DESIGN AND CONSTUCTION OF A MULTI-USER INTERCOM TELEPHONE SYSTEM

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CERTIFICATION

This is to certify that this project design was carried out by UKWUOMA PIOUS AKUSHIE (98/7889EE), for the award of the Bachelor of Engineering Degree in Electrical & Computer Engineering of the Federal University of Technology, Minna Niger State.

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DECLARATION

I hereby declare that this project design was carried out by be under the supervision of Mr Usman Abraham. During the 2001/2002 academic session.

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DEDICATION

To God Almighty who has made this life worth living.

AND

To my brothers- Mr. Uche Ukwuoma, Paschal Ukwuoma and Chinedu Ukwuoma.

ACKNOWLEDGEMENT

I give God almighty the glory for the steadfast love and kindness showered on me throughout my stay in Minna.

My sincere appreciation goes to my new supervisor Mr. U Abraham who has decided to assist in making my graduation from this department a success. My appreciation also goes to my H.O.D. Engr.(DR.) Y.A Adediran for his fatherly care to the entire students of the department.

I am highly indebted to my siblings who stood there for me morally, financially and spiritually to see that my trying times in school were subdued. Thanks to you all.

Special thanks go to my ever loving mother- Mrs. Philomina Ukwuoma who is making sure that my purpose in life is achieved successfully.

It wouldn't be complete if I fail to appreciate the efforts and cooperation of my friends and colleagues in the department towards the actualization of this work,they are Raji B.K, Adi Terngu. Elijah Olakunle and all the alumni members of the Federal Polytechnic Nekede, Owerri.

Finally, special thanks to my late daddy Mr. J.U Ukwuoma, though the time we spent together was very short, those memories will forever remain in heart. May your gentle soul rest in perfect peace. Amen!

ABSTRACT

Telecommunication has come to stay because of its immense contribution to the betterment of life for mankind. Its importance cannot be overemphasized because without it, life will not worth living.

To this end, The Multi User Intercom system has been developed to help improve on the communication need of man.The system makes use of basic logic gates at the exchange, thereby making it a complete digital exchange to improve performance. Also, CMOS Bilateral switches were used together with decoder to fasten and discriminate between switched terminals. Another vital improvement is the introduction of teleconferencing switch to enable all the terminals to communicate at one time.

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CHAPTER ONE

GENERAL INTRODUCTION

1.0 INTRODUCTION TO COMMUNICATION

Communication is the sending, processing and receiving of signals, ideas, information and messages. The information to be sent takes either of written message, voice message or an electrical signal etc.

A communication system can be defined as a system of sending, transmitting, processing and receiving of signals. The means of communication can be of a radio link, optical fiber, satellite and telephone network. Telecommunication is therefore the transfer of information from one point to a distance point. It is the key part of today's civilization. The prefix 'Tele' is from the ancient Greek word meaning 'Far'. One can now imagine a world without ready access to reliable, economical and efficient means of communication. Therefore, in the world of constant competition, communication is very important to the biological survival of all living creatures. Even animals do communicate by body movement or by making sound to indicate danger. All of these systems of communication are severally restricted by the limitations of the human operator to encode and decode the information.

1.1 BELL'S MAGNETIC TELEPHONE

The basic units of the Bell's invention consist of a transmitter, a receiver and a single wire. The transmitter and receiver were alike, each containing a flexible metallic 9aluminium like) diaphragm and a horse shoe magnet with a wire coil. Sound waves striking the diaphragm cause it to vibrate in the field of the magnet. This vibration generates an electric current in the coil that varies in propagation with vibrations of the diaphragm. The current travels through the wire to the receiving station where changes in the strength of the magnetic field are provided.

This variation in magnetic field strength causes the receiving diaphragm to vibrate which reproduces the original sound.

1.2.0 INTERNAL COMMUNICATION

An internal communication 'INTERCOM' as it is called is a device which allows conversation between people in a miniaturized environment say about 500m radius. The intercom is a cheap and efficient means of communicating within an office block, organization or a parastatal.The design of an intercom is made to suit the need of the environment in which it is to be utilized with adequate considerations for future expansion.

However, with the rapid development of telecommunication in recent years, it has become one of the most interesting subjects of study in the world. The simple design and implementation of the intercom makes it desirable and effective. It has the same operating principles as the telephone network system, the distinguishing factor being the type of transmission employed in intercoms.(the wires interconnecting the various systems of an intercom system must not transverse any public place. They must be confined within the boundary of the premises using the system.) Various classifications are used for intercom systems. One method is based on the control of the stations and is classified as:

i. Principal subsidiary or master-slave system

ii. Independent station system.

In the principal subsidiary system, control is vested on one system called the 'MASTER STATION'. The master station controls the setting up of connections and flows of conversations in the system. It also monitors conversations with the other stations. The master station initiates calls and go on-air and off-air at will. The system usually has two stations. The master and the slave stations. The independent station system has station with independent and equal access to other stations and also privacy of discussion between two parties is guaranteed. Some intercom systems have separate mouth- piece and ear piece, while others use only one transducer which serves both for talking and listening. This is made possible using a mechanical switch commonly called press- to – talk switch:

Another classification is based on transmission system:

i. Wired intercom system

ii. Wireless intercom system.

The wired intercom is more popularly used because the circuit and transmission using cables is relatively cheap and easily achievable. It doesn't require modulation and demodulation as well as privacy of discussion is achieved or else there is a deliberate 'tapping' of the wires physically.

Disadvantages of wired intercom system include:

i. Non- optimum usage of transmission lines.

ii. There is high cost of installation, infrastructure, and at times maintenance.

iii. The intercom system is limited to only where cable would be used.

iv. There are also problems of 'cross-talk' interference.

On the other hand, no form of wiring or dedicated cables is required for the operation of the wireless intercom. Hence the signal is propagated through AC (alternating current) means or modulated and propagated through space at very high frequency (VHF). This type of intercom transmits and receives signal through the existing AC mains and is used to power the system. The voice frequency is modulated at transmitting station and picked up at the other station after demodulation.

The advantages of this type of intercom are as follows:

i. Elimination of difficulties associated with cable installation.

ii. It can be used in any area where there are AC mains lines of the same phase.

iii. It saves the cost of providing dedicated intercom lines which are difficult to maintain.

The disadvantages include:

i. Limitations of communication to only stations connected to the phase of the AC mains.

ii. Increase complexity of circuitry.

iii. It is more expensive to realize and implement.

iv. Privacy could only be assured when there are more than two stations only by transmitting at different frequencies.

Wireless intercom could therefore be summarily put as an important aspect of intercom that involves the frequency or amplitude modulation of a carrier signal with instantaneous values of the modulating signal, and the voice frequency signal.Generally, an intercom system would consist of the following:

i. Ear- piece and mouth –piece (transducer).

ii. Audio circuit (speech path).

iii. Signaling circuit.

iv. Switching circuit.

v. Decoding and conditioning circuit.

1.2.1 <u>LITERATURE REVIEW</u> [1]

It's now taken for granted in developed nations that by pressing a few buttons, people can talk to families, friends, or business associates across the world. The technology that has led to one of the most complex creations of the 20^{th} century, the telephone network has evolved over the past hundred years or so. The first electrical means of communication was not the telephone, however, but the telegraph which allowed message to be sent in codes (usually Morse codes) to be received and printed at a distant location. The age of commercial telegraphy dawned in 1839 by William Fothergil Cooke and Charles Wheastone in London.

In 1889, Almon Stronger developed an automatic switching system that could set up a telephone call without intervention by a human operator. In 1901, Gulgliolimo Marconi demonstrated that, rapid waves could be used to transmit information over long distances when he sent a radio message across the Atlantic Ocean.

In 1947, William Shocey, John Bardenen and Walter Brattain invented transistors. This enable the electronics revolution to take place and provide the basis for a computerized rather than mechanical, telecommunication network. In 1965, Charles Kuo put forward the theory that information could be carried using optical fibers. Optical fibers form the backbone of the global transmission network, because of its speed. The modern telephone network can be viewed as a globally distributed machine that operates as a single resource.

Much of it uses interconnected computers. The network that most people use to carry voice traffic can also be used to transfer data in the form of pictures, texts and images.

1.2 OBJECTIVE OF STUDY

This project was carried out in order to develop a cheap, affordable and efficient means of communication for local industries, organizations or even in a large establishment like the department of Electrical and Computer Engineering where the Head of Department needs to communicate with his members of staff comfortably without wasting energy and time in sending for them, he can conveniently discuss through this affordable intercom system. Past researches have been studied with a view to improve on them by replacing the light Emitting Diodes (LED) at the handed set with tone generator circuits to notify the called. Digital Exchange has also replaced the manual connectors at the control unit. Also a teleconferencing switch has been included.

1.4 SCOPE OF STUDY

The scope of this project is to design an electrical circuit that will enable the sending and retrieval of information within a short distance or within an establishment where time is very precious to waste in walking from one office to the other just to deliver a message. The study carried out in this project has been limited to wired intercom with provision for only three terminal stations which will automatically operate from one handset to the other without any interference in conversation.

1.5 PROJECT METHODOLOGY

This design operates on the method of direct current (dc) flowing through the exchange circuit when one person at each terminal station lifts his handset, this current operates the combinational logic circuit of the exchange which triggers the relays connected to each handsets, immediately the tone generator circuit sounds at any terminal, the called person picks up his earpiece and sound begin to flow through the amplifiers due to the establishment of communication between the caller and the called.

1.6 JUSTIFICATION

As it is a known fact that time is precious for profit oriented establishments, it would save many establishments time and energy to move from one office to the other for search and retrieval of information, in other words, having a cheap, reliable and effective means of communication which is totally automatic eliminating the manned-operator at the exchange would be a welcomed development of our time. This justifies the design of this project.

CHAPTER TWO

TELEPHONE SYSTEM

2.0 THE TELEPHONE PRINCIPLE [2]

Normally, the telephone system comprises the transmitter, receiver, channel, Cable, exchange, and other several components such as switches, alarms, power source.

The telephone system operates on the principle of electromagnetism brought about by varying air pressure. When a person speaks into a microphone, he sends sound as air from the mouth into the microphone; which is then converted into an electrical energy by the microphone at the receiving end. This electrical energy is converted back into sound by the loudspeaker in the earpiece, this is now heard as message. The following is a simplified block diagram of a communication system.

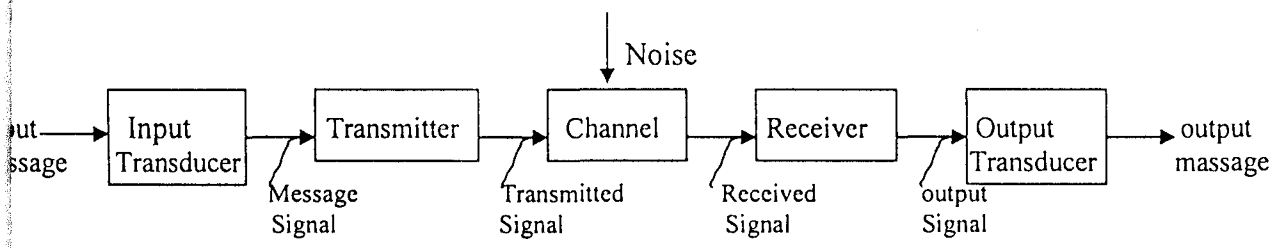

Fig 2.1 : simple block diagram of a communication system

2.1 INPUT TRANSDUCER

A transducer is a device that converts one form of energy to the other. The input transducer converts sound waves in the frequency range of 0.3kHz - 3.4kHz to electrical signal which is fed by wires to the transmitter. Here a microphone acts as the input transducer.

2.2 TRANSMITTER

The transmitter couples the message to the terminal. It is the transmitter that if necessary, will modulate, but here, I avoided modulation because of the use of wired intercom system. The telephone transmitter contains tiny particles of carbon called carbon granules. They are closely held in small compartment between a piece of carbon which is cup-shaped and another piece which is dome – shaped with the aid of moving front electrode, which moves only when the diaphragm converges as a result of changes in air pressure, the carbon granules compresses, thereby increasing their contact area which causes the resistance of the circuit to reduce thereby giving rise to a high flow of current.

2.3 CHANNEL

This is the medium through which the transmitted signals get to the receiver. It may have many different forms ranging from the ground, underground or overhead cables, to sky and space. It could be wired or non-wired channel.

A common characteristic of all channels is that the signal passing through it undergoes degradation which may be due to noise or interference, fading, multiple transmission paths for video signals e.t.c.

2.4 RECEIVER

This extracts and processes the desired signal from the received signal at the output. Amplification of poor signals is performed; delaying of the received signal is also performed. In fact a good receiver should be able to select 'well' the desired signal and reject 'well' any unwanted signal.

2.5 OUTPUT TRANSDUCER

This is a device that converts the received electrical output signal into the desired form by the user, hence sound. The output transducer hence is the loud speaker which converts electrical signal to sound waves; other examples are cathode ray tube (CRT), meters and oscilloscopes.

2.6 OPERATIONAL AMPLIFIERS [4]

The basic amplifier is represented below in fig 2.2. The amplifier has two inputs, which are denoted by V_{i+} and V_{i-}, and a signal output, V_0. Positive and negative power supplies of equal magnitude are normally used (although single supply operation is possible) and are shown as $+V_s$ and $-V_s$ (for simplicity these connectors are not normally shown on circuit diagrams).

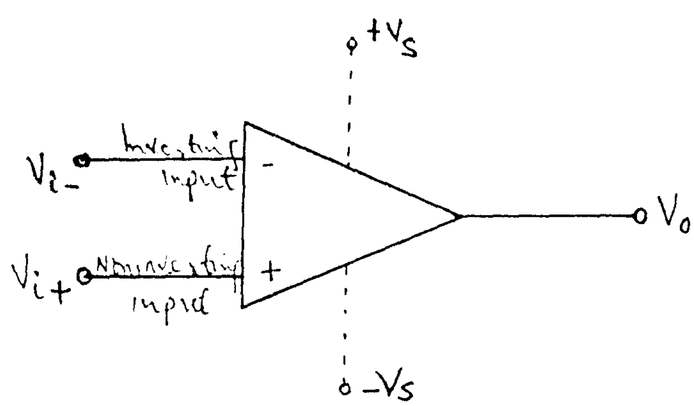

Fig2.2: Basic Operational Amplifier symbol.

Ideal operation of the amplifier is shown in the transfer characteristic of the fig 2.3 below.

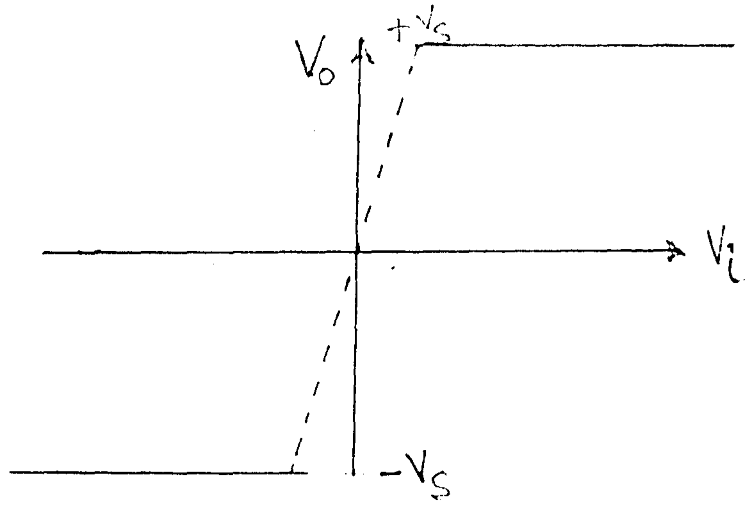

Fig2. 3 Ideal transfer characteristics (solid line)

Here, V_i represents the difference between the voltages applied to the two inputs (V_i+ and v_i-). It can be seen that if V_i is positive, even by only a small amount, the output V_0 is positive and constant, having a magnitude slightly less than that of the supply voltage (the output saturation voltage). Similarly negative values of V_i produce a constant output. In practice a finite change in V_i will be needed in order to change V_0 from one level to the other as shown by the dotted line. Also the change over will occur for a value of V_i that is not precisely equal to zero. For a characteristic having a finite slope, the input/output relationship is written as:

$V_o = A\ (V_i+ - V_i-)$ 1.1

Where A is the gain of the amplifier in the region between the two output saturation voltages. The value of A is large for practical amplifiers (typically more than 50,000) and theoretically infinite for ideal ones. A, may be the open loop gain i.e gain without feedback.

2.6.1 INVERTING MODE OF AN AMPLIFIER

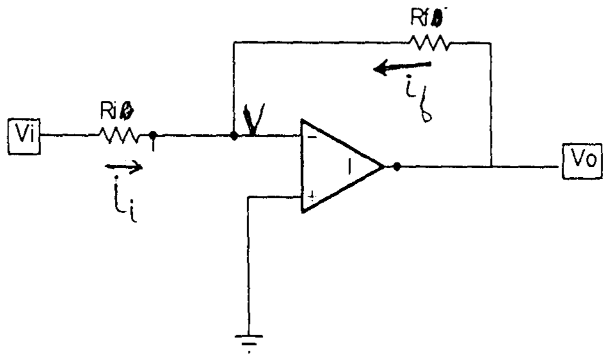

Fig2. 4: Operational Amplifier configuration.

This basic configuration is shown above where the resistors R_i and R_f are the input and feedback resistors respectively. Let the currents in the input and feedback resistor be i_i and i_f, If the input resistance of the amplifier is so high that the current flowing into the inverting input may be neglected then;

$$i_i + i_f = 0 \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots 1.2$$

Applying ohm's law to each resistor, thus

$$\frac{V_i - V}{R_i} + \frac{V_o - V}{R_f} = 0 \quad \ldots\ldots\ldots\ldots 1.3$$

Hence equation 1.1 becomes,

$$V_0 = -AV$$

And $V = -V_o / A$ 1.4

Substituting (1.4) in (1.3)

$$\frac{V_i + V_o/A}{R_i} + \frac{V_o + V_o/A}{R_f} = 0 \quad \ldots\ldots\ldots\ldots\ldots\ldots 1.5$$

For large value of A, V tends to 0, and 1.3 reduces to

$$\frac{V_i}{R_i} + \frac{V_o}{R_f} = 0 \quad \ldots\ldots\ldots\ldots\ldots 1.6$$

Then $V_o + - (R_f / R_i)\, V_i$ 1.7

2.6.2 NON INVERTING MODE OF AN AMPLIFIER

•Sample

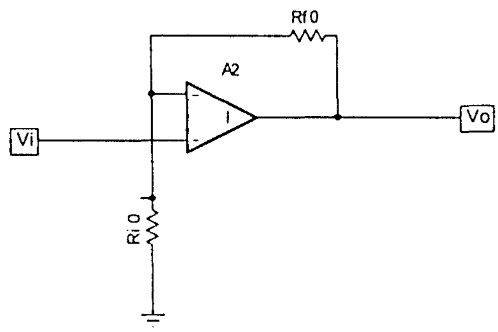

Fig 2.5: Non inverting feedback Amplifier circuit

Considering the circuit above. In some applications, the sign change associated with the inverting mode of operation is not required. The potential V- at the inverting input may be derived from V_o since R_i and R_f from a potential divided;

$$V- = \frac{V_o R_i}{R_i + R_f} \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots 1.8$$

As before, currents flowing into amplifier are assumed to be negligible

From equation (1.1), $V_o = A\,(V_i - V-)$

Therefore, $V_i - V- = V_o / A$ 1.9

As A tends to ∞, $V_i = V-$ and then

$$V_i = V_o \frac{R_i}{R_i + R_f} \quad \ldots\ldots\ldots\ldots\ldots\ldots 2.0$$

$$Vo = Vi \,\frac{(Ri + Rf)}{R_i} = Vi\,(1 + Rf / Ri) \quad \ldots\ldots\ldots\ldots\ldots\ldots 2.1_i$$

2.7 FREQUENCY RESPONSE

2.7.1 OPEN LOOP BEHAVIOUR COMPENSATION [3]

The circuits discussed earlier all depended in the assumption that the open loop gain A remained very large (ideally infinite) under all operating conditions. In practices, this cannot be true for all frequencies. For stable operation with feedback configuration used, the high gain must be preserved for low frequencies including DC.

However, for stable operation under all conditions, the gain must be made to fall or 'roll off' at high frequencies this will occur in any case due to stray capacitance, but additional capacitance is also need in order to define the frequency at which roll-off starts to occur. Roll-off is desirable not only to ensure stability but also to avoid amplification of signals outside. the required range of frequencies, since this would increase the noise content.

This additional capacitance may be internal to the IC amplifier or external (external compensation). Internal compensation has the advantage that stability guaranteed under all operating conditions and an external capacitor is not required. The disadvantage is that open loop bandwidth by the manufacture cannot be change by the user. The widely used UA741 amplifier is of this type.

External compensation gives greater flexibility; but care should be taken as unsuitable choice of compensating capacitor can cause instability.

The simplest way of modeling this effect is by a single low pass filter as shown below.

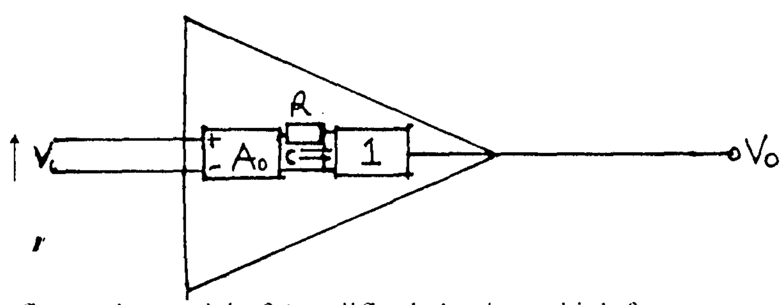

Fig 2.6: first-order model of Amplifier behavior at high frequency.

From the diagram above, C is the total effect of all stray capacitance plus the compensating capacitor

A_o = low frequency (dc) gain

$\boxed{1}$ = ideal buffer amplifier whose gain is unity

Then $V_o = A_oV \dfrac{(1/jwc)}{R + 1/jwc} = A_oV\,(1/(1+jwcR))$ 2.2

Where $w = 2\prod f$

When w tends to 0, Gain tends to A_o

When w tends to ∞, Gain tends to 0

From equation (2.2)

$A = \dfrac{V_o}{V} = A_o\ (1/(1+jwcR))$ 2.3

$A = A_{o\,1} / j\,(w/w_o)$ 2.4

Where $w_o = 1/cR$

In decibels, Gain (dB) = $20Log_{10}/A/ = 20Log_{10}\dfrac{Ao}{\sqrt{1+(w/w_o)^2}}$ ...2.5

it is useful to consider these cases;

1. $w<<w_o$, the gain tends to 20log10A when is the dc gain.
2. $w>>w_o$, this implies that w/wo is much longer than unity, so equation 2.5 becomes;

Gain (dB) + $20Log_{10}A_o + 20Logw_o - 20Log_{10}w$ 2.7

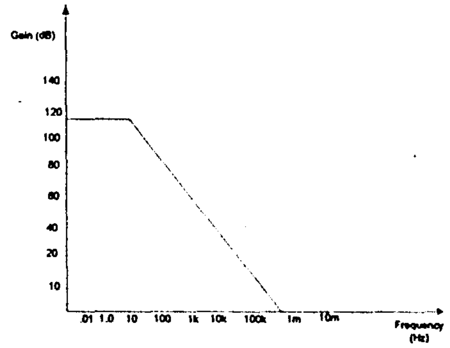

Fig 2.7: Frequency Response of the μA741 amplifier ($f_o \pm 5kz$)

2.8 MOSFET DIGITAL LOGIC CIRCUITS [5]

The MOSFET digital logic circuits can be grouped into two namely; NMOS logic circuit, which contain only n-channel transistors, and CMOS (complementary metal oxide semiconductor) logic circuits which contain both n-channel and p-channel transistors.

2.8.1 NMOS INVERTER

The inverter is the basic circuit of most MOS logic circuits. The design technology used in NMOS logic circuits are developed from the dc analysis results from NMOS inverter. Extending the concepts developed from the inverter to NOR and NAND gates is then direct. The circuit diagram below shows an NMOS inverter with a resistor load.

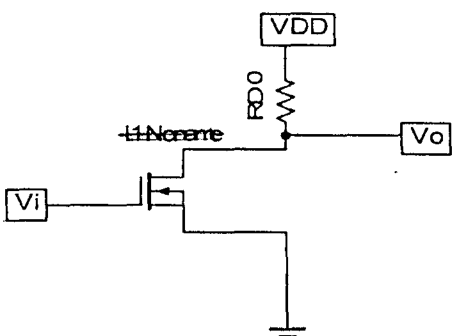

Fig 2.8: NMOS Inverter with Resistor Load.

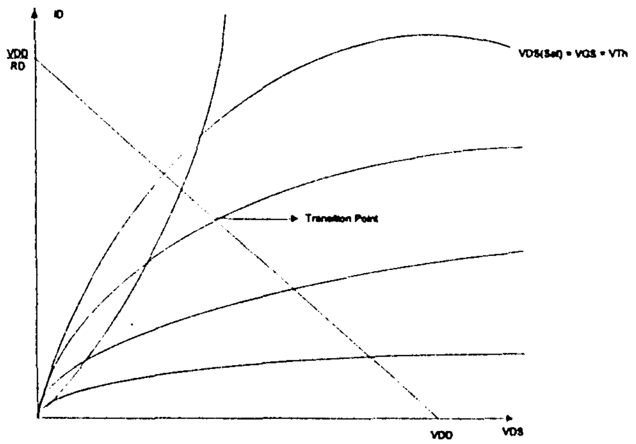

Fig2. 9: Transistor Characteristic and Load line Graph.

When the input voltage is less than or equal to the threshold voltage, or $V_i \leq V_{th}$, the transistor is cut off,

$i_D = 0$, and the output voltage is $V_O = V_{DD}$. The maximum output voltage is designed as logic 1 level. As the input voltage just become greater than V_{th}, the transistor turns ON and is biased in the saturation region.

Then, output voltage $V_o = v_{DD} - i_D R_D$ 2.8

2.8.2 NMOS LOGIC CIRCUITS

NMOS logic circuits are found by combining driver transistor in parallel, series or series-parallel combinations to derive a desired output logic function.

2.8.3 NMOS NOR GATE

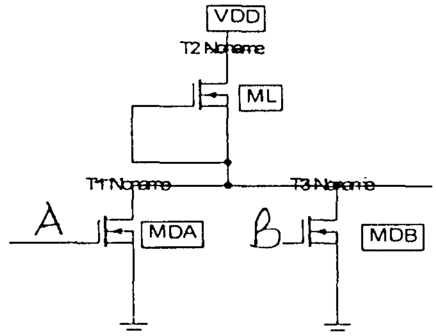

Fig 2.10: Two input NMOS NOR logic Gate with Depletion Load.

The NMOS NOR logic gate contains additional driver transistors connected in parallel. If A = B = logic 0, the both MDA and MDB are cut off and the NMOS inverter configuration with ML and MDA is the same as previously considered and output voltage goes low, similarly, if A = Logic 0 and B -= logic 1, we again have the same inverter configuration. If A = B = logic 1, then both MDA and MDB turn ON and the two driver transistors are effectively in parallel. The value of the output voltage now changes slightly.

The CMOS logic gates also work in the same principle but only that gates of a PMOS and and NMOS are connected together, an additional PMOS and NMOS transistors are connected in series or parallel to form specific logic circuits.

2.8.4 NOISE MARGIN

The word 'Noise' means transient, unwanted variations in voltage or currents. In digital circuits, if the magnitude of noise at logic node is too large, logic errors can be introduced into the system. However, if the noise amplified is less than a specified value, called the NOISE MARGIN, the noise signal will be attenuated as it passes through a logic gate or circuit, while the logic signal will be transmitted without errors.

Noise signal are usually generated outside the digital circuit and transferred to logic nodes or interconnect lines through parasite capacitance or inductances. In digital systems however, the noise margins are usually defined in terms of static voltages say

$NM_L = 0.321V$ and $NM_H = 2.59V$

2.9 TRANSMISSION GATES

Transistor can act as switches between driving circuits and load circuits. Transistor used to perform these functions are called transmission Gates.

CHAPTER THREE

3.0 CIRCUIT DESIGN AND ANALYSIS

Since the objective of this project is to design an intercom system for communication within a department which would be implemented using locally available components; designed as simple as possible; privacy of conversation guaranteed; having a teleconferencing facility; ease of installation and expansion possible.

In the design of this project in order to meet the set objectives, I wish to specify the system design layout as follows:

- Analogue and digital ICs;
- Use of Hard wired system;
- Implementation using a star configuration in which the sets are connected to a common control unit;
- Maximum voltage used is DC 12 V;
- Timer circuits for timing of call duration is 3hrs;
- Maximum current used is 200mA;
- Operating temperature is 0-70^{0}C;
- Output power is $\approx$ 10 W.

3.1 OPERATING PRINCIPLE OF THE DESIGN

The system is divided into three for proper analysis:

1. **Selection circuit:** - This comprises of the handset containing the push-to-stay switches, the monostable 555 timer multivibrators and logic gates attached to the 555 timer, the decoder and relays.

2. **Decoding and conditioning circuit:** - This comprises of the 3-8 decoder that decodes which of the handset that has been lifted up.

3. **Speech part circuit:** - This consists of the microphones and loud speakers, the audio amplifier and bilateral switches.

The project design has three (3) terminals with each containing push-to-stay switches, 555 timer astable tone generators, ear and mouth-pieces.

Let's assume that the first caller to initiate a call is the Terminal 1, he can only call terminals 2 and 3, when he lifts his handset and presses S2 switch, a high state is sent to a combinational logic circuit consisting OR and AND gates and finally to the Reset input of the other 555 timer monostables, this S2 switch then triggers the 555 timer monostable to start ringing. Output of same monostable 2 is sent to a combination logic circuit of an OR gate and AND gate which are connected to the relays RL1 and RL2. Immediately terminal 2 lifts up handset H2 to receive call,a high state is also sent to the decoder and the code 011 is decoded at the address lines of the decoder,its corresponding decimal equivalent output line goes high, this high output is also tied to same logic circuit, finally the relays RL1 and RL2 are triggered ON and communication is established between terminals 1 and 2 through the microphone of H1 to the audio amplifier, to the bilateral switches and back to the audio amplifier, then finally to the loud speaker in H2. However, to avoid interruption from either of the two terminals while communication is going on between two terminals, the output of the other monostable 555 timer multivibrators are tied to a NOR gate and connected as feedback to the first AND gate of the called. So that the called terminal monostable can only be triggered or ring if and only if the two other monostable outputs are low. A one shot monostable multivibrator (CD4538) is used at the terminals to reset it for 1 second so as not to allow residual calls to come in.

3.2 DECODER CIRCUIT [6]

The Decoder is a logic circuit that provides code conversion; it can only have one output high at a time. The CD4028 3-8 decoder as used in the design has the table below:

H3	H2	H1	0	1	2	3	4	5	6	7
0	0	0	1	0	0	0	0	0	0	0
0	0	1	0	1	0	0	0	0	0	0
0	1	0	0	0	1	0	0	0	0	0
0	1	1	0	0	0	1	0	0	0	0
1	0	0	0	0	0	0	1	0	0	0
1	0	1	0	0	0	0	0	1	0	0
1	1	0	0	0	0	0	0	0	1	0
1	1	1	0	0	0	0	0	0	0	1

Table 3.1: Truth Table of a 3-8 Decoder

3.3 IMPLEMENTATION OF SWITCHING USING THE DECODER TRUTH TABLE

When Caller 1 lifts up Handset H1 and tries to call Terminal 2 by pressing S2, the ringer circuit at terminal 2 begins to ring, a high state is sent to the address of the decoder, immediately the called picks up his handset (H2),a high state is also sent to the decoder and a binary code 011 is decoded by the decoder, this code decimal equivalent output line is made high and relay RL1 and RL2 are switched ON by the combinational logic circuits as shown above.

3.4 THE SELECTION CIRCUIT

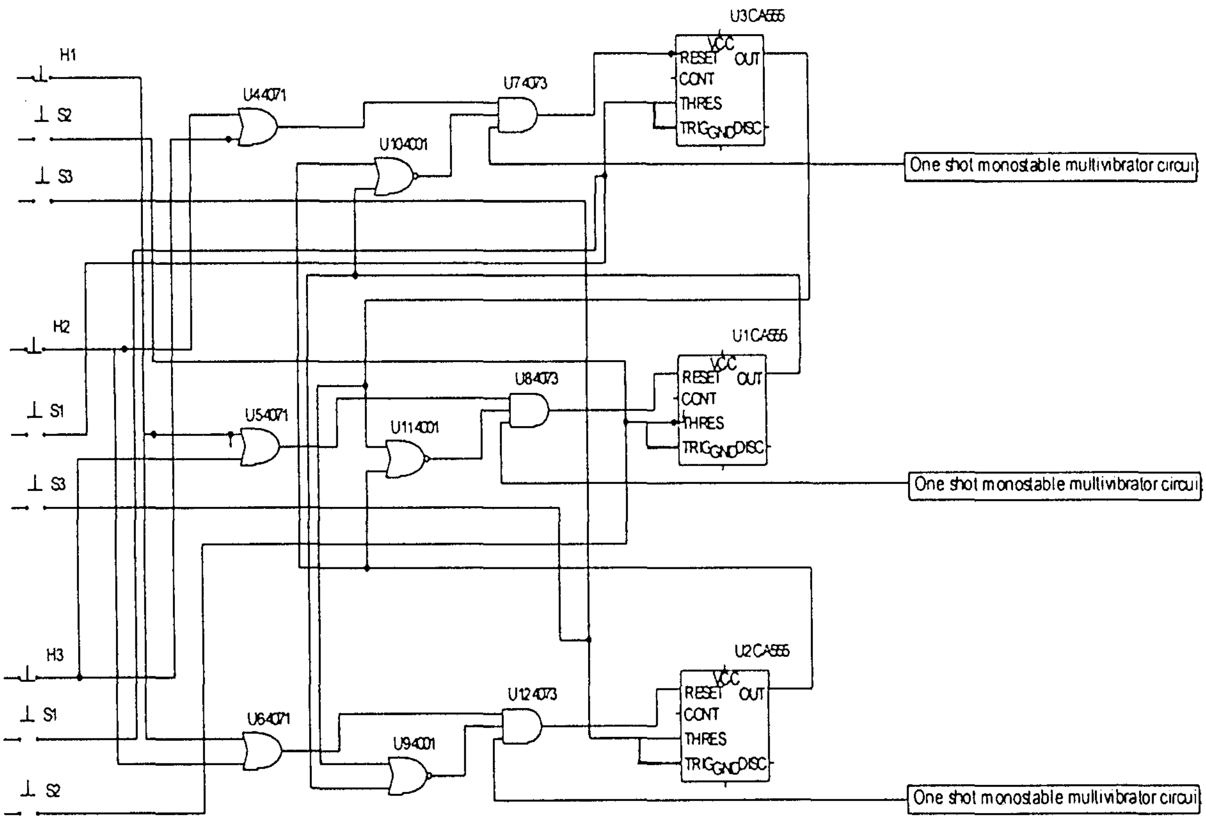

fig3.1: Selection circuit diagram

3.4.1 ONE SHOT MONOSTABLE MULTIVIBRATOR CIRCUIT

This circuit has been included in the selection circuit to make sure that a terminal could be reset for 1 second so that it can receive only fresh calls after reset. So when a handset is dropped, the two other monostables are reset. This was achieved by the using the CD4028 IC. It was found that residual calls were coming in immediately a called terminal drops and there is another handset still hanging up. To avoid this, each hand set has to be reset for one second.

From design calculation, the CD4538 was adopted and reset time was calculated to be T = RC, where R and C are resistor and capacitor each.

To obtain a 1s, I used R = 10MΩ and C = 100nF. Hence $T = 10*10^{6}*100*10^{-9} = 1s$

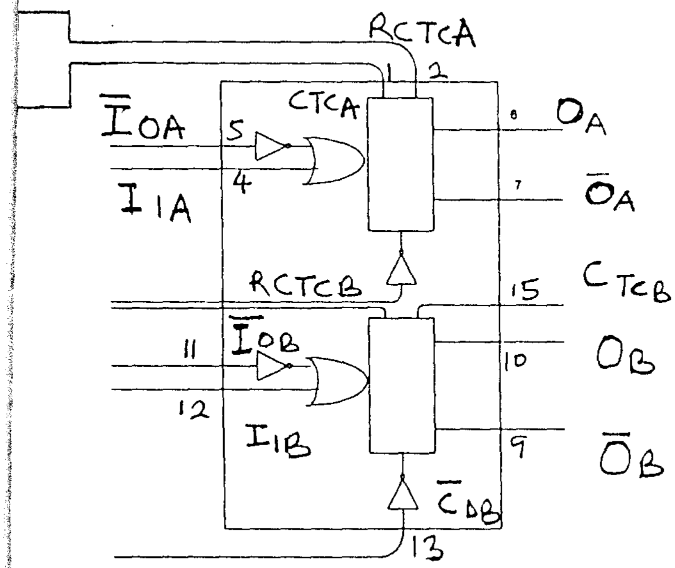

Fig3.2: Connection of the external timing components R and C

3.4.2 555 TIMER USED AS CALL PERIOD TIMER

The 555 timer monostable can be used to time the call period by connecting resistor and capacitor as shown below.

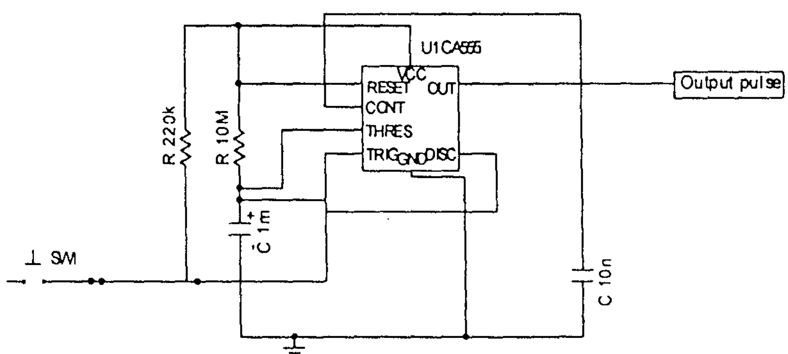

fig3.3: 555 timer monostable as call period timer circuit.

From the figure above, we can determine the pulse duration of the timer by substituting the values of $R=10M\Omega$ and $C=1000uF$ into the equation below:

$T = 1.1RC$

$T = 1.1*10*10^{6}*1000*10^{-6} = 11000$ seconds

$T = 11000/3600 = 3.056 \approx 3hrs$.

3.5 THE SPEECH PATH CIRCUIT

The speech circuit consists of the handset, that is the microphone and earpiece, the LM386 IC, the CD4016A Quad Bilateral switch. The position of the interconnecting points for switching between terminals a, b, c, d and n are shown below where n is an extension terminal. In an unoperational state, all terminals are separated from each other by the relays which remain open. When the switches are pressed to select a particular terminal for conversation, and the decoder enables the relays attached to the two terminals to be switched, a communication is established and speeches made then flow through the mouthpiece of either terminals to the LM386 via the interconnecting cables where it is amplified, the to the CD4016A bilateral switch which has a filtering and distortion characteristic en-route the earpiece at the receiver terminal and an LM386 which further re-amplifies it before it further reaches the earpiece of the receiver. Hence the speech is received at the receiver very clearly.

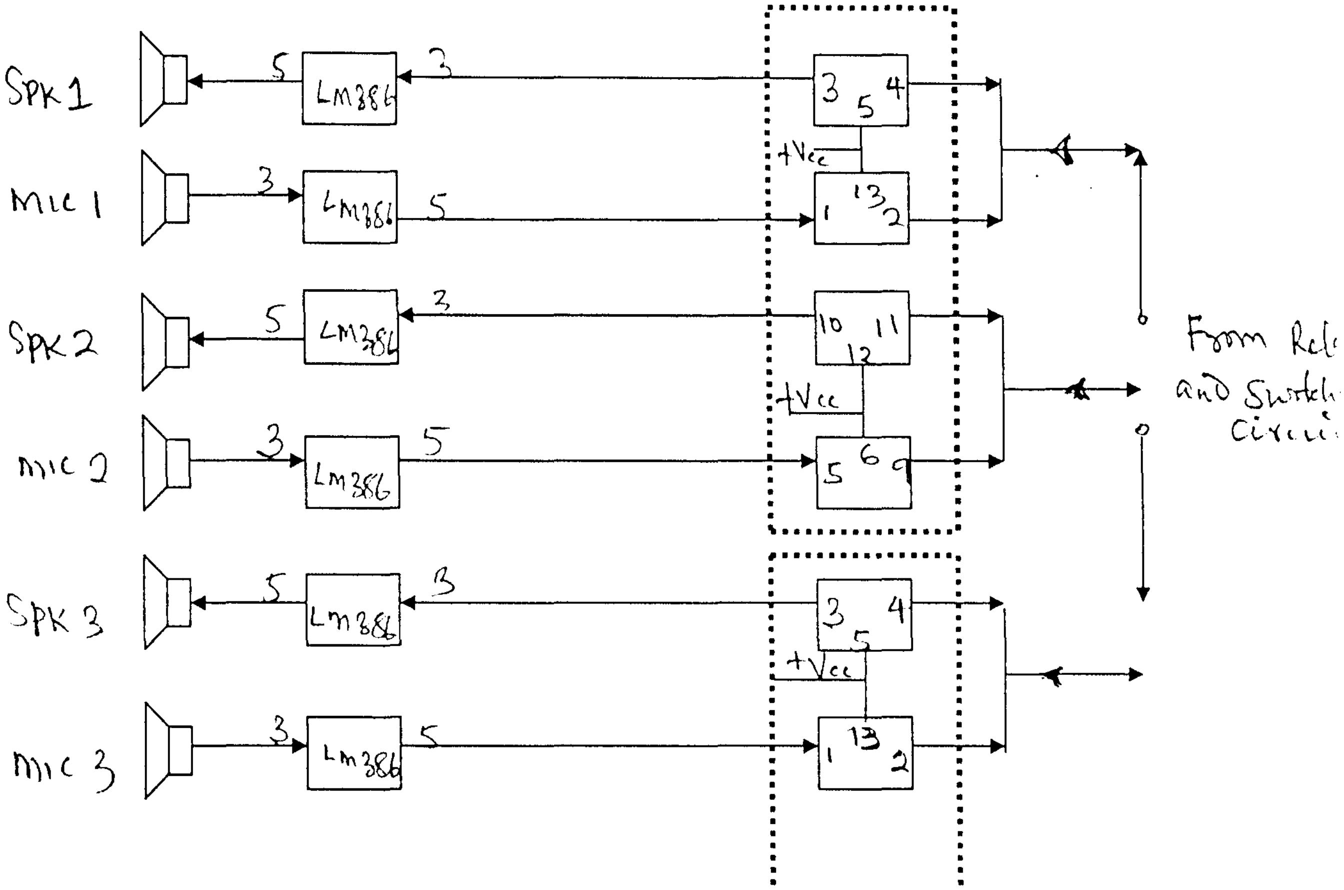

fig 3.4: Speech path from terminal units.

3.6 THE POWER SUPPLY UNIT

The power unit has two outputs of 12V DC and 5V DC. It was designed using the LM 7812 and LM 7805 voltage regulators. It was powered by a 240V AC supply from mains.

The 12V output supply power to the relays and the amplifier while the 5V output supply power to the analogue ICs. Capacitors were now connected in parallel for ripple elimination. A 50 mA anti surge fuse in the transformer primary winding provides further protection against short circuits and excessive mains voltage.

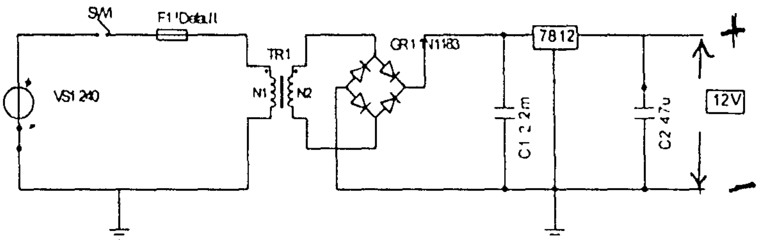

fig3.5: Power supply circuit

3.7 THE 555 TIMER USED AS A TONE GENERATOR

This is a basic 555 square wave oscillator used to produce a 1 KHz tone from an 8 ohm speaker. Frequency is about 1.44/(R1 + 2*R2)C where R1 (220) is much smaller than R2 (470K) to produce a near square wave. Lower frequencies can be obtained by increasing the 470K value, higher frequencies will probably require a smaller capacitor as R1 cannot be reduced much below 220Ω. Lower volume levels can be obtained by adding a small resistor in series with the speaker (10-100 ohms). The speaker is directly driven from the 555 timer output. The series capacitor (100 uF) increases the output by supplying an AC current to the speaker and driving it in both directions rather than just a pulsating DC current which would be the case without the capacitor.

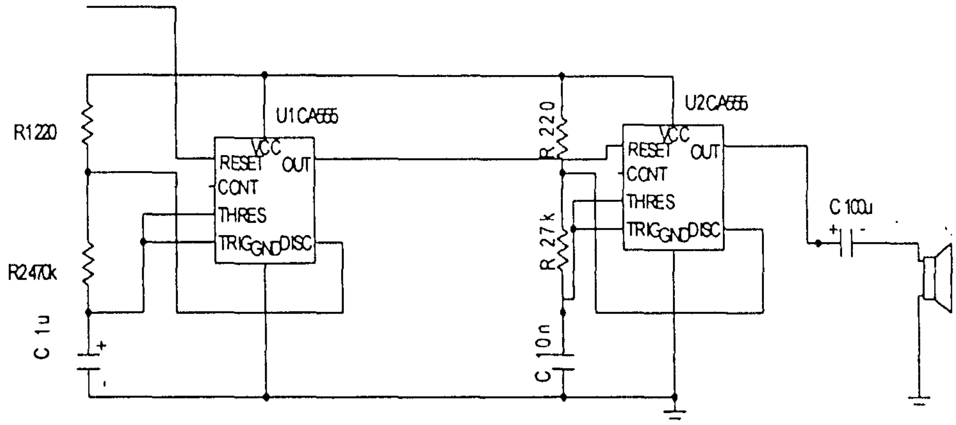

fig3.6: 555 timer astable multivibrator tone generator diagram

from the fig. above, we can note that R1, R2 and C determine the frequency of tone.

$$\text{Hence, frequency} = \frac{1.44}{(R1 + 2R2)C} \quad \ldots\ldots\ldots\ldots\ldots\ 3.1$$

$$f = \frac{1.44}{(220 + 2 * 470k) * 1*10^{-6}}$$

$$= \frac{1.44}{(220 + 940 * 10^{3}) * 10^{-6}}$$

$$= \frac{1.44*10^{6}}{220 + 940 * 10^{3}} \quad = 1.5316\text{Hz}$$

CHAPTER FOUR

PROJECT IMPLEMENTATION AND TESTING

4.0 TESTS

Before construction work on the complete system commenced, each of the subsystems described earlier were simulated in Electronic work bench software and tested after wards on bread board to certify that they worked independently.

Each subsystem was tested one after the other as the design progresses.

- *the amplifier circuits were first tested to ensure that they worked without any noise.*
- *the bilateral switches were tested and coupled to ensure that they were able to switch the voice circuits when addressed.*
- *the switching circuits were implemented and tested on the board.*
- *the signal/selection circuits were also tested.*
- *the tone generator and power supply units were finally coupled and tested.*

The following results were obtained:

Sub-system	Parameter tested	Test Equipment	Results
Amplifier circuit	Noise	12V DC supply and voice input signal	Satisfactory
Tone generator	Output frequency	Electronic work bench simulation	1.5316Hz
Selection circuit	Pulse period	Electronic work bench simulation	3hrs.
Complete system	Range	Communication cable	Not fully determined due to unavailability of lengthy cables
Power supply unit	Workability	Power supply and multimeter	Satisfactory.

Table 4.1: Test and obtained Results

4.1 CONSTRUCTION

The construction of this design was implemented by fixing the speakers on the handsets as well as fixing the switches for each terminal units. Then the central controlling unit was soldered on a 14.5 x 6.5cm Vero board using a 40W soldering iron and 60/40 flux cored soldering lead.

4.1.1 PRECAUTIONARY MEASURES

- All components were tested before soldering on the board.
- The Vero board and component leads were properly cleaned to remove any dust particle that could cause short circuits on the surface.
- Each terminal unit was connected via a 8 wire pvc sheathed single core 0.6mm to allow their mounting away from the central control unit.
- Brush was used to clean out any solder remaining on the board.

4.2 TELECONFERENCING FACILITY

A modification of previous designs was achieved in this project by incorporating a Teleconferencing switch at the control unit. Only the master station can initiate a teleconference by pressing the Teleconferencing switch which causes all the three terminals to ring at once. Hence each person at the terminal picks up the handset and talk simultaneously.

CHAPTER FIVE

SUMMARY AND CONCLUSION

5.1 CONCLUSION

This report covers extensively the design and construction of a three station multi-user intercom system. The system was designed primarily to enhance internal communication within the Department of Electrical and Computer Engineering, F.U.T, Minna. And also any other organization that has the same structure.

The "Centrally controlled independent multi-user intercom" was chosen for its stability to provide privacy of communication and Teleconferencing facility.

The system is divided into two sections, the control unit containing the power unit and the teleconferencing unit and all timing circuits. Then the terminal unit containing the earpiece and mouthpiece.

5.2 LIMITATIONS

The work done here would have been better than this if only some of these limitations were not there, they include:

a. Financial constraints, most of the desired features of this project were not realized because of the financial involvements attached. For example,

More than three terminals would have been implemented by using multiplexer switches or even cascading more than one. Better finishing would have been given to the package to promote marketability.

Longer cables would have been used for longer distance coverage.

The system could have been microprocessor based to enable multi access link.

b. The gain of the LM386 was very low, thus causing low amplification at the output, but after connecting a bypass capacitor across pin 1 and pin 8, the gain increased from 20dB to about 200dB.

5.3 RECOMMENDATIONS

The work so far can be improved upon by using microprocessor at the control unit to make the system more intelligent in order to be in line with modern trend.

The manually operated turn ON and OFF switches should be replaced with thyristors so that it can automatically go OFF when a communication has been established.

5.4 REFERENCES

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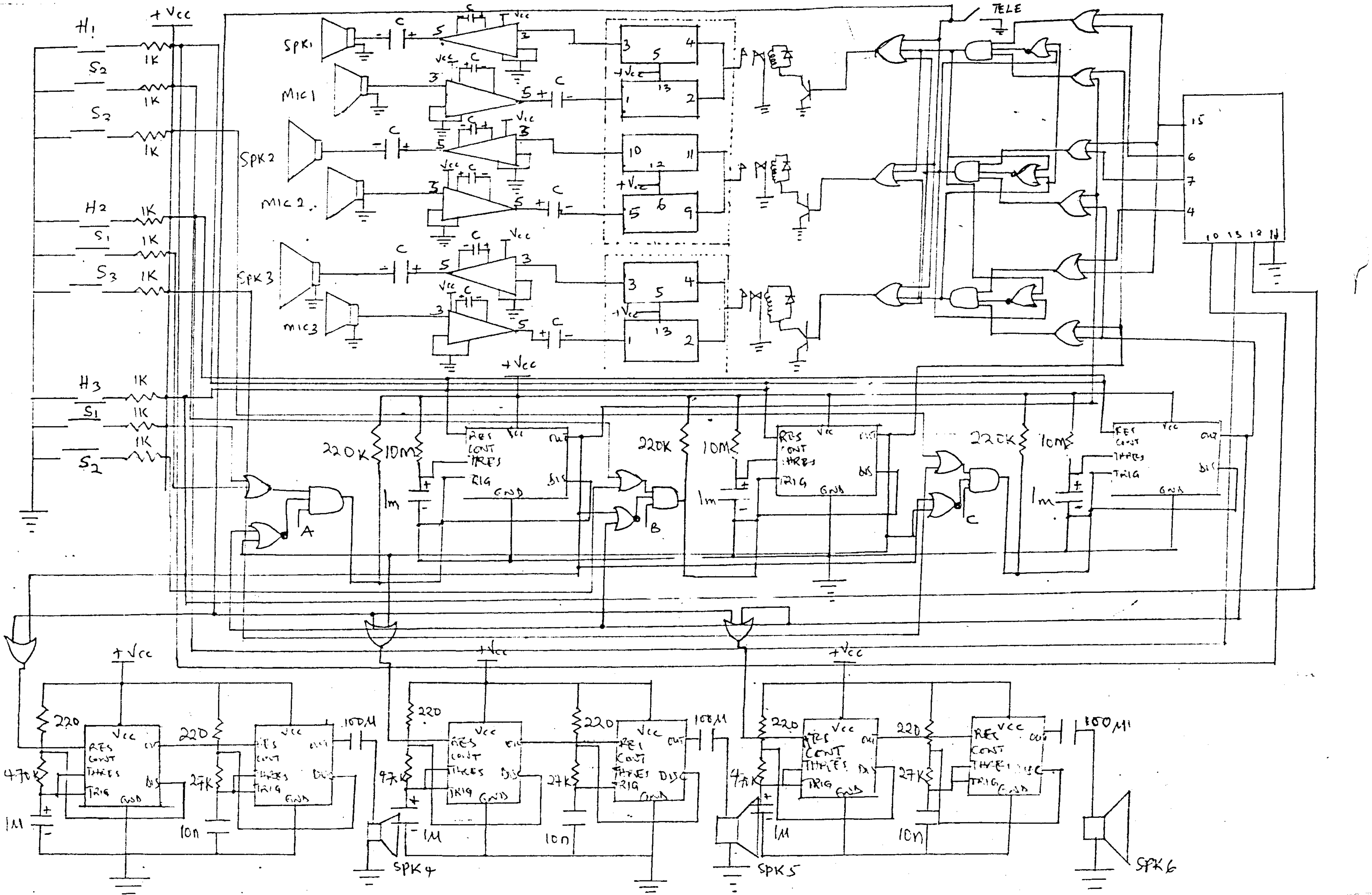
H1
S2
S3
H2
S1
S3
H3
S1
S2
1K
+Vcc
SPK1
MIC1
SPK2
MIC2
SPK3
MIC3
TELE
A
B
C
220K
10M
1m
RES
CONT
THRES
TRIG
GND
DIS
OUT
Vcc
220
470K
47K
27K
10n
1M
100M
SPK4
SPK5
SPK6