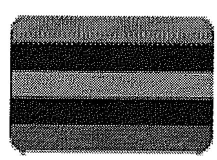

Figure 1.2: capacitor color code

Table 1.2: Capacitor color code

Colour Code	
Colour	Number
Black	0
Brown	1
Red	2
Orange	3
Yellow	4
Green	5
Blue	6
Violet	7
Grey	8
White	9

1.4.5 Electrolytic Capacitors

Electrolytic capacitors are polarised and they must be connected the correct way round; at least one of their leads will be marked + or -. They are not damaged by heat when soldering.

There are two designs of electrolytic capacitors; axial where the leads are attached to each end (220µF in picture) and radial where both leads are at the same end (10µF in picture). Radial capacitors tend to be a little smaller and they stand upright on the circuit board.

It is easy to find the value of electrolytic capacitors because they are clearly printed with their capacitance and voltage rating. The voltage rating can be quite low (6V for example) and it should always be checked when selecting an electrolytic capacitor. It the nrrner-t parts list does not specify a voltage, choose a capacitor with a rating which is greater than the project's power supply voltage. 25V is a sensible minimum for most battery circuits.

1.4. Tantalum Bead Capacitors

Tantalum bead capacitors are polarised and have 16w voltage ratings like electrolytic capacitors. They are expensive but very small, so .they ate)üsed where a large capacitance is needed in a small size.

Modem tantalum bead capacitors are printed with their capacitance and voltage in full. However older ones use a colour-code system which has -two' stripes (for the two digits) and a spot of colour for the number of zeros to give the value in µF. The standard colour code is used, but for the spot, grey is used to mean x 0.0l and white means x 0.1 so that values of less than 1OµF can be shown. A third colour stripe near the leads shows the voltage (yellow 6.3V, black 1OV, green 16V, blue 20V, grey 25V, white 30V, pink 35V).

For example: blue, grey, black spot means 68µF
For example: blue, grey, white spot means 6.8µF
For example: blue, grey, grey spot means 0.68µF

1.5 Diodes

In electronics, a diode is a component that restricts the direction of movement of charge carriers. Essentially, it allows an electric current to flow in one direction, but blocks it in the opposite direction. Thus, the diode can be thought of as an electronic version of a check valve. Circuits that require current flow in only one direction will typically include one or more diodes in the circuit design.

Early diodes included "cat's whisker" crystals and vacuum tube devices (called thermionic valves in British English). Today the most cornmon diodes are rnade from semicondııctor materials such as silicon Of germaniurn.

Diodes allow electricity to flow in only one direct:ion. The arrow of the circuit syrnbol shows the direction in which the currerit cari flow. Diodes are the electrical version ofa valve and early diodes were actually called valves,

1.5.1 Forward Voltage Drop

Electricity uses up a little energy pushing its way through the diode, rather like a person pushing through a door with a spring. This means that thefe is a small voltage across a conducting diode. It is called the 'forward voltage drop' and is about 0.7V for all normal diodes which are made from silicon. The forward voltage drop ofa diode is almost constant whatever the current passing through the diode so they have a very steep characteristic (current-voltage graph).

1.5.2 Reverse Voltage

When a reverse voltage is applied a perfect diode does not conduct, but all real diodes leak a very tiny current ofa few μA Of less. This can be ignored in most circuits because it will be very much smaller than the current flowing in the forward direction. However, all diodes have a maximum reverse voltage (usually 50V Of more) and if this is exceeded the diode will fail and pass a large current in the reverse direction; this is called 'breakdown'.

Ordinary diodes can be split into two types:

Signal Diodes can pass low currents of l OOrnA Of less.

Rectifier Diodes can pass high currents.

1.5.3 Connecting and Soldering Diodes

Diodes must be connected the correct way round, and circuit diagrams may be labeled or '+' for anode and 'k' or '-' for cathode. The cathode is marked by aline painted on the body of the diode. Diodes are labeled with their côde iri small print, and you may need a magnifying glass to read this on small signal diodes. Srnall signal diodes can be damaged by heat when soldering, but the risk is small unless you iare using a germaniurn diode (codes beginning OA...) in which case you rnust place :a hea{.sink, such as a crocodile clip, on the lead being soldered. Rectifier diodes are qüite tôbust .and no special precautions are needed for soldering them [4].

1.5.4 Rectifier Diodes (high current)

Rectifier diodes are used in power supplies to convert altemating current (AC) to direct current (DC), a process called 'rectification'. They are also used elsewhere in circuits where a large current must pass through the diode.

All rectifier diodes are made from silicon and therefore have a forward voltage drop of 0.7V. The table shows maximum current and maximum reverse voltage for sorne popular rectifier diodes.

1.5.5 Bridge Rectifiers

There are several ways of connecting diodes to make a rectifier to convert AC to DC. The bridge rectifier is one of them and it is available in special packages containing the four diodes required. Bridge rectifiers are rated by their maximum current and maximurn reverse voltage. They have four leads or terminals: the two DC outputs are labelled '+' and '-', the two AC inputs are labeled '-'.

Here In this project I used only the 1N4148 switching diode. This means that it can quiekly switch between the conducting states to the inhibiting state. for the 1N4148 diode, the time it takes to switch between these two states is 4ns (which is pretty fast).

Figure: 1.3: Switch.ing diode

1.5.6 General description

The 1N4148 and 1N4448 are high-speed switching diodes fabricated in planar technology, and encapsulated in hermetically sealed leaded glass SOD27 (D0-35) packages.

1.5.7 Features

1. Hermetically sealed leaded glass SOD27 (D0-35)package

2. High switching speed: max. 4 ns

3. General application

4. Continuous reverse voltage: max. 100 V

5. Repetitive peak reverse voltage: max. 100 V

6. Repetitive peak forward current: max. 450 mA.

1.5.8 Signal Diodes (low current)

Signal diodes are used to process information (electrical signals) in circuits, so they are only required to pass small currents of up to 100mA. General purpose signal diodes such as the 1N4148 are made from silicon and have a forward voltage drop of 0.7V.

Gennanium diodes such as the OA90 have a lower forward voltage drop of 0.2V and this makes them suitable to use in radio circuits as detectors which extract the audio signal from the weak radio signal.

For general use, where the size of the forward voltage drop is less important, silicon diodes are better because they are less easily damaged by heat when soldering, they have a lower resistance when conducting, and they have very low leakage currents when a reverse voltage is applied.

1.6 Integrated Circuit

A monolithic integrated circuit (also known as IC, microchip, silicon chip, computer chip or chip) is a miniaturized electronic circuit (consisting mainly of semiconductor devices, as well as passive components) which has been manufactured in the surface ofa thin substrate of semiconductor material.

A hybrid integrated circuit is a miniaturized electronic circuit constructed of individual semiconductor devices, as well as passive components, bonded to a substrate or circuit Board.

The integrated circuit was made possible by experimental discoveries which showed that semiconductor devices could perfonn the functions of vacuum tubes and by mid-20th-century technology advancements in semiconductor device fabrication. The integration of large numbers of tiny transistors into a small chip was an enonnous improvement over the manual assembly of vacuum tubes and circuits using discrete electronic components. The integrated circuit's mass production capability, reliability, and ease of adding complexity prompted the use of standardized ICs in place of designs using discrete transistors which quickly pushed vacuum tubes into obsolescence. There are two main advantages of ICs over discrete circuits - cost and perfonnance. The cost is low because the chips, with all their components, are printed as a unit by photolithography and not constructed a transistor at a time. As of 2006, chip areas range from a few square mm to around 250 mm2, with up to 1 million transistors per mm2. Among the most advanced integrated circuits are the microprocessors, which control everything from computers to cellular phones to digital microwave ovens. Digital memory chips are another family of integrated circuit that is crucially important to the modern information society. While the cost of designing and developing a complex integrated circuit is quite high, when spread across typically millions of production units the individual IC cost is minimized. The performance of ICs is high because the small size allows short traces which in turn allows low pôwer logic (such as CMOS) to be used at fast switching speeds[5].

ICs have consistently migrated to smaller feature sizes over the years, allowing more circuitry to be packed on each chip - see Moore's law. As thefeature size shrinks, almost everything improves - the cost per unit and the switching power consumption go down, and the speed goes up. However, IC's with nanometer-scale devices are not without their problems, principal among which is leakage current, although these problems are not insurmountable and will likely be solved or at least ameliorated by the introduction of high-k dielectrics. Since these speed and power consumption gains are apparent to the end user, there is fierce competition among the manufacturers to use finer geometries. This process, and the expected progress over the next few years, is well described by the International Technology Roadmap for Semiconductors, or ITRS.

Digital integrated circuits can contain anything from one to millions of logic gates, flip-flops, multiplexers, and other circuits in a few square millimeters. The small size of these circuits allows high speed, low power dissipation, and reduced manufactiıring cost compared with board-level integration. Analog integrated circuits perform analog functions like amplification, active filtering, demodulation, mixing, ete. ADCs and DACs are the key elements of mixed signal ICs. They convert signals between analog and digital formats. Analog ICs ease the burden on circuit designers by having expertly designed analog circuits available instead of designing a difficult analog circuit from scratch.

1.6.1 Popularity of ICs

Only a half century after their development was initiated, integrated circuits have become ubiquitous. Computers, cellular phones, and other digital appliances are now inextricable parts of the structure of modem societies. That is, modem computing, communications, manufacturing and transport systems, including the Intemet, all depend on the existence of integrated circuits. Indeed, many scholars believe that the digital revolution brought about by integrated circuits was one of the most significant occurrences in the history of mankind.

1.6.2 What Can an IC DO?

In consumer electronics, ICs have made possible the development of many new products, including personal calculators and computers, digital watches, and video games. They have also been used to improve or lower the cost of many existing products, such as appliances, televisions, radios, and high-fidelity equipment.

1.6.3 IC Types

Integrated circuits are often classified by the number of transistors and other electronic components they contain:

- SSI (small-scale integration): Up to 100 electronic components per chip
- MSI (medium-scale integration): From 100 to 3,000 electronic components per chip
- LSI (large-scale integration): From 3,000 to 100,000 electronic components per chip
- VLSI (very large-scale integration): From 100,000 to 1,000,000 electronic components per chip
- ULSI (ultra large-scale integration): More than 1 million electronic components per chip

There are two major kinds of ICs:

- analog (or linear) which are used as amplifiers, timers and oscillators
- digital (or logic) which are used in microprocessors and memories

Some ICs are combinations of both analog and digital

But here I only concemed with two types of integrated circuit, the LF356 - JFET Input Operational Amplifiers and the TL074C Quad Low-Noise J-FET Operational Amplifier. The LF356 has an 8-pin DIL and the TL074C hasa 14-pin DIL.

LF356 this is op amps with JFET input devices. These JFETs have large reverse breakdown voltages from gate to source and drain eliminating the need for clamps across the inputs. Therefore large differential input voltages can easily be accommodated without a large increase in input current. The maximum differential input voltage is independent of the supply voltages. However, neither of the input voltages should be allowed to exceed the negative supply as this will cause large currents to flow which can result in a destroyed unit. Exceeding the negative common-mode limit on either input will force the output to a high state, potentially causing a reversal of phase to the output. Exceeding the negative common-mode limit on both inputs will force the amplifier output to a high state. In neither case does a latch occur since raising the input back within the common-mode range again puts the input stage and thus the amplifier in a normal operating mode.

Exceeding the positive common-mode limit on a single input will not change the phase of the output however, if both inputs exceed the limit, the output of the amplifier will be forced to a high state. This amplifier will operate with the common-mode input voltage equal to the positive supply. In fact, the common-mode voltage can exceed the positive supply by approximately 100 mV independent of supply voltage and over the full operating temperature range. The positive supply can therefore be used as a reference on an input as, for example, in a supply current monitor and/or limiter. Precautions should be taken to ensure that the power supply for the integrated circuit never becomes reversed in polarity or that the unit is not inadvertently installed backwards in a socket as an unlimited current surge through the resulting forward diode within the IC could cause fusing of the intemal conductors and result in a destroyed unit. All of the bias currents in these amplifiers are set by FET current sources. The drain currents for the amplifier are therefore essentially independent of supply voltage. As with most amplifiers, care should be taken with lead dress, component placement and supply decoupling in order to ensure stability. For example, resistors from the output to an input should be placed with the body close to the input to minimize "pickup" and maximize the frequency of the feedback pole by minimizing the capacitance from the input to ground. A feedback pole is created when the feedback around any amplifier is resistive. The parallel resistance and capacitance from the input of the device (usually the inverting input) to ac ground set the frequency of the pole. In many instances the frequency of this pole is much greater than the expected 3 dB frequency of the closed loop gain and consequently there is negligible effect on stability margin. However, if the feedback pole is less than approximately six times the expected 3 dB frequency a lead capacitor should be placed from the output to the input of the op amp. The value of the added capacitor should be such that the RC time constant of this capacitor and the resistance it parallels is greater than or equal to the original feedback pole time constant.

- Advantages

1. Replace expensive hybrid and module FET op amps
2. Rugged JFETs allow blow-out free handling compared with MOSFET input devices
3. Excellent for low noise applications using either high or low source impedance=-very low *l/f* comer
4. Offset adjust does not degrade drift or common-mode rejection as in most monolithic amplifiers
5. New output stage allows use of large capacitive loads (5,000 pF) without stability problems
6. Intemal compensation and large differential input voltage capability

- Applications

1. Precision high speed integrators
2. Fast D/A and A/D converters
3. High impedance buffers
4. Wideband, low noise, low drift amplifiers
5. Logarithmic amplifiers
6. Photocell amplifiers
7. Sample and Hold circuits

-Common Features

1. Low input bias current: 30pA
2. Low lnput Offset Current: 3pA
3. High input impedance: 10120
4. Low input noise current:
5. High common-mode rejection ratio: 100 dB
6. Large de voltage gain: 106 dB

TL074C These low noise JFET input operational amplifiers combine two state-of-the-art analog technologies on a single monolithic integrated circuit. Each intemally compensated operational amplifier has well matched high voltage JFET input device for low input offset voltage. The BIFET technology provides wide bandwidths and fast slew rates with low input bias currents, input offset currents, and supply currents. Moreover, the devices exhibit low noise and low harmonic distortion, making them ideal for use in high fidelity audio amplifier applications.

These devices are available in single, dual and quad operational amplifiers which are pin-compatible with the industry standard MC1741, MC1458.

-Features

1. Low power consumption
2. Wide common-mode and differential voltage range
3. Low input bias and offset currents

4. Low noise en=l8nV/YHz(typ)
5. Output short-circuit protection
6. High input impedance J-FET input stage
7. Low harmonic distortion: O.Ol% (typ)
8. Intemal frequency compensation
9. Latch up free operation
10 High slew rate: 13 V/µs (typ)

1.7 Switches

Electrical switches. Top, left to right: circuit breaker, rnercury switch, wafer switch, DIP switch, surface mount switch, reed switch. Bottom, left to .right: wall switch (U.S. style), miniature toggle switch, in-line switch, push-button switch, rocker switch, microswitch.

A switch is a device for changing the course (or flow) ofa circuit. The prototypical model is a mechanical device (for example a railroad switch) which can be disconnected from one course and connected to another. The term "switch" typically refers to electrical power or electronic telecommunication circuits. In applications where multiple switching options are required (e.g., a telephone service), mechanical switches have long been replaced by electronic variants which can be intelligently controlled and automated.

The switch is referred to as a "gate" when abstracted to mathematical form. In the philosophy of logic, operational arguments are represented as logic gates. The use of electronic *gates* to function as a system of logical gates is the fundaınental basis for the computer-i.e. a computer is a system of electronic switches which function as logical gates.

1.7.1 A Simple Electrical Switch

In the simplest case, a switch has two pieces of metal called contacts that touch to make a circuit, and separate to break the circuit. The contact material is chosen for its resistance to corrosion, because most metals form insulating oxides that would prevent the switch from working. Sometimes the contacts are plated with noble metals. They may be designed to wipe against each other to clean off any contamination. Nonmetallic conductors, such as conductive plastic, are sometimes used. The moving part.thatapplies the operating force to the contacts is called the actuator, and may be a toggle or dolly, a rocker, a push-button or any type of mechanical linkage[6].

1.7.2 Biased Switches

A biased switch is one containing a spring that retutns the actuator to a certain position. The "on-off" notation can be modified by placiiig parentheses around all positions other than the resting position. For example, an (011)-off-(on) switch can be switched on by moving the actuator in either direction away fröl11 the centre, but retums to the central off position when the actuator is released.

The momentary push-button switch is a type of biased switch. The most common type is a push-to-make switch, which makes contact when the button is pressed .and breaks when the button is released. A push-to-break switch, on the other hand, breaks contact when the button is pressed and makes contact when it is released. An example of a push-to-break switch is a button used to release a door held open by an electromagnet. Ch[illegible] push button switches do exist but are even less common.

1.7.3 Power Switching

When a switch is designed to switch significant power the transitional state of the switch as well as the ability to stand continuous operating currents must be considered.

When a switch is on its resistance is near zero and very little power is dropped in the contacts, when a switch is in the off state its resistance is extremely high and even less power is dropped in the contacts. However when the switch is flicked the resistance must pass through a state where briefly a quarter (or worse if the load is not purely resistive) of the loads rated power is dropped in the switch.

For this reason most power switches (most lightswitches and almost all larger switches) have spring mechanisms in them to make sure the transition between on and off is as short as possible regardless of the speed at-which the user moves the rocker.

1.7.4 Contact Bounce

Contact bounce (also called chatter) is a commôfi.prôblem with mechanical switches and relays. Switch and relay contacts are usually made ofspfingy metals that are forced into contact by an actuator. When the contacts strike tögether, their momentum and elasticity act together to cause bounce. The result is a rapidly pulsed electrical current instead of a clean transition from zero to full current. The Waveform is then further modified by the parasitic inductances and capacitances in the switch aııdWifing, resulting in a seri es of damped sinusoidal oscillations. This effect is usually unnctieeable in AC mains circuits, where the bounce happens too-quickly to affect most equipment, but causes problems in some analogue and logic circuits that are not designed to cope with oscillating voltages.

Sequential digital logic circuits are particularly vulnerable to contact bounce. The voltage waveform produced by switch bounce usually violates the amplitude and timing specifications of the logic circuit. The result is that the circuit may fail, due to problems such as metastability, race conditions, runt pulses and glitches. There are a number of techniques for debouncing (dealing with switch bounce) They can be split into timing based techniques and Hysteresis based techniques.

1.8 Loudspeaker

A loudspeaker, or speaker, is an electromechanical transducer which converts an electrical signal into sound. The term Joudspeaker is used to refer to both the device itself, and a complete system consisting of one or more loudspeaker drivers (as the individual units are often called) in an enclosure. The loudspeaker is the most variable element in an audio system, and is responsible for marked audible differences between systems.

1.9 Microphone

A microphone, sometimes referred to as a mike or mic (both IPA pronunciation: [maɪk]), is an acoustic to electric transducer fhatconverts sound into an electrical signal. Microphones are used in many applications such as telephones, tape recorders, hearing aids, motion picture production, live and recorded audio engineering, in radio and television broadcasting and in computers for recording voice, VoIP and numerous other computer applications.

1.9.1 Principle of Operation

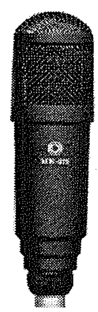

Figure 1.4: An Oktava condenser microphone.

A microphone is a device made to capture waves in air, water or hard material and translate it to an electrical signal. The most common method is via a thin membrane producing some proportional electrical signal. Most microphones in use today for audio use electromagnetic generation (dynamic microphones), capacitance change (condenser microphones) or piezoelectric generation to produce the signal from mechanical vibration[7]

1.10 Safety Guidelines

In this project applications of low voltage are used. So here safety guidelines are not included human safety but included components safety. Also the technical mistakes whish can occur during connecting parts to the circuit cannot be avoided, so heat and current should be taken carefully.

- üne of the component which are used in this circuit is the I.C., which is so sensitive, so while connecting its pins to <the circuit they have to be attached in accordance with the manufacturing instructions layouts in order to keeping it working properly and without damaging it.
- An other component used in this circuitislöudspeaker, which has to be chosen suitable to the out put signal so as not to destrôy diaphfagm.
- An other component used in this circuit is capclcitor.Jt should be taken care about connecting it in right way to avoid damaging it
- While connecting the circuit components to tlie pgyveı- supply we have to be aware of misconnecting its polarity to assure the safety of useôcomponents.
- While the circuit is on, avoid touching the sensitive Côrhp<Jnerits]ike the diodes and capacitors and I.C. to avoid interfering with the out put signal:
- While soldering the parts to the circuit we have to be careful so as not tö burrsthe parts which are sensitive and can be harmed by heat.

1.11 Summary

This chapter covered background information on electronic circuit components. In addition safety guidelines for hardware electronic project were presented.

CHAPTERTWO

VOICE FREQUENCY

2.1 Overview

A voice frequency (VF) or voice band is one of the frequencies, within part of the audio range, that is used for the transmission of speech.

The term voice frequency can also be used to refer to the band of the electromagnetic spectrum between 300 and 3000 Hz.

Voice Inversion

Voice Inversion scrambling is an analog method of obscuring the content of a transmission It sometimes used in public service radio, automobile racing, cordless telephones and the Family Radio Service. Without a descrambler, the transmission makes the speaker sound like "Donald Duck". It is called "voice inversion", but the technique operates on the passband of the information and so can be applied to any information being transmitted.

There are various forms of voice inversion which offer differing levels of security. Overall, voice inversion scrambling offers little true security as software and even hobbyist kits are available from kit makers for scrambling and descrambling. The cadence of the speech is not changed. It is often easy to guess what is happening in the conversation by listening for other audio cues like questions, short responses and other language cadences.

The simplest form of voice inversion splits the voice information into two bands and inverts them around a carrier frequency. This will make the low tones of your voice sound like high ones and vice versa.

There are more advanced forms of voice inversion which are more complex and require more effort to descramble. üne method is to use a random code to choose the carrier frequency and then change this codein feal time. This is called Rolling Code voice inversion and one can often hear the "ticks" i:ıi llietrarismission which signal the changing öf the inversion point.

Another method is Split Band Voice Inversion sometimes called VSB. This is where the band is split and then each band is inverted separately. A rolling code an also be added to this method for Split Band Variable lnversion.

Common carrier frequencies are: 2.868K.Hz, 2.632K.Hz, 2. 718K.Hz, 2.868K.Hz, 3.023K.Hz, 3.107K.Hz, 3.196KHz, 3.333K.Hz, 3.339K.Hz, 3.496K.Hz, 3.729K.Hz and 4.096K.Hz [8].

2.3 Voiceband

In electronics, voiceband means the typical human hearing frequency range that is from 20Hz to 20K.Hz. In telephony, it means the frequency range normally transmitted by a telephone line, generally about 200-3600 Hz. Frequency-division multiplexing in telephony normally uses 4 K.Hz carrier spacing. The rate at which the amplitude of a signal drops off near the upper and lower limits can vary with the design of the filters.

2.3.1 Why use Voiceband Leased Line Modenıs?

1. Many PTTs do not offer pure galvanic copper connections (as needed for baseband modems) or cannot guarantee that the whole link will be over copper wire. Most analogue leased lines are routed via the telephony infrastructure using PCM or carrier systems. For these lines the only option is to use a voice band type of modem that uses signals that can be transported via the telephone network.

2. In a great number of countries, voiceband transmission is the only option because of the lack ofa long distance digital infrastructure.

3. Voiceband modems in combination with analogue leased lines offer a cost effective altemative to digital leased lines for <lata rates below 64 Kbps.

4. Voiceband modems can be used over long--distanceJines. Baseband modems can only cover a limited distance which depends on thet:rarı.s:riiitteddata rate.

2.4 Voice coil

A voice coil is the coil of wire attached to the apex of the moving cone of a loudspeaker. It provides the motive force to the cone by the reaction of magnetic field to the current passing through it.

2.4.1 Operation

By driving a current through the voice coil, a magnetic field is produced. This magnetic field causes the voice coil to react to the magnetic field from a permanent magnet fixed to the speaker's frame, thereby moving the cone of the speaker. By applying an audio waveform to the voice coil, the cone will reproduce the sound pressure waves, corresponding to the original voice, music, ete.

2.4.2 Design Considerations

Because the moving parts of the speaker must be of low mass (to accurately reproduce high-frequency sounds), voice coils are usually made with the lightest-weight wire possible. Because of this, passing too much current through the coil can cause it to overheat. Voice coils wound with flat wire (so-called flat-wound voice coils) are better able to dissipate heat than coils made of round wire. Modem coils may also use a ferrofluid in the gap between the coil and the magnet frame to focus the magnetic field and assist in cooling the voice coil under high power conditions; the ferrofluid conducts heat away from the voice coil to parts of the speaker that have both more thermal mass and are better-able to dissipate the heat.

Excessive current can also cause the voice coil to extend beyond its normal excursion limits, causing a thumping noise and distortion. In extreme cases the voice coil has been known to tear off the cone.

2.4.3 Other Uses for the Term

Nowadays the term has been generalized and refefstô any coiled wire that is used to move an object back-and-forth within a magnetic field: Iıfpafticular, it is commonly used to refer to the coil of wire that moves the fead-write disk heads in a moving-head disk drive. in this application, a very light weight coil of wires is mounted within a very strong magnetic field produced by rare earth permanent magnets. By means of a servomechanism driving the voice coil, the heads of the disk driye can be positioned very quickly and accurately [9].

2.5 Voice Identification

The forensic science of voice identification has come a long way from when it was first introduced in the American courts back in the mid 1960's. in the early days of this identification technique there was little research to support the theory that human voices are unique and could be used as a means for identification. There was also no standardization of how identification was reached, or even training or qualifications necessary to perform the analysis. Voice comparisons were made solely on the pattem analysis ofa few commonly used words. Due to the newness of the technique there were only a few people in the world who performed voice identification analysis and were capable of explaining it to a court. Gradually the process became known to other scientists who voiced concems, not as to the validity of the analysis, but as to the lack of substantial research demonstrating the reliability of the technique. They felt that the technique should not be used in the courtroom without more documentation. Thus the battle lines were drawn over the admissibility of voice identification evidence with proponents claiming a valid, reliable identification process and opponents claiming more research must be completed before the process should be used in courtrooms.

Today voice identification analysis has matured into a sophisticated identification technique, using the latest technology sciertce has to offer. The research, which is still continuing today, demonstrates the validityarıdreliability ofthe process when performed by a trained and certified examiner using established, standardized procedures. Voice identification experts are found all over lhe World.. No longer limited to the visual comparison of a few words, the comparisôn>öfhunıan voices now focuses on every aspect of the words spoken; the words themselves,the.Waythe words flow together, and the pauses between them. Both aural and spectrographicarialysis is combined to form the conclusion about the identity of the voices in question,

The road to admissibility of voice identification evidenceinJhe. courts of the United States has not been without its potholes. Many courts have had Jo rule on this issue without having access to all the facts. Trial strategies and budgets hav.e resulted in incomplete pictures for the courts. To compound the problem, courts have utilized different standards of admission resulting in different opinions as to the admissibility of voice identification evidence. Even those courts which have claimed to use the same standard of admissibility have interpreted it in a variety of ways resulting in a lack of consistency. Although many courts have denied admission to voice identification evidence, none of the courts excluding the spectrographic evidence have found the technique unıeliable. Exclusion has always been based on the fact that the evidence presented did not present a clear picture of the technique's acceptance in the scientific community and as such, the court was reluctant to rely on that evidence. The majority of courts hearing the issue have admitted spectrographic voice identification evidence.

2.5.1 The Sound Spectrograph

The sound spectrograph, an automatic sound wave analyzer, is a basic research instrument used in many laboratories for research studies of sound, music and speech. It has been widely used for the analysis and classification of human speech sounds and in analysis and treatment of speech and hearing disorders.

The instrument produces a visual representation of a given set. of sounds in the parameters of time, frequency and amplitude. The analog spectrograph is composed of four basic parts; (1) a magnetic tape recorder/playback unit, (2) a tape scanning device with a drum which carries the paper to be marked; (3) an electronic variable filter, and{4) an electronic stylus which transfers the analyzed information to the paper. The analog sound spectrograph samples energy levels in a sfüa.Uifrequency range from a magnetic tape recording and marks those energy levels ön electrically sensitive paper. This instrument then analyses the next small frequency :ra:nge· atı.d samples and marks the energy levels at that point. This process is repeated ufüil tfıe entire desired frequency range is analyzed for that portion of the recording. Tlie firtısl:ıed product is called a spectrogram and is a graphic depiction of the pattems, in the fôrn:ı ôf hars or formants, of the acoustical events during the time frame analyzed. The machitı.e will produce a spectrogram in approximately eighty seconds. The spectrogram is in the form of an X,Y graph with the X axis the time dimension, approximately 2.4 seconds in length, and the Y axis the frequency range, usually 0 to 4000 or 8000 Hz. The degree of darkness of the markings indicates the approximate relative amplitude of the energy present for a given frequency and time.

Recent developments in sound spectrograph have produced computerized digital sound spectrographs ranging from dedicated digital signal analysis workstations to PC-based systems for acquisition, analysis editing, and playback. These sophisticated computer-based systems provide high fidelity signal acquisition, high- speed digital processing circuitry for quick and flexible analysis, and CD-quality playback. The computerize-based systems accomplish all the same tasks of the analog systems, but with the cornputer-based systems the examiner gains a host of comparison and measurement tools not available with the analog equipment.. The computer-based systems are capable of displaying multiple sound spectrogram, adjusting the time alignment and frequency ranges and taking detailed numeric measurements of the displayed sounds. With these advances in technology, the examiner widens the scope of the analysis to create a more detailed picture of the voice or sound being analyzed.

The accuracy and reliability of the sound spectrograph, either analog or digital, has never been in question in any of the courts and never considered an issue in the admissibility of voice identification evidence, This may be due in part to the wide use of the instrument in the field of speech and hearing for non-voice identification analysis of the human voice and, in part to the fact that given the same recording of speech sounds the sound spectrograph will consistently produce the same spectrogram of that speech.

The.corıtest comes in the interpretation oftlıe speôtrögraı:ıs. Proponents of the aural and spectrographic technique of voice identification base their decisi ons on the theory that all human voices are different due to the physical uniqueııess of the vocal track, the distinctive environmental influences in the learning process ôf.speech development, and the unique development of neurological faculties which are responsible for the production of speech. Opponents daim that not enough research has been completed.to validate the theory that intraspeaker variability is less than interspeaker variability.

2.5.2 The Method of Voice Identification

The method by which a voice is identified is a multifaceted process requiring the use of both aural and visual senses. In the typical voice identification case the examiner is given several recordings; one or more recordings of the voice to be identified and one or more recorded voice samples of one or more suspects. It is from these recordings the examiner must make the determination about the identity ofthe unknown voice [10].

The first step is to evaluate the recording of the unknown voice, checking to make sure the recording has a sufficient amount of speech with which to work and that the quality of the recording is of sufficient clarity in the frequency range required for analysis. The volume of the recorded voice signal must be significantly higher than that of the environmental noise. The greater the number of obscuring events, such as noise, music, and other speakers, the longer the sample of speech must be. Some examiners report that they reject as many as sixty percent of the cases submitted to them with one of the main reasons for rejection being the poor.quality of the recording of the unknown voıce.

üince the unknown voice sample has been dyterınined to be suitable for analysis, the examiner then tums his attention to the voice sarııplys of the suspects. Here also, the recordings must be of sufficient clarity to _allow coınpa:ıjson, .although at this stage, the recording process is usually so closely controlled that tlıe guality of recording is not a problem.

The examiner can only work with speech samples which are the saıııe as the text of the unknown recording. Under the best of circumstances the süspec:ts willrepeat, several times, the text of the recording of the unknown speaker and these wô:rds will be recorded in a similar manner to the recording of the unknown speaker. For exaniple, if the recording of the unknown speaker was a bomb threat made to a recorded telephone line then each of the suspects would repeat the threat, word for word, to a recorded telephone line. This will provide the examiner with not only the same speech sounds for comparison but also with valuable information about the way each speech sound completes the transition to the next sound.

There are those times when a voice sample must be obtained without the knowledge of the suspect. it is possible to make an identification from a surreptitious recording but the amount of speech necessary to do the comparison is usually much greater. If the suspect is being engaged in conversation for the purpose of obtaining a voice sample, the conversation must be manipulated in such a way so as to have the suspect repeat as many of the words and phrases found in the text of the unknown recording as possible.

The worst exemplar recordings with which an examiner must work are those of random speech. It is necessary to obtain a large sample of speech to improve the chances obtaining a sufficient amount of comparable speech.

As in any other form of identification analysis, as the quality of the evidence with which the exaıniner has to work declines, the greater the amount of evidence and time necessary to complete the analysis, and the less likely the chance for a positive conclusion.

üince the evidence has been determinedsto be sufficient to perform the analysis, the exaıniner then begins the two step processcof voice. sample comparison; one aural (listening) and the other spectrographic (visual). TheseXareitwô different but interwoven and equally important analytical methods whichthe exa:rrıiı:ief combines to reach the final conclusion. The first step is an aural comparison of the voice samples.2 Here the examiner compares both single speech sounds and series of speechsourids ofthe known and unknown samples. At this stage the exaıniner is conducti:ng; 'a nıınifier of fasks; comparing for similarities and differences, screening out less usefulpôftiôris of the saınples, and indexing the samples for further analysis. An exaınple of the 'iıitial aifral comparison is the screening of the samples for pronunciation similarities or discrepancies such as the word "the" may be said with a short "a" sound ora long "e" sound. If the word is not pronounced in the same manner it loses comparison value.

üince the examiner has located those portions to be used for the analysis, a more detailed aural comparison is undertaken. This comparison can be accomplished in many different ways. üne of the most commonly used methods of aural comparison is re-recording a speech sound sanıp le of the unknown followed immediately by a re-recording of the same speech sounds of the suspect. This is repeated several times so that the final product is a recording of specific speech sounds, in altemating order, by the unknown speaker followed by the suspect. Such comparisons have been greatly facilitated by the use of audio digital recording equipment which allows for the digital recording, storage, and repeated playback of only the desired speech sounds to be examined.

During the aural comparison the examiner studies the psycholinguistic features of the speakers voice. There are a large number of qualities and traits which are exaınined from such general traits as accent and dialect to inflection, syllable grouping and breath pattems. The examiner also scrutinizes the samples for signs of speech pathologies and peculiar speech habits.

The second step in the voice identification process is the spectrographic analysis of the recorded samples. The sound spectrograph is an automatic sound wave analyzer with a high quality, fully functional tape recorder. The speech samples to be analyzed are recorded on the sound spectrograph. The recording is then analyzed in two and one half second segments. The product is a spectrogram, a graphic display of the recorded signal on the basis of time and frequency with a general indication of amplitude.

The spectrograms of the unknown speaker are then visually compared to the spectrograms of the suspects. Only those speech sounds which are the same are compared.[3] The comparisons of the spectrograms are based on the displayed pattems representing the psychoacoustical features of the captured speech. The examiner studies the bandwidths, mean frequencies, and trajectory of vowel formants; vertical striations, distribution of formant energy and nasal resonances; stops, plosives and fricatives; interformant features, the relation of all features present as affected during articulatory changes and any peculiar acoustic patteming.[4] The examiner looks not only for similarities but also for differences. The differences are closely examined to determine if they are due to pronunciation differences or if they are indicative of different speakers.

When the analysis is complete the examiner integrates his findings from both the aural and spectrographic analyses into one of five standard conclusions; a positive identification, a probable identification, a positive elimination, a probable elimination, or no decision. In order to arrive at a positive identification the examiner must find a minimum of twenty speech sounds which possess sufficient aural and spectrographic similarities. There can be no differences either aural or spectrographic for which there can no accounting.

The probable identification conclusion is reached when there are less then twenty similarities and no unexplained differences. This conclusion is usually reached when working with small samples, random speech samples or recordings of lower quality. The result of positive elimination is rendered when twenty differences between the samples are found that can not be based on any fact other than different voices having produced the samples. A probable elimination decision is usually reached when working with limited text or a recording of lower quality. The no decision conclusion is used when the quality of the recording is so poor that there is insufficient information with which to work or when there are too few common speech sounds suitable for comparison.

2.6 Voice System

The primary function of a voice system is to reinforce speech. Podium, lavalier, and handheld microphones are frequently used to achieve this purpose. Two characteristics of a voice system are very important: intelligibility and natura! tona} quality.

2.6.1 Intelligibility

A sound system that does a reasonable job of reproducing the human voice so that the speaker can be understood has performed as intended. Simply stated, for a sound system to have adequate intelligibility, the audience must be able to understand what is spoken.

2.6.2 Natura] Tonal Quality

In order for a voice to sound natural, a sound system's frequency response must be low enough to reproduce the deep male voice but also high enough to replicate a soprano's highest fundamental tone.

Additional tones, called harmonics, are created by the human voice. These harmonics give the voice an open and airy sound. Harmonics are a primary benefit to the blending of several voices. Without these upper harmonics, voices would tend to sound harsh or strident.

A sound system requires a low-frequency response of approximately 80 Hz for those deep male voices. A soprano's highest fundamental tone is approximately 1100 Hz. The upper harmonics are multiples of the fundamental tones, and a sound system needs a frequency response of at least 6,000 to 8,000 Hz for these tones. A system with 10,000 Hz or higher is preferable.

2.7 Analog Voice

This document discusses how analog voice signals are measured, the units used, and the points of reference used when you measure.

The quality of a transmission system is defined by the difference between spoken voice at one end and reproduced voice at the ()ther end. Anyone who uses the telephone experiences both good and bad connections,)aııd can prol:,a,bly describe the quality of a particular connection in a subjective way.

2.7.1 Analog Voice Characteristics

Analog is defined as a signal that has a continuously and smoôthl)'.varyiriğ amplitude or frequency. Humarı speech, and everything else you hear, is in analog form, and early telephone systems were analog as well. Analog signals are often depicted as smooth sine waves, but voice and other signals are more complex than that, since they contain many frequencies. The figure in the Analog Voice Measurement section shows the typical distribution of energy in voice signals.

The vertical axis is relative energy and the horizontal axis is frequency. The figure 2.1 ın the Analog Voice Measurement section shows that the voice frequencies that contribute to speech can extend from below 100 hertz to above 6000. However, most of the energy necessary for intelligible speech is contained in a band of frequencies between 200 and 4000 [*11].*

ln order to eliminate unwanted signals (noise) that can disturb conversations or cause errors in control signals, the circuits that carry the telephone signals are designed to pass only certain frequencies. The ranges of frequencies that are passed are said to be in the pass band. Zero to 4000 hertz is the pass band ofa telephone system voice channel-a VF channel. (Sometimes this band is called a message channel.) Bandwidth is the difference between the upper limit and lower limit of the pass band. Therefore, the bandwidth of the VF channel is 4000 hertz. However, the transmission of speech does not require the entire

VF channel. The voice pass band is restricted to 300 through 3300 hertz. Hence, any signal carried on the telephone circuit that is within the range of 300 to 3300 hertz is called an in-band signal. Any signal that is not within the 300 to 3300 hertz bands, but is within the VF channel, is called an out-of-band signal. Ali speech signals are in-band signals. Some signaling transmissions are iri~baiidand some are out-of-band.

2.7.2 Analog Voice Measurement

Any waveform can be characterized in terms of frequencies and power. The quantities commonly used to describe various aspects of transmission performance are frequency and power. Many performance standards are stated in terms of power at a particular frequency. The unit used to measure ifrequeric)' is the hertz, abbreviated as Hz or seen with the f symbol. Hertz equals one (0.00000000125) cycle per second and measures the waves or frequencies of electric changes each second ..

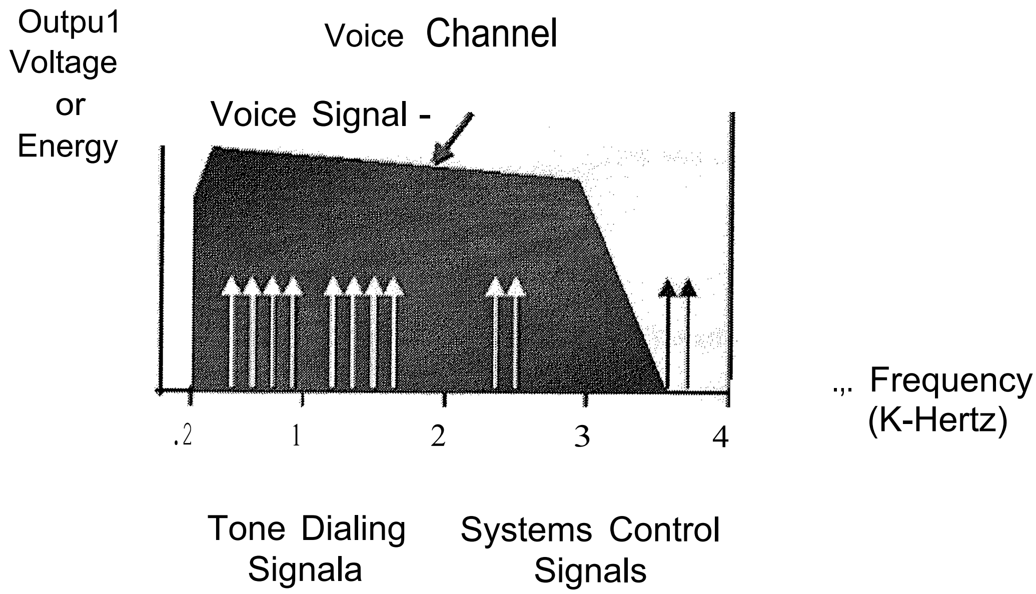

Figure 2.1: Voice measurement

As is common in most electrical systems, power is measured in units of watts, abbreviated W. Since the power encountered in transmission systems is relatively small (compared to the power of a light bulb), power is usually expressed in milliwatts, abbreviated mW.

$$1mW = \frac{1W}{1000} = 0.001W = 10^{-3}W$$

In transmission, the common interest is in power ratios rather than in absolute power. In addition, transmission is concemed with an extremely wide range of absolute power values. For these reasons, a convenient mathematical expression of relative power, the decibel (dB), is commonly used. In order to describe relative power in terms of decibels, you must define the reference point from which you measure. Based upon the transmission parameter that is measured, you can use different forms of decibel measurement. Each form of measurement has a specifically defined reference point. When you use the appropriate units of power related to specific references, you can measure absolute power, relative power, and power gains and losses.

2.7.3 Milliwatt and Hertz

Since the power in telephone circuits is small, the milliwatt is used as the basic power measurement unit, just as the foot is used as the basic measurement of length. Most measurements of absolute power in transmission are made in milliwatts or in units that are directly related to milliwatts.

The frequencies that are used in testing usually fall within the voice frequency band. Commonly used pure (sine wave) test tones are 404 Hz, 1004 Hz, and 2804 Hz. (Th~4-!Jz offset is not always stated. However, actual test frequencies should be offset by 4 Hz in order to compensate for effects that some carrier facilities have on test tones.) A measurement of 1004 Hz is near the voice-band frequencies that carry much of voice power, 404 Hz is near the low end of the spectrum, and 2804 Hz is in the rangeofhigher-frequency components of the voice spectrum that are important to the intelligibility of speech.

In addition to pure test tones, "white noise" within specific frequency ranges is used for certain tests. White noise test tones are complex waveforms that have their power evenly distributed over the frequency range of interest. "White noise" is a signal that contains all the audio frequencies in equal amounts, but which manifests no recognizable pitches or tones

2.8 Electronic Voice Phenomenon

Electronic voice phenomena (EVP) is the practice of using radios, tape recordersorother electronic audio devices in an attempt to pick up communications from ghosts or spirits. EVP proponents claim that some electronic devices can pick up ghostly communications that are often inaudible to the human ear. Skeptics say there are prosaic explanations for the phenomenon rather than communication from ghosts, spirits or other paranormal sources [12].

2.8.1 Raudive Voices

Taking their inspiration from Jürgenson, EVP phenomena were investigated by the German parapsychologist Hans Bender and by the Latvian psychologist Konstantin Raudive. Following the publication of Raudive's book on his research (Breakthrough, 1971) these phenomena are now often referred to as "Raudive Voices" [7].

Professor Bender, notable parapsychologist from the University of Freiburg, , eventually wrote in his conclusion that these voices were "susceptible to a paranormal interpretati on".

Raudive developed several different approaches to recording EVP, and he referred to:

1. Microphone voices: one simply leaves the tape recorder running, with no one talking; he indicated that one can even disconnect the microphone.

2. Radio voices: one records the white noise from a radio that is not tuned to any station.

3. Diode voices: one records from what is essentially a crystal set not tuned to a station.

Raudive delineated a number of characteristics of the voices, (as laid out in

Breakthrough):

1. "The voice entities speak very rapidly, in a mixture of languages, sometimes as many as five or six in one sentence."

2. "They speak in a definite rhythm, which seems forced on them."

3. "The rhythmic mode imposes a shortened, telegrarn-style phrase or sentence."

4. Probably because of this, " grammatical rules are frequently abandoned and neologisms abound."

2.9 Summary

This chapter presented the voice frequency and its types, also the method of voice Identification and measurements, that is used for the transmission of speech.

CHAPTER THREE

CIRCUIT OPERATION

3.1 Overview

In this chapter I anı going to explain the function of IC types which are used in the Donald duck circuit, and dividing the circuit into three sections with brief explanation about each section.

3.2 LF356 JFET Input Operational Amplifier

This op anıp with JFET input devices. This JFET has large reverse breakdown voltages from gate to source and drain eliminating the need for clamps across the inp~s. Therefore large differential input voltages can easily be accommodated without a large increase in input current. The maximum differential input voltage is independent of the supply voltages. However, neither of the input voltages should be allowed to exceed the negative supply as this will cause large currents to flow which can result in a destroyed unit.

Exceeding the negative common-mode limit on either input will force the output to a high state, potentially causing a reversal of phase to the output. Exceeding the negative common-mode limit on both inputs will force the amplifier output to a high state. In neither case does a latch occur since raising the input back within the common-mode range again puts the input stage and thus the anıplifier in a normal operating mode.

Exceeding the positive common-mode limit on a single input will not change the phase of the output however, if both inputs exceed the limit, the output of the amplifier will be forced to a high state.

This amplifier will operate with the common-mode input voltage equal to the positive supply. In fact, the common-rnode voltage can exceed the positive supply by approximately 100 mV independent of supply voltage and over the full operating temperature range. The positive supply can therefore be used as a reference on an input as, for example, in a supply current monitor and/or limiter. Precautions should be taken to ensure that the power supply for the integrated circuit never becomes reversed in polarity or that the unit is not inadvertently installed backwards in a socket as an unlimited current surge through the resulting forward diode within the IC could cause fusing of the intemal conductors and result in a destroyed unit[13].

All of the bias currents in these amplifiers are set by FET current sources. The drain currents for the amplifiers are therefore essentially independent of supply voltage. As with most amplifiers, care should be taken with lead dress, component placement and supply decoupling in order to ensure stability. For example, resistors from the output to an input should be placed with the body close to the input to minimize "pickup" and maximize the frequency of the feedback pole by minimizing the capacitance from the input to ground.

A feedback pole is created when the feedback around any amplifier is resistive. The parallel resistance and capacitance from the input of the device (usually the inverting input) to AC ground set the frequency of the pole. ln many instances the frequency of this pole is much greater than the expected 3dB frequency of the closed loop gain and consequently there is negligible effect on stability margin. However, if the feedback pole is less than approximately six times the expected 3 dB frequency a lead capacitor should be placed from the output to the input of the op amp. The value of the added capacitor should be such that the RC time constant of this capacitor and the resistance it parallels is greater than or equal to the original feedback pole time constant.

3.2.1 LF356 Applications

1. Precision high speed integrators
2. Fast D/A and A/D converters
3. High impedance buffers
4. Wideband, Jow noise, low drift amplifiers

3.2.2 LF356 Features

1. Replace expensive hybrid and module FET op amps
2. Rugged JFETs allow blow-out free handling compared with MOSFET input devices
3. Excellent for low noise applications using either high or .low source impedance very low 1/f comer
4. Offset adjust does not degrade drift or common-mode rejection as in most monolithic amplifiers
5. New output stage allows use of large capacitive Joads (5,000 pF) without stability problems
6. Intemal compensation and large differential input voltage capability

3.2.3 LF356 Schematic Diagram

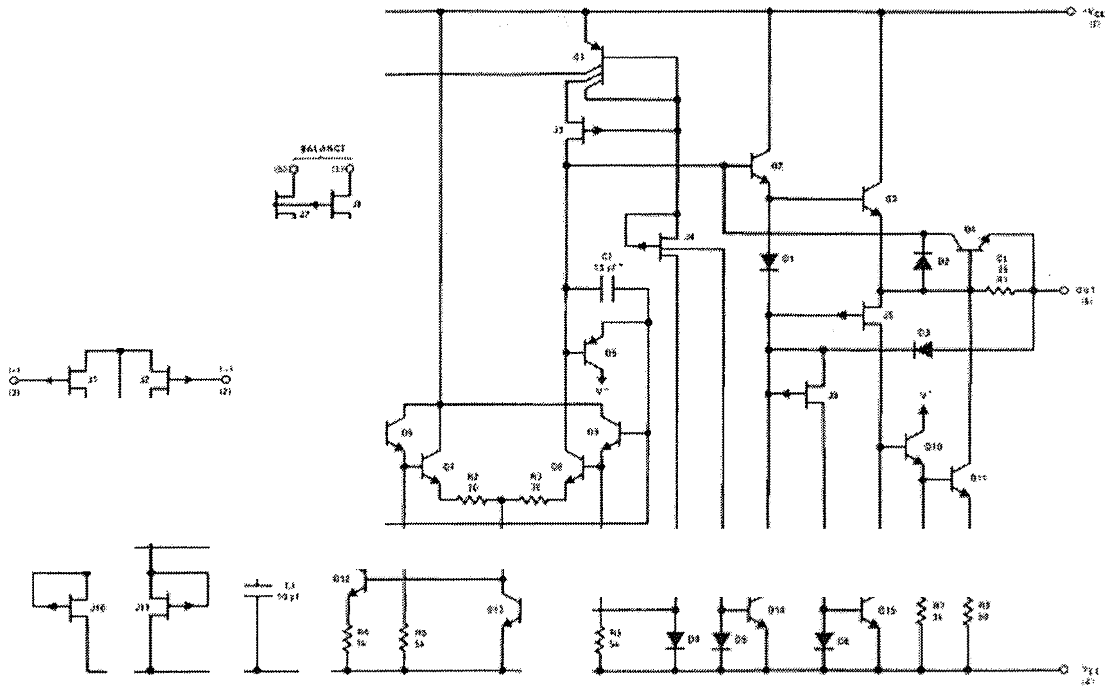

Figure 3.1: LF356 Schematic diagram

3.3 TL074C Low Noise, JFET Input Operational Amplifiers

These low noise JFET input operational amplifiers combine two state-of-the-art analog technologies on a single monolithic integrated circuit. Each intemally compensated operational amplifier has well matched high voltage JFET input device for low input offset voltage. The BIFET technology provides wide bandwidths and fast slew rates with low input bias currents, input offset currents, and supply currents. Moreover, the devices exhibit low noise and low harmonic distortion, making them ideal for use in high fidelity audio amplifier applications.

These devices are available in single, dual and quad operational amplifiers which are pin-compatible with the industry standard MCl741, MC1458, and the MC3403/LM324 bipolar products. [14].

3.3.1 TL074C Features

1. Low power consumption

2.Wide common-mode and differential voltage range

3.Low input bias and offset currents

4.Low noise en=18nV/'1Hz(typ)

5.0utput short-circuit protection

6.High input impedance J-FET input stage

7.Low harmonic distortion: 0.01% (typ)

8.Intemal frequency compensation

9.Latch up free operation

10.High slew rate: 13 V/µs (typ)

3.3.2 TL074C Schematic Diagram

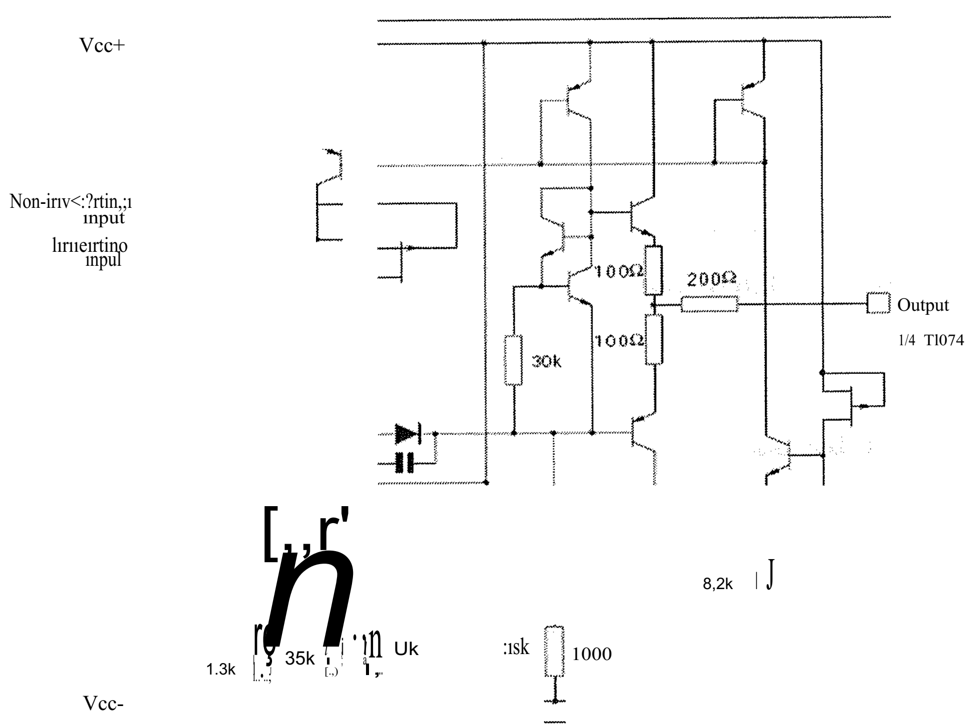

Figure 3.2: TL074C Schematic diagram

3.3.3 TL074C Pin Connections

In table 3.1 we can see the pin connections of TL074C low noise, JFET input operational amplifiers.

Table 3.1: TL074C Pin Connections

PadN	PadName
	Output1
2	Inverting iiput 1
3	Non-inverting input 1
4	
5	Non-:inverting input 2
	Inverting input 2
7	Output 2
8	Output 3
9	Inverting inf}_ut 3
10	NoıF:invertiri_g_input 3
	Vcc -
12	Non-:inverting input 4
13	Inverting jı:ıput 4
·14	Outout 4

3.4 Input Circuit

The input of Donald duck circuit start with the voice signal from the microphone which passing through the 220nF capacitor and variable resistor to ICI (LF356) as shown in figure figure 3.3.

From the name, variable resistor we can know that it is increase and decrease the voice signal which passing through it to ICI which amplified the voice signal, and this JFET have large reverse breakdown voltages from gate to source and drain eliminating the need for clamps across the inputs.

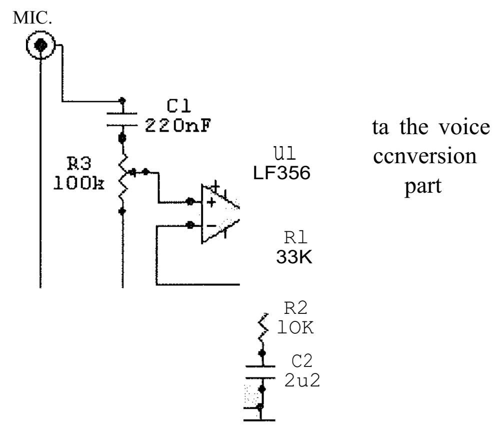

Figure 3.3: The circuit input part

3.5 Voice Conversion

in this part of our circuit we received the amplified voice signal, and here it is divided into four frequency ranges by IC2 (TL074C) which act as band-pass filters A2-A5. ~

The four signals are passed through IC3 (TL074C) half wave rectifies A6-A9. During the negative half cycle, the op amps inverted the signal with unity gain since the diodes then conducts. During the positive half cycles, the diodes are in the blocking state.

Now the frequency of the signal at the junction of the diodes and the feedback resistors is twice that of the input signal at the relevant rectifier. This show us why the frequency range has to be divided into a number of ranges: the more ranges, the smaller, the intermodulation distortion.

The rectifier stages are followed by another set of band-pass filters Alü-Al3 (TL074C) that is tuned to twice the frequency of the filters preceding the rectifiers.

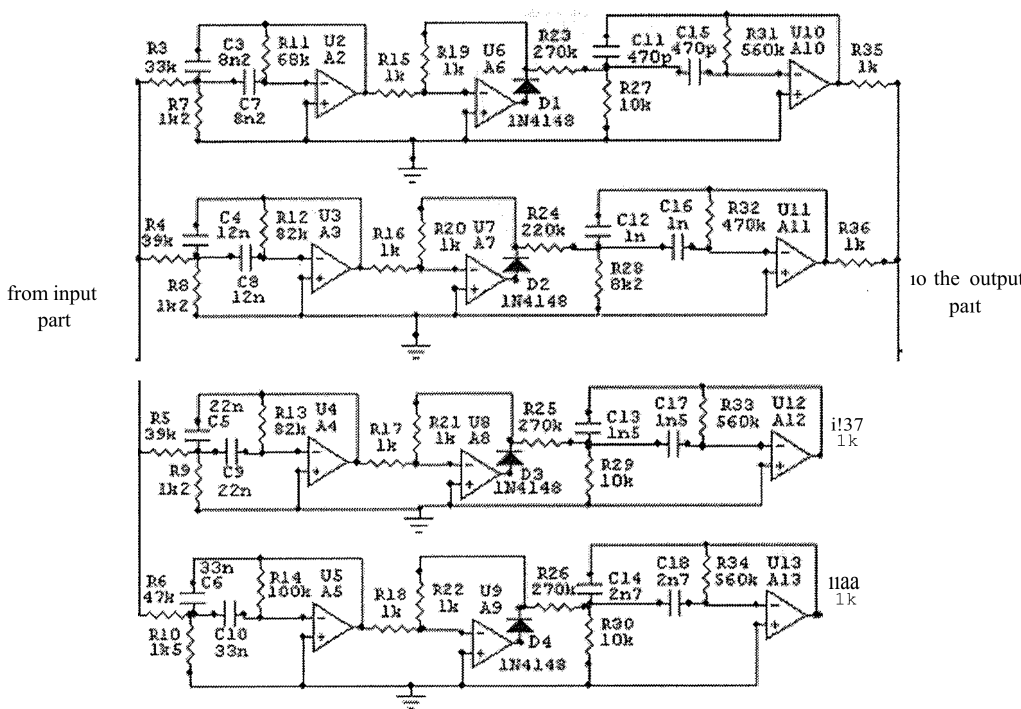

Figure 3.4: The voice conversion part of the circuit

3.6 Output Circuit

In the output the four signals are then recombined to make Donald duck speech available at the output of IC5 (LF356) which amplified the voice signal before we get it from the load speaker.

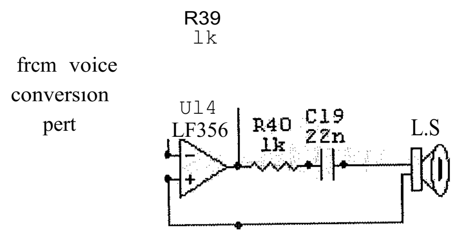

Figure 3.5: The circuit output part

3.7 Summary

This chapter explained the Donald duck circuit operation, and how we amplified the voice signal and filter it by using some electronic device to get the Donald duck speech.

CHAPTER FOUR

PROBLEMS SOLUTION

4.1 Overview

Chapter four will show us the problems that we have seen when we connect the Donald duck circuit and the possible solutions of these problems.

4.2 Digital Multimeter Test

Digital multimeter test was a quick test for the circuit devices, and to check if the voltage can pass through the electronic component in Donald duck circuit, also to protect the devices from the damage. All of this was done before the connection with the power supply machine.

Short circuit was First problem which I had in Donald duck circuit. I connected the circuit with +5V and -5V from the power supply, the power supply shows that there is a short circuit which can damage the devices in the circuit. After that when I checked the circuit again I found out that there was some devices touching each other and this was the reason of short circuit.

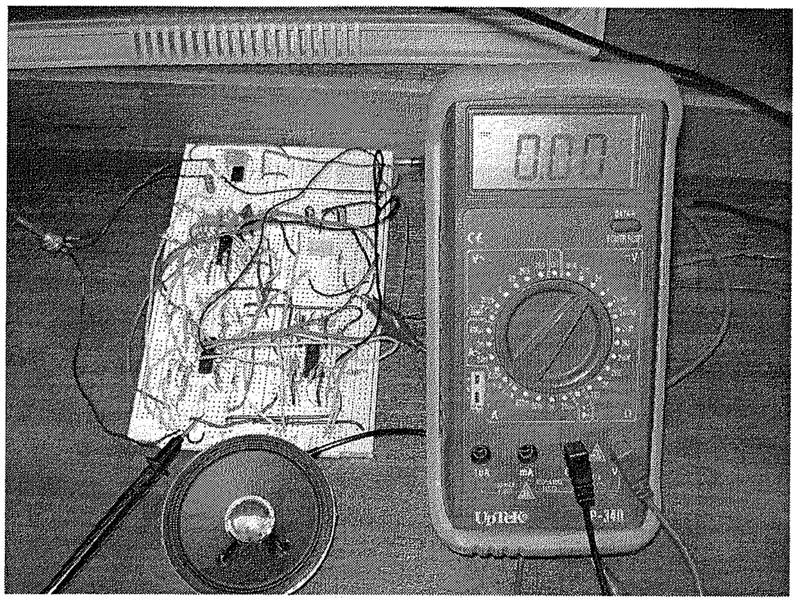

Figure 4.1: digital nnıltinreter machine (DMM)

By digital multi meter as shown in figure 4.1 I checked also the voltage that pass from the power supply machine to the ICs in the circuit. So from leg 7 of ICl and IC5 digital multi meter must measure +5volt and from Jeg 4 must be -5volt, also from leg 4 of IC2,IC3,and IC4 it must measure +5volt and from leg 11 of these ICs must be -5volt, but in ICl and IC3 I found out that the voltage which passing to leg 7 of ICl and to leg 4 of IC3 is less than +5volt. The solution of this problem was to remove the wires that I used for this connection and connect another type of wires because the wires which I used was too thin and it has a small amount of copper so it had a bad connection.

4.3 Oscilloscope Test

Oscilloscope machine is one of the important equipment which we use in the laboratory. By using oscilloscope machine we can check and control the signal that we apply to our circuit.

in Donald duck circuit everything was done on the project board as shown in the circuit diagram but when I tried it I did not get any output voice signal. So by oscilloscope machine I can control the voice signal where it is passing and in which point or node it stops.

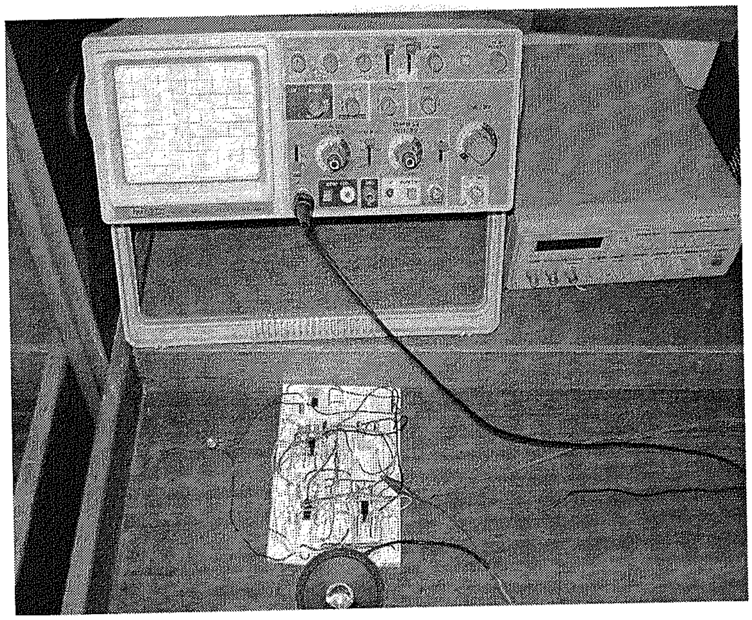

Figure 4.2: Oscilloscope machine

Voice signal passed through the first part of the circuit which is an input part without any problem and here the signal amplified by first IC (LF356). In second part of the circuit (voice conversion part) the signal passed through each node without any problem. So the problem Jeft in the circuit output part. Then when I connected the oscilloscope and talk from the microphone the oscilloscope did not show that the voice signal passing through this part.

I exchanged the devices one by one in the output part of the circuit to check if they are working well or not.

4.4 Circuit Output Solution

As I mentioned before, I tried to connect other devices with a same value just to be sure that the devices or the electronic component which I used is working well because it can be damaged during tests which I did but that does not effect.

So I decide to remove the output circuit part and.try to connect a new part as shown in figure 4.3. After that when I connected the circuit and switched on the power supply I got the Donald duck voice, but it was not so clear and what I mean here that when you speak some time you can not understand some words. Then I start to change some devices in the circuit trying to get the closest voice to Donald duck voice.

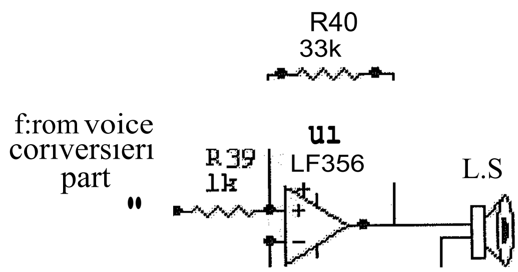

Figure **4.3:** Circuit output part

4.5 Summary

This chapter shows us how we can check our circuit by different methods and different equipment to find out the missing point or the mistake that we did.

CONCLUSION

As known that Donald duck is a famous carton film which everybody know the aim of this project is to show how we can convert the humarı voice into Donald duck voice by using the electronic component.

So first chapter presented an introduction to electrönic component and showed us general information and applications of these electronic component that used in Donald duck circuit, and that help us to connect the circuit in correct way and save our components from the damage. As an example the diodes must be connected with respect to their polarity otherwise they can be damaged, as well as the integrated circuit, because it is very sensitive and it can be damaged from any wrong connection.

in voice frequency chapter we passed over the frequency types and it is operations. Voice frequency or a voice band is one of the frequencies, within part of the audio range that is used for the transmission speech. Voice inversion scrambling is an analog method of obscuring the content ofa transmission, the transmission makes the speaker sound like Donald duck, but the technique operates on the pass band of the information and so can be applied to any information being transmitted.

Third chapter which is the circuit operation showed us the Donald duck circuit in three sections. The input, voice conversion section, and the output section. in the input section I mentioned about the voice frequency amplified by the LF356 amplifier, and in second section (voice conversion section) the voice frequency divided into four frequency ranges by TL074C integrated circuit which act asa band-pass filter. After that the four signals are passed through TL074C (IC3) half wave rectifier. Then the signals followed by another set of band-pass filter. The circuit output part amplified the signal again before the voice comes out from the loud speaker.

Forth chapter shows us the problems that we had in Donald duck circuit and the suitable solution for each problem.

After a great deal of working over this experiment of preparing this project theoretically and practically; it has been found-out that too much knowledge gained and too much techniques leamed by using simple components to get the Donald duck voice.

From this project I get a lot of experience which help me in the end to finish this project. Also I leamed how I can design, build and check the circuit and replace some component trying to get the Donald duck voice.

REFERENCES

[1] http://en.wikipedia.org/wiki/Donald _Duck

[2] http://en.wikipedia.org/wiki/Resistor

[3] http://xtronics.com/kits/ccode.htm

[4] http://www.kpsec.freeuk.corn/components/diode.htm

[5] http ://en. wikipedia. org/wiki/Integrated _circuit

[6] http://en.wikipedia.org/wiki/Switch

[7] http://en.wikipedia.org/wiki/Microphone

[8] http://en.wikipedia.org/wiki/Voice_inversion

[9] http ://en.wikipedia. org/wiki/Voice _coil

[1O]http://www.owlinvestigations.corn/forensic _articles/aural_ spectrographic/method _of voice identification.html

'- [1 1] http://www. cisco. corn/warp/public/7 8 8/signalling/ define_ analog_ voice.html

[12] http://en.wikipedia.org/wiki/Electronic_ voice_phenomenon

[13] web.mit.edu/6.301/www/LFl 55.pdf

[14] www.ensc.sfu.ca/reference/data-sheets/TL07X.PDF