DESIGN AND CONSTRUCTION OF A PUBLIC ADDRESS SYSTEM (200W) STEREO

BY

ELIJAH OLAKUNLE

97/5984EE

DEPARTMENT OF ELECTRICAL/COMPUTER ENGINEERING, SCHOOL OF ENGINEERING AND ENGINEERING TECHNOLOGY.FEDERAL UNIVERSITY OF TECHNOLOGY MINNA, NIGER STATE.

SEPTEMBER, 2003

DECLARATION

I, Elijah Olakunle, 97/5984EE hereby solely declare that this project work "DESIGN AND CONSTRUCTION OF PUBLIC ADDRESS SYSTEM (200W) STEREO" is an original concept produced as a result of my personal effort.

All information derived from published work used in this project has been dully acknowledged.

ELIJAH OLAKUNLE

21-10-2003

DATE

CERTIFICATION

This is to verify that this project titled "DESIGN AND CONSTRUCTION OF PUBLIC ADDRESS SYSTEM (200W) STEREO" was carried out by ELIJAH OLAKUNLE for the award of Bachelor of Engineering in Electrical/ Computer Engineering F.U.T Minna.

....
Engr. Jacob Tsado
(Project supervisor)

....
Date

....
Engr. M.N. Nwohu
(Head of Department)

13/08/04
....
Date

....
External Examiner

....
Date

DEDICATION

This project is dedicated to God almighty that has been the source of my inspiration, the glory and the lifter of my head. I also dedicate this project to my parents.

ACKNOWLEDGEMENT

With a heart of appreciation to God, I give him the glory for giving me the privilege to start and complete my course.

I am grateful to my parents Mr.& Mrs. O. Elijah and siblings for their financial and moral support through out my stay on campus.

My special thanks goes to my project supervisor Engr. J Tsado and my project coordinator Engr. M.S. Ahmed for their examination and encouragement during the construction of this project. I also extend my appreciation to my former H.O.D Dr. Y. A. Adediran and my lecturers that impacted a wealth of their knowledge on me.

Finally, I acknowledge the support and advise of my colleagues especially Adi Temgu, Okechukwu Igwe and Ukwuoma Pious.

ABSTRACT

The Public Address System (200W) stereo is design to provide an affordable system with maximum efficiency and at minimum cost to the Nigerian Market.

PAS (200W) stereo is a multi-system that deals with recording and reproduction of audio signals and is based on Analogue Technology.

The system is made up of four distinguishing features, which are: Disco-mixer, Amplifier, Speakers, and Tape player/recorder. It is basically hardware without any software support. A thorough analysis of the system, which is contained in the chapters, allows for easy understanding of the work done.

LIST OF FIGURES

TABLE OF CONTENT

CHAPTER ONE

GENERAL INTRODUCTION

1.1 INTRODUCTION

In simple terms, the public Address-200W Stereo (PAS-200W) is an electronic system with incorporated units that deals with recording and reproduction, which is based on Analogue Technology.

The public Address system obviously means having a system of loudspeakers placed so that output of microphones or other sources can be fed to a relatively large number of people. It is though, convenient to realize that there are three finite different applications:

1. The audience is remote from the microphones. A typical example is that of a sporting event where the commentator is usually in a small room some distance from the loudspeakers.
2. The audience and the microphones are in the same room or hall. In the other words, the audience or at least much of it can hear the person speaking but probably with a degree of difficulty. What is wanted is reinforcement.
3. The amplification of music at a rock concert and similar events. This is very important in the entertainment world.

This project is carried out based on the second application of P.A listed above. The design and construction of the project is therefore influenced by its application.

The layout of the design analysis, procedure of construction, testing, results and discussion have been carefully done to allow for high comprehension and easy understanding of the project.

1.2 AIMS & OBJECTIVES

Stereo amplifiers, mixers, speakers, tape recorders and players have under gone many years of evolution and development. These products, which are sold and used today in homes, religious houses, entertainment industry, and theatres, are very sophisticated indeed. Their reliability is better than ever before, performance too has increased by leaps and bounds, as it has ease of use for the viewer. Despite all of this the price is still affordable.

The pace of development shows no sign of slackening but it is unfortunate that the cost of assembling these appliances to achieve a good public address system in our society works against the domestic users. This is due to the unfavorable economic situation, especially high cost of imported goods.

Hence, the aim and objective of this project is to produce a prototype P.A system which will integrate the following features:

- Disco Mixer;
- Amplifier;
- Speakers;
- Tape Player and Recorder.

This aimed at achieving the following:

1. Reduce the cost of acquisition of a good P.A system i.e. a P.A system that is free of noise, low and clear.
2. Reduce the size of the P.A system.
3. Eliminate too many interconnection of system hence eliminating unwanted interference and wire complexity.
4. Produce a high frequency sound system that can be easily marketable.

5. Produce a local P.A system that stands a chance of replacing imported ones.

6. Allows for low cost of maintenance of combination of systems put together.

1.3 LITERATURE REVIEW

The evolution of recording and reproduction of audio signals started with the invention of the phonograph by T.A Edison. Since then research and efforts to improve techniques have been determined by the ultimate aim of recording and reproducing an audio signal faithfully, i.e. without introducing distortion of noise of any form.

With the introduction of the gramophone, a disc phonograph, in 1893 by P. Berliner, the original form of our present record war born. This model could produce a much better sound and could also be reproduced easily.

Around 1925, electric recording started, but an acoustic method was still used in the sound reproduction system, where the sound was generated by a membrane and horn, mechanically coupled to the needle in the groove in playback. When recording the sound picked up by was transformed by horn and membrane into a vibration and coupled directly to a needle which cut the groove onto the disc.

Further developments such as electric crystal pick-up and, in the 1930s, broadcast AM audio station made the S.P (standard playing 78rpm record) popular. In 1948 with the development, popularly increased as Long-playing (LP) and Extended play (EP) with improvement in record sound quality were introduced. At same time, companies like General Electric and Pickering developed the lightweight pick-up cartridge, with only a few grams of stylus pressure

The true short of progress towards the ultimate aim of faithful recording and reproduction of audio signals was the introduction of stereo recording in 1956. This began a race between manufactures to produce stereo reproduction tape recorder, originally for industrial master use. However, the race led to a simplification of techniques, which in turn, led to development of equipment for domestic use.

Broadcast radio began its move from AM to FM with subsequent improvement of sound quality, and in the early 1960s, stereo FM became a reality. In the same period Philips developed the compact cassette recorder, which would eventually conquer the world.

The three basic media available in the early 1960s: tape, record and FM broadcast, were all analog media. Development since then includes:

i. Development in turn tables;

ii. Development in tape recorders with particular attention paid on the recording and reproduction heads, recording tape as well as tape path drive mechanism. Also the introduction of compression/expansion system such as Duly, UBX etc. improved the available signal-to-noise ratios.

Due to bulky nature of open real tape recorders, the invention of compact cassette recorder began in 1963, which made it possible for millions of people to enjoy recording and playing back music with reasonable tone quality and easy operation. The impact of the compact cassette was enormous and tape recorders for recording and playing back these cassettes became quite indispensable for music lovers, and for those who use the cassette recorders for a myriad of purposes such as taking notes for study, recording speeches, dictation for 'talking letters' and for hundreds of other applications.

LIMITATION OF ANALOG AUDIO RECORDING

Despite the spectacular evaluation of techniques and the improvements in equipment, by the end of the 1970s the industry had almost reached the level above which few further improvements could be performed without increasing dramatically the price of the equipment. This was because quality, dynamic range, distortion (in its broadcast sense) is all determined by the characteristics of the medium used (record, tape, broadcast) and by the processing equipment. Analogue reproduction techniques had just reached the limits of their characteristics.

In summary, in spite of all the spectacular improvements in analogue technology, it is clear that the original dynamic range is still seriously affected in the analog reproduction chain. Similar limits to other factors affecting the system: frequency response, signal-to-noise ratio, distortion, etc. exist simply due to the analogue processes involved. These reasons however prompted manufactures to turn to digital techniques for radio production.

1.3 PROJECT OUTLINE

The project has been outlined carefully chapter by chapter in which each chapter explains the concise topic in detail.

Chapter one gives the general introduction of the whole project work. It also contains the literature review, project objective.

Chapter two handles the hardware design of the project. In this chapter, the system's design is done in modules for easy debugging of faults. Each design of these modules formed the sections of this chapter.

Chapter three takes a look at the implementation aspect of this project. In this chapter, the modular designs in chapter two are interfaced and built on a veroboard.

Chapter four takes care of the testing, result and discussion of the project work.

Chapter five, which is the last chapter highlights the problem faced, made some recommendations and also conclusion of this project.

A maximum measure of comprehensibility was ensured throughout the project. This is indicated by the pattern of presentation and consistent system of notation. This will ensure maximum reader satisfaction.

CHAPTER TWO

SYSTEM DESIGN

2.1 INTRODUCTION

The design of this PAS-200W stereo is purely hardware and there is no software support for it. It was designed with electronic chips and components interfaced in functional parts to work as a system. This chapter gives look on how each block of the system's block diagram was designed and analysed, description of chips used and their features were also given.

Generally, the system is made up of op-amps, combination of transistors and other few passive components for specific purposes.

2.2 OPERATIONAL AMPLIFIERS

An operational amplifier, op-amp, is in essence a complex integrated circuit consisting of transistors and resistors. Because of its uniform and clearly definable mode of operation, it makes circuit design much easier and it must be said, normally cheaper.

Terminals of an Op-Amp

The circuit symbol of an op-amp is shown in fig. 2.0. The two vertical lines, marked +UB are -UB are the terminals for the symmetrical supply voltage.

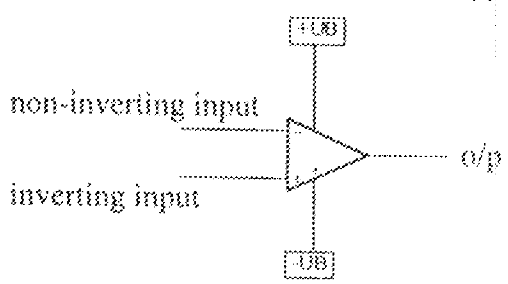


Fig. 2.00 terminals of an op-amp

A symmetrical ± 15v line powers the op-amp. This enables the control (0V) potential to be the earth so that the amplifier swings in either the positive or negative direction.

The op-amps used in this project are the TL074 and Tl072 with pin configuration shown below in fig2.01

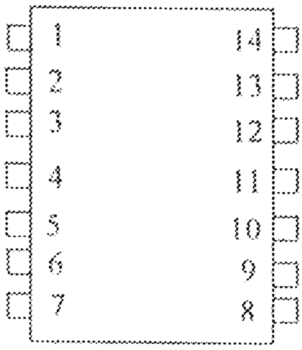


Fig2.01a: TL074

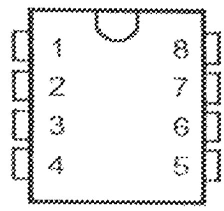


fig2.01B: TL072

The TL074 and TL072 op-amp ICs were used because of their excellent properties listed below:

1. Low distortion
2. Low noise
3. High-impedance JFET inputs ($10^{12}\Omega$)
4. Short-circuit-proof, bipolar output stage.

2.3 SECTION 1: DISCOMIXER

By the term disco mixer or sound mixer, it means equipment designed to accept the electrical output from microphones and other sources such as tape machines and CD players and it allows an operator to combine these in ways which are artistically and technically satisfactory.

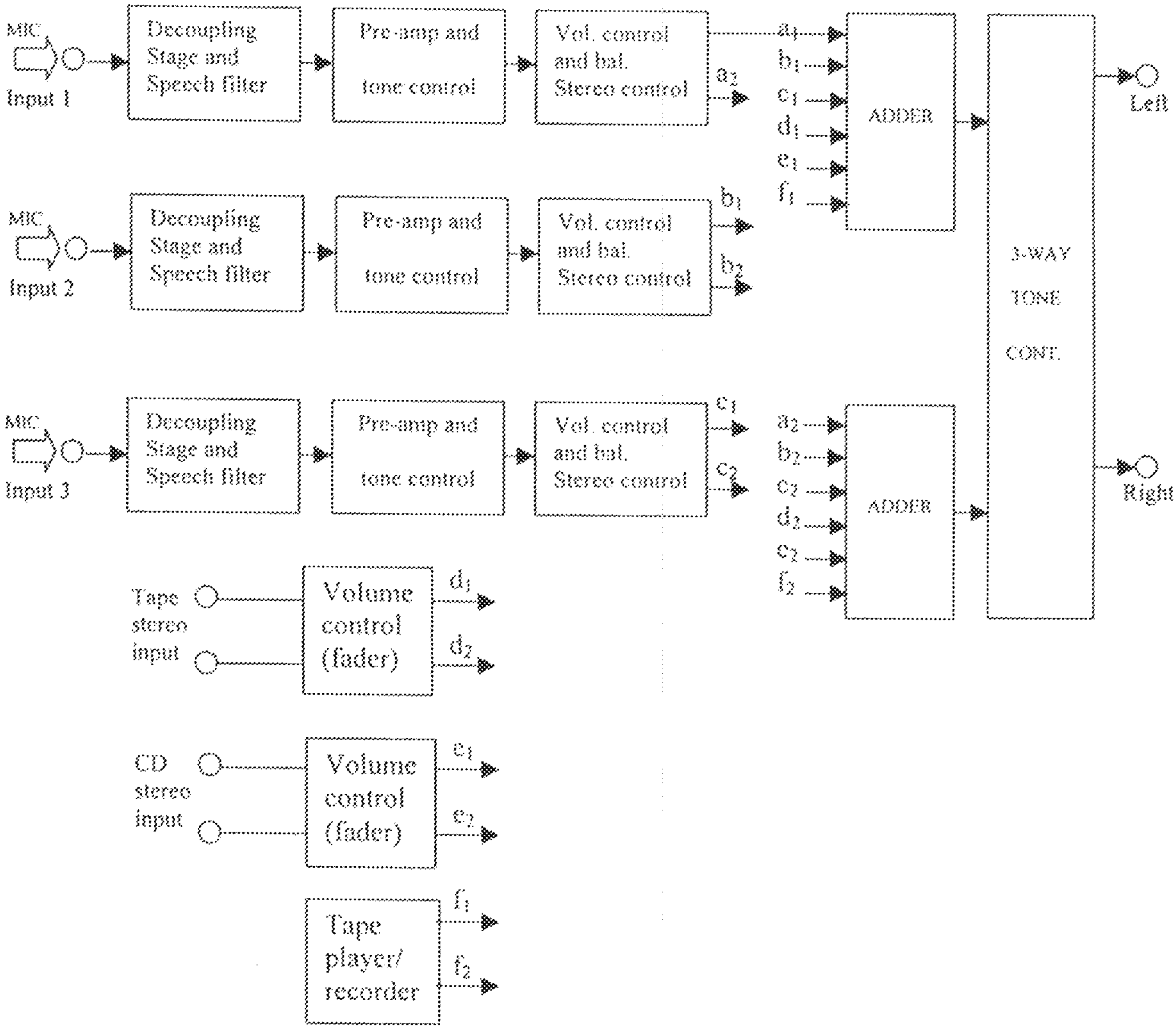


Fig2.02 Block diagram of the Disco Mixer

A disco mixer is invariably provided with stereo inputs as shown in fig2.2. The tape and CD inputs do not need a preamplifier since the relevant units provide high enough output. The volume controls are of course stereo potentiometers. The modular consists of the following functional blocks:

1. 3-channel stereo mixers
2. CD inputs
3. Tape inputs
4. Adder
5. Stereo tone control (3-way)

2.3.1 3-CHANNEL STEREO MIXER

The mixer helps for blending of the various sound sources. Each line has its mic input, which feeds the decoupling stage, speech filter, preamp and tone control, volume control and stereo balance sequentially.

Decoupling Stage

The decoupling stage is designed to remove the signal earth from the circuit earth. This reduces or eliminates hum, which are caused by differing earth potential.

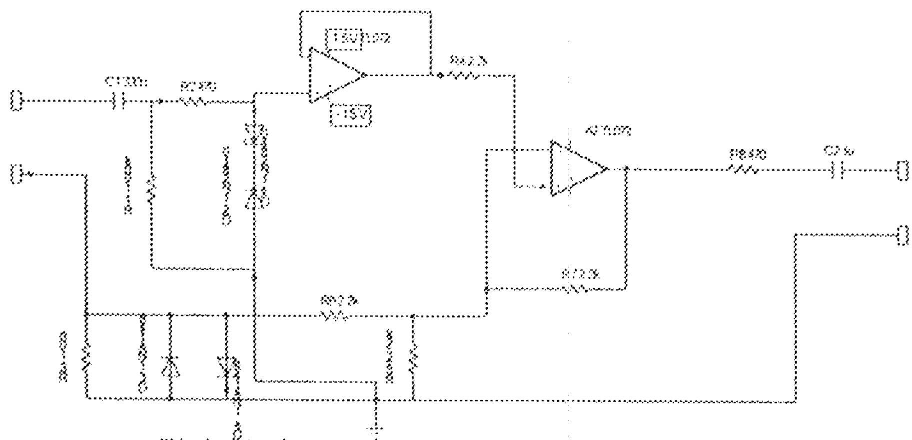


Fig2.03 circuit diagram of the Decoupling stage

From the circuit in fig2.03 the op-amp A1 is an impedance converter with a decoupling network. 330nF and 100K forms high pass filter with cut off frequency, fc,:

Fc = 1/2 ∏ RC = 1/2 ∏(100K)330n = 4.2Hz

This decouples DC component of the input signal. The hum potential, which is normally dropped across the interconnecting wire, is dropped across R3 (100Ω). Diodes D3 and D4 (1n4148) limits the maximum permissible hum potential to ± 700mV.The second op-amp operates as a differential voltage from the input signal on the earth line. This converts the input signal referred to the signal line earth into a signal that is referred to the circuit earth.

$$\text{Gain} = \frac{R7 + R5}{R5} = \frac{2k2 + 2k2}{2k2} = 2 \approx 6dB$$

Speech Filter

The speech filter in fig2.4 provides a slight pre-emphasis of the presence range of about 2000-6000Hz. The filter contains an additional high pass section, which provides a slight attenuation of the fundamental frequencies of the speech. This section is particularly useful in combating the muffled murmuring so often present in the reproduction of public address installation.

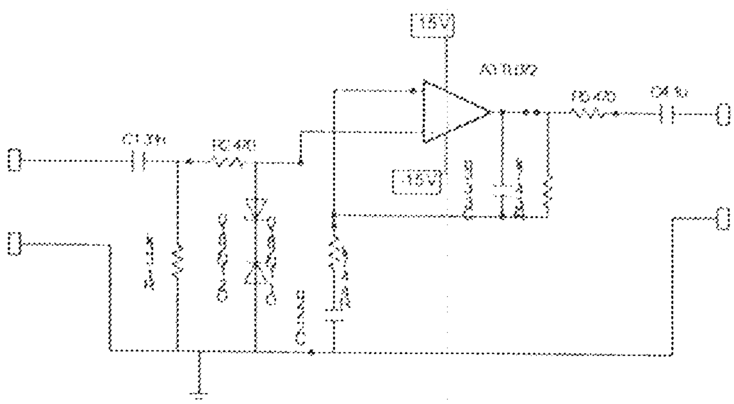


Fig 2.04 circuit diagram of speech filter

The R1-C1 network serves as decoupling element for direct-voltages and also as high-pass-section. Its cut-off frequency is around 150Hz.

Fc=1/(2[]R1C1) = 1/(2[]*33K*33n) = 146.15Hz.

The (4K7), R1 resistors and C2, (22n), (3n3), CB, capacitors provides frequency-dependent feedback. At low frequency, capacitor C2 has a high reactance and the whole signal is fed back having a unity feedback.

$$Gain = \frac{4k7 + \alpha}{\alpha} = 1$$

At high frequency in the presence range, the capacitor C2 has a low reactance hence the feedback is determined by potential divider R3 and R4. The amplification is

$$Gain = \frac{4k7 + 4k7}{4k7} = 2 = 6dB$$

To determine the bandwidth of the circuit see appendix A

Pre-Amp & Tone Control.

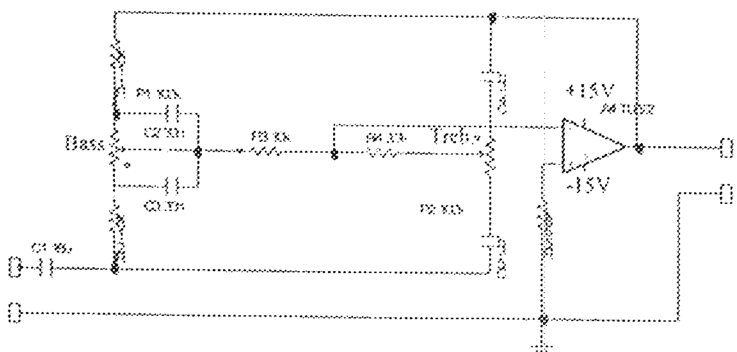


Fig2.05 circuit diagram of pre-amp & tone control

The pre-amp & tone control circuitry forms the major part of the mixer. This sound mixer allows the operator to combine the output from the microphone in ways which are artistically and technically satisfactory.

Bass section:

F_C = 1/(2[]RC)

= 1/(2*[]*10K*330n) =530Hz

Max. Gain = 110k/(10k) = 11 ≈ 20dB

Treble section:

F_C =1/(2[]RC)

= 1/(2*[]*3.3k*3n) =16kHz

Max. gain = 103.3k/3.3k = 33 ≈30dB

Stereo Balance And Volume Control.

The signal from the pre-amp tone control is separated into left-hand and right-hand component. The signal may be applied to the left-hand or right-hand adder or both with the aid of a stereo balance control, which was included in the fader control section, as shown in fig2.06

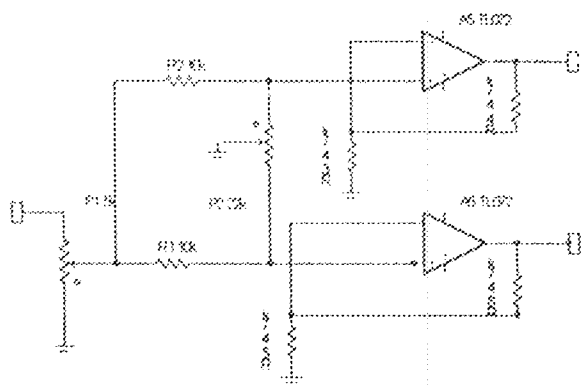


Fig 2.06 circuit diagram of combined stereo bal. and volume control.

The op-amps raise the level of the relevant signal x2 to offset the attenuation of the balance control and they ensure low output impedance.

$$Gain = \frac{4k7 + 4k7}{4k7} = 2 = 6dB$$

2.3.2 CD CHANNEL INPUT

The output of any CD is fed into the P.A.S via the A.V stereo jack plug provided in the front panel. With the use of a potentiometer, which is connected to the input signal, the volume is controlled. Thereafter, it is fed into the adder circuit in section 2.3.4 as one of the input.

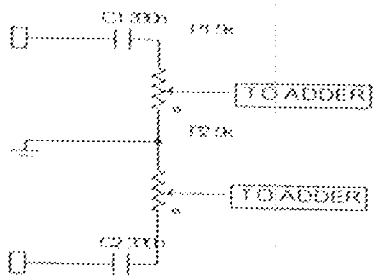


Fig2.07 Circuit diagram of CD, Tape fader control

2.3.3 TAPE CHANNEL INPUT

The output of any tape is fed back into the P.A.S via the AV stereo jack plug provided in the front panel. The same network is used as shown in section 2.3.2 fig2.7

2.3.4 ADDER

The Adder is a summing amplifier that adds several signals. The signals from the 3-cahnnel mic lines, tape or CD and also from the tape player are added together using the adder circuit shown below:

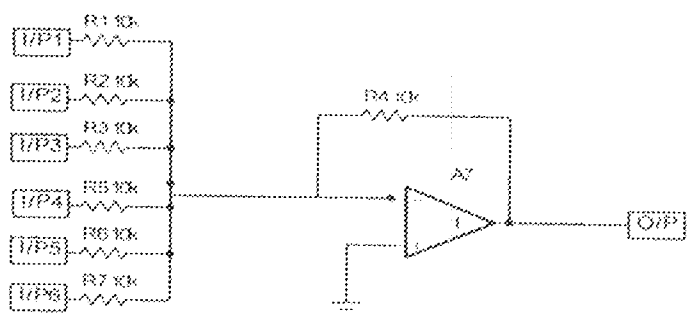


Fig2.08 circuit diagram of adder Circuit

Output current: I (o/p) = I1 + I2 + I3 + I4 + I5 + I5 + I6 = -Vo/Rf

$$\Rightarrow \qquad V_0 = -R_f\left(\frac{V1+V2+V3+V4+V5+V6}{R}\right)$$

$$V_o = \frac{-10k}{10k}\left(V1+V2+V3+V4+V5+V6\right)$$

$$= -(V1 + V2 + V3 + V4 + V5 + V6)$$

The other channel has the same analysis. To determine the bandwidth see appendix.

2.3.5 STEREO TONE CONTROL

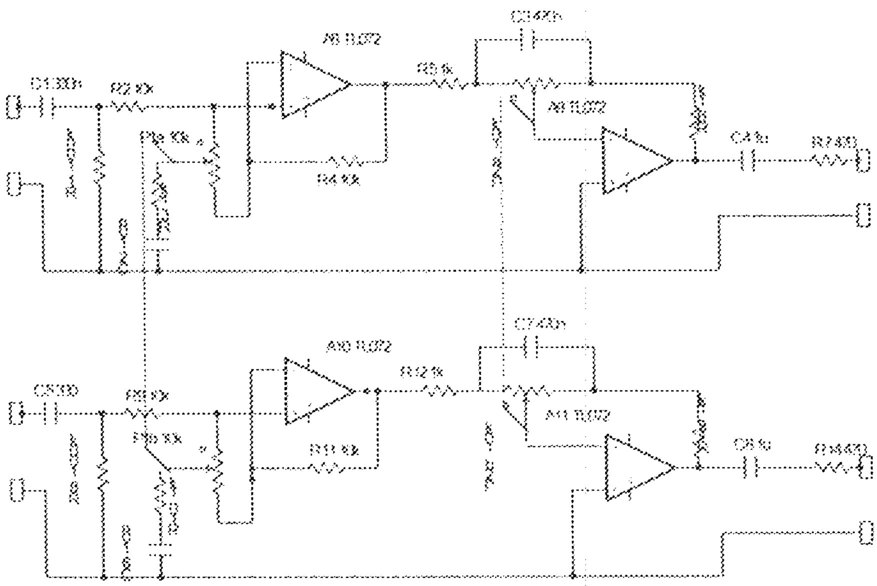


Fig2.09a Circuit diagram of stereo tone control

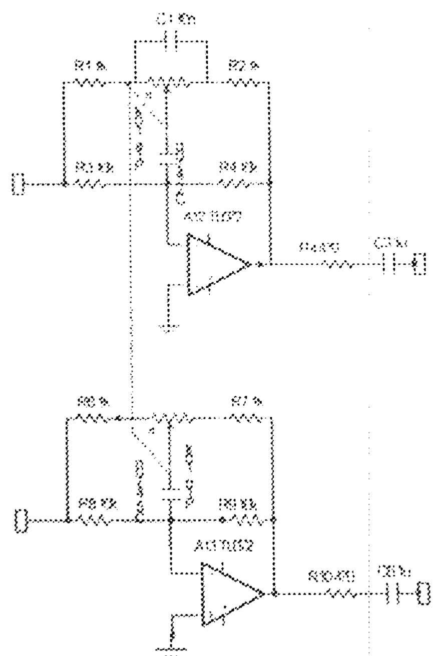


Fig2.09b circuit diagram of midrange tone control

The stereo tone control is a straightforward control for adjusting high- frequency (treble), mid-range frequency and low frequency (bass) reproduction. The high frequency and low frequency control makes up one circuit, which is connected in series with the mid-range frequency as shown in the block diagram below. This makes it a three-way tone control.

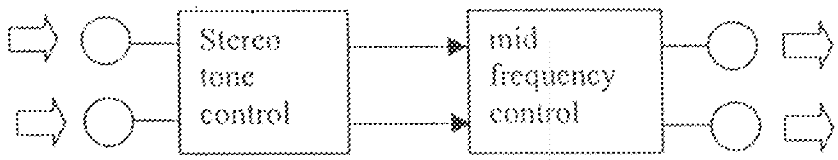


Fig2.10 Block diagram of three-way tone control

In the high frequency and low-frequency control circuit, only the capacitors C2, C3 and C6, C7 are needed for the frequency control.

FOR HIGH FREQUENCY,

When potentiometer, P1 is turned fully anticlockwise, R2 and C2 form a low-pass section at the inverting input of the op-amp. Resistor R3 provides the maximum attenuation in the highest frequency,

R2 C2, fc2 = 1/2[]R2C2 = 1/2[](10K)(10n) = 1.5kHz

When P1 is turned fully clockwise, R4 and C2 form a low-pass section in the feed back loop that produces a completely different frequency response in which the high frequencies are lifted.

fc2 = 1/2[]R4C2 = 1/2[](10K)(10n) = 1.5kHz

Gain = 10K +1K/1K = 11 ≈ 20dB

FOR LOW- FREQUENCY (BASS CONTROL)

It is provided by A2. If we ignore capacitor C3, the stage becomes a standard inverting amplifier when potentiometer P2a is turned fully anticlockwise; it forms a series network with resistor R5. The amplification of the stage will then be below unity.

When P2a is turned fully clockwise, it forms a series network with resistor R6. The amplification is then greater than unity.

Amplification = (R6 + P2a)/R5 = (10K + 1K)/1K = 11 ≈ 20dB

When the upper of P2a is anywhere between the extreme positions, one part of the potentiometer resistance is in series with R5, and the other with R6. Capacitor C3 short-circuits P2a at mid range and treble frequency, so that the potentiometers act on bass frequency only.

FOR MID RANGE FREQUENCY CONTROL,

From fig2.7.3, adjusting P2a determines how much of the resistance of this control is in series with R1 and how much with R2. Depending on the setting of the control, inverting op-amp A1 provides a gain of between -20dB and +20dB. At high frequencies, capacitor C3 has very low impedance and short-circuits the potentiometer. This means that the circuit functions only as bass and mid range are concerned. However, capacitor C2 and resistors R3 and R4 prevent the circuit acting on the bass frequencies. This is because at these frequencies, C2 has high impedance, so that so that R3 and R4 determine the amount of feedback instead of the potentiometer. The overall result is that the current controls mid range frequencies only.

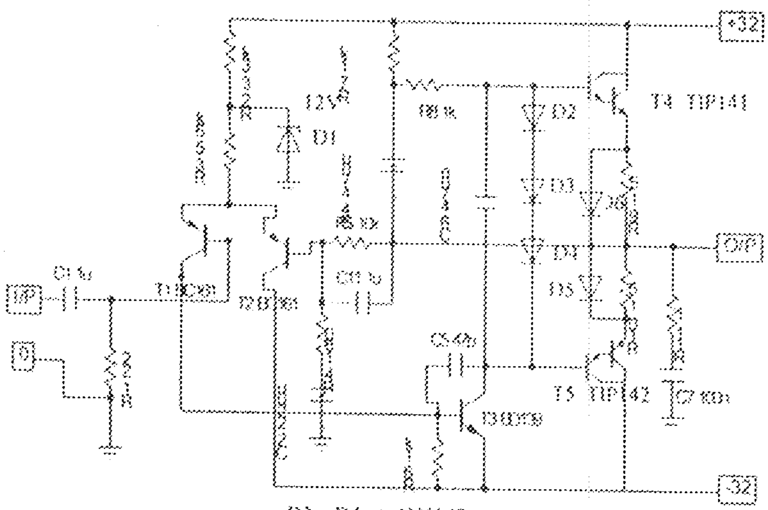
2.4 SECTION 2: OUTPUT & LEVEL DISPLAY.

This section consist of the following:

i. Right & left hand 100W amplifier each

ii. Drive level indicator for R and L channel. This section is made up of two modules, with module one consisting of the right and left channel amplifiers and the other module consisting of the right and left channel level indicators

I. 100W Amplifier



D2 – D4 => 1N4148

D5 – D6 => 6A1

5

Fig2.11: circuit diagram 100W amplifier

The amplifier is a general-purpose output amplifier. It is built as a stereo version with one on the right channel and another on the left channel.

Single Stereo

Power output	-	100W	200W
Supply voltage	-	±32V	±32V
Transformer voltage	-	2*24V	2*24V
Transformer rating	-	150W	300W
Rectifier current	-	3.3A	6.6A
Capacitor rating	-	40V	40V
Heat sink	-	1.6/KW	0.8/KW

For the purpose of stereo output, Transfer rating – 300W,

Rectifier current- 6A

From fig2.10, transistor T1 and T2 form a differential amplifier. The potential at the base of T2 is compared with the input voltage (at the base of T1). When the base voltage of T2 is higher than T1, T1 is ON since the potential across its base-emitter junction is about 700mV where as that of T2 is slightly lower (so that T2 cannot conduct). The base of T3 is then driven by T1, so that T3 is also ON and connects the bases of T4 and T5 to ground. The output of the circuit then drops. Since the output is dropped across potential divider R4-R5 of T2, it drives the base of . Consequently the base potential of T2, which at the onset was slightly higher than that of T1, drops until the base voltages of the differential amplifier are equal. The ratio of R4 and R5 determines the amplification of the circuit. Diodes D1 and resistor R2 hold the voltage at the upper end of R3 stable at about 12V to ensure that any hum at the supply is suppressed. Diodes D2-D4 generate the requisite base bias voltage for the Darlington transistors. The value of R9 and R10 is relatively high, which makes setting the quiescent current unnecessary. Network R7-R8-C4 is a bootstrap current which ensures that when, for example, the base voltage of T4 rises to 10V, the output voltage of the current rises by same amount. Capacitor C4 raises the voltage at the junction of R7 and R8 also by 10V

HFE = 1000 Typ.

BV_{BEO} = 5V

V_{CC} = 32V

$V_M = V_{CC} - 5V = 27V$

$$I_{MAX} = \frac{27}{4} A$$

RL= 4Ω

$$P = \frac{6.75 \times 27}{2} = 91.25W$$

$$Iav = \frac{Vm}{\pi Rl} = \frac{27}{\pi \times 4} = 2.14A$$

$$Gain = \frac{R5}{R4} = \frac{10k}{180} \approx 55$$

It is calculated to give full power when the input signal is 500mV. —

II. DRIVE LEVEL INDICATOR

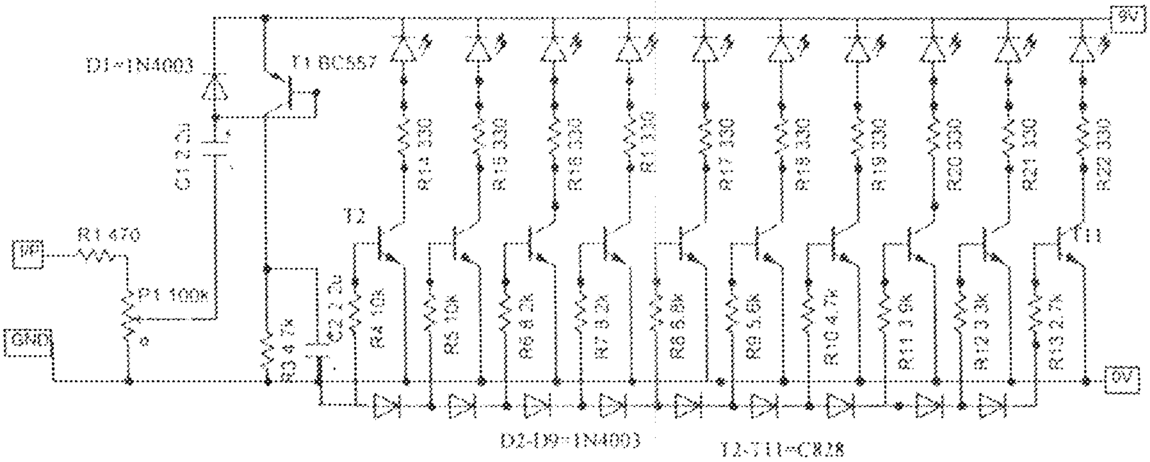


Fig2.12: circuit diagram of Drive level indicator

D1-D9 = IN4003

D10-D19 = LED

T1 = BC557

T2-T11 = C828, C829

The Drive Level Indicator circuit is a combination of LED array that acts as a VU Meter for visual monitoring of output level. Each channel consists of 10 LEDs arranged in the increasing order from left to right.

For maximum power display; VCE of T1 (BC557) ≥ 6.3V

Ic for each of the LED can be derived using the formula; $I_c = I_b * h_{FE}$

Resistors R14- R22 acts as limiting resistors to the diodes.

2.5 SECTION 3 SPEAKER CONNECTION

This section consists of the following:

i. Three-way network unit

ii. Woofer speaker, Midrange speaker, Tweeter speaker.

The 3-way network unit is a circuitry, which separates the o/p signal of the amplifiers with the bass frequency, treble and midrange frequencies. Hence the separate signals are therefore fed into the respective speakers. As a result of this the life span of the speakers e.g. tweeter is extended as bass frequencies are prevented from entering with it.

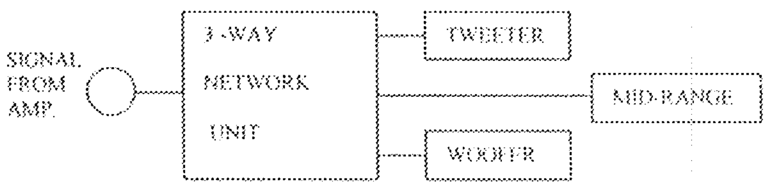


Fig 2.13 Block diagram of speaker connection

2.6 SECTION 4: TAPE REC0RDER / PLAYER

A small tape player (walkman) was incorporated into the system. This was designed to perform the following operations:

I. Receiving of F.M stereo

II. Playing of audiocassette

III. In- to –in recording from mixer

For the purpose of in-to-in recording from the mixers of the two channels an adder was designed and constructed which allows the combined signal to be fed into the recorder of the tape. A two input inverting adder was used for this purpose. Refer to Fig 2.08.

2.7 POWER SUPPLY

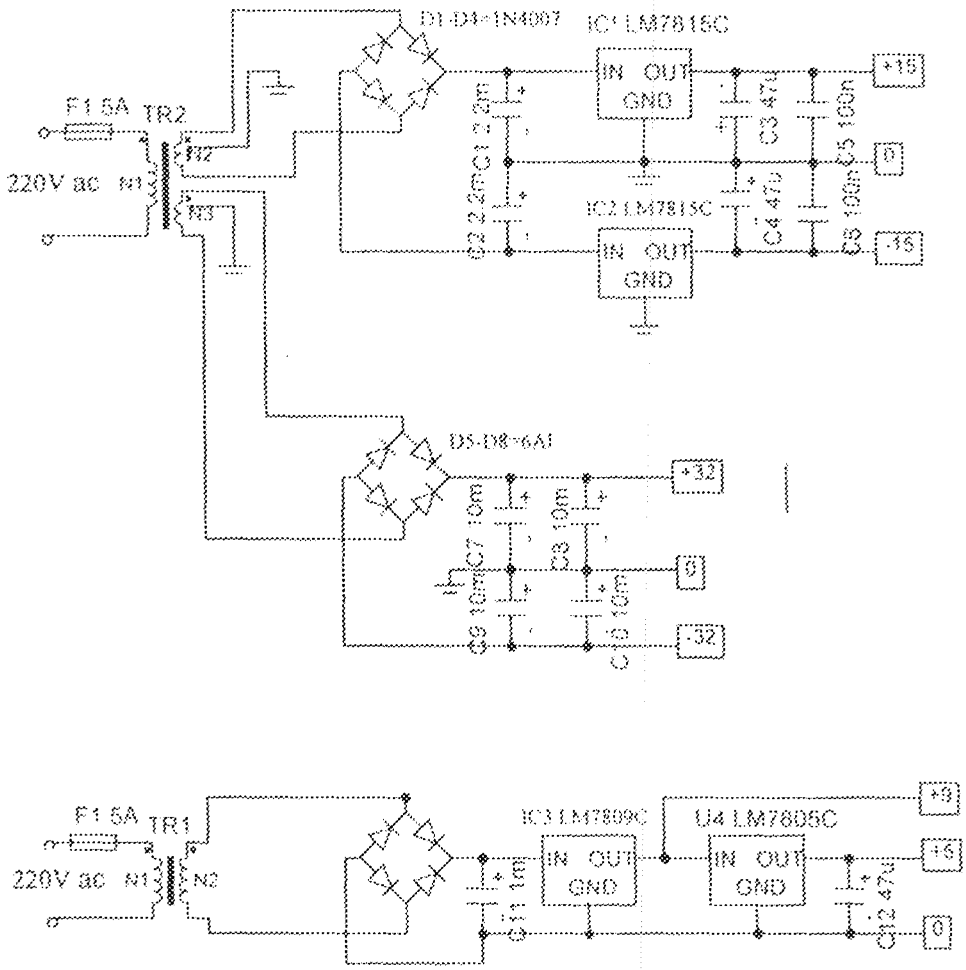


FIG 2.14a,b,c circuit diagram of Power supply.

The power section consists of various outputs as shown in fig2.14. The mains supply voltage used in the design was 220v r.m.s, 50Hz from NEPA.

+32V - 0 - -32V Power Supply Unit.

A transformer with multiple output center tapped was used. Hence on the former, a center-tapped 220/24V was used. A bridge rectifier was used to rectify the AC voltage to DC voltage and was filtered by a 20000uf /40V capacitor on each side of the supply rail. The filtered voltage was later connected to the amplifier circuit.

The transformer was chosen based on the fact that its output current should be 10A, which is enough to drive the whole circuit.

The r.m.s. Voltage was calculated based on the formula below:

$$Vr.m.s = \frac{Vpeak}{\sqrt{2}}$$

$$Vpeak = 32V$$

$$Vr.m.s = \frac{32}{\sqrt{2}} = 22.63$$

to allow for voltage drop, 24V center tapped was chosen.

The rectifier was chosen based on the following properties/ratings

- Silicon rectifier
- Peak reverse voltage (PRV) = 100V
- Average maximum forward current = 6A

The capacitor rating was 20,000uF, 40V, 1.5 times above 24Vrms.

The +15 -0- -15V Power supply unit

The centre tapped secondary output, which was mounted on the same transformer with the 24V centre tapped. A full bridge rectifier was used to rectifier the AC voltage to DC, which was filtered by a 2,200uF, 35V on both rail supply. The output of the filtered capacitor was passed through the IC regulator to give a regulated DC voltage of +15V and -15V, the capacitor 47uF and 100nF were connected to the output of the regulators to reduce ripples. The circuit diagram is shown in fig2.14b

Rms voltage for secondary = 15V

$V_p = \sqrt{2}V_{rms} = \sqrt{2} * 15 = 21.2V_p$

$V_{dc} = 2V_p/\Pi$; for a bridge rectifier.

$= 2*21.2V/\Pi = 13.5V$

Diodes of PIV rating greater than 21.2V and with a forward current of 1A where chosen for the rectification of the AC voltage.

For the capacitor filter, the ripple voltage of the rectifier output was estimated to be about 5V.

V = ripple voltage

V = 5V

$\Delta t = 1/2t$ = time between peaks of the rectified input waveform of the capacitor.

$\Delta t = 1(2*20) = 10ms$

$I = C\Delta V/\Delta t$ where I = 1A

$C = 1*0.01/5 = 0.002F$

Therefore C = 2000uF

In reality, a capacitor of 2200uF was used. The voltage rating of the filter capacitor was chosen to be one half the peak voltage; hence a filter capacitor of 2,200uF, 35V was used on both supply rails. The IC regulator with fixed voltage of +15V and -15V, 1A each was used.

5V, 9V supply

For the +5V, +9V power supply unit, the same design procedure was carried out as discussed in section 2.72. For the output voltage, IC regulator 7809 and 7805 were used to get the fixed output voltages respectively.

CHAPTER THREE

CONTRUCTION PROCEDURE

The construction of PAS-2OOW Stereo involved careful arrangement of the various components in each section as specified by design and inter connected for easy troubleshooting and maintenance.

The tools and equipment used for the construction are listed below.

3.1 TOOLS AND EQUIPMENT

a. Side cutter

b. Long nose pliers

c. Picker

d. Soldering iron

e. Sucker

f. Digital multi-meter (DMM)

g. Signer generator

h. High impedance miniature speaker

i. Scourge

j. DC regulated power supply

3.2 CONSTRUCTION DETAILS

A temporary workbench was setup and it had the following components and Equipments listed above in section 3.1. For each stage of the construction, the components were carefully selected and collected together. These components were properly arranged on the new board following an appropriate layout. The components were later soldered with the use of soldering iron and lead with each other.

3.2.1 CONSTRUCTION OF THE POWER SUPPLY UNIT

The power supply section was constructed as a module with different outlets as specified in the design diagram. The power supply units is i.e. + 32v – 0 – -32v, +15 – 0 – -15v and the +9, +5v, had their bridge rectifier mounted first on the Vero board, the filtering capacitor were also connected to it. The +32v – 0 – -32v output w ere connected to a fast blowing fuse of 5A, while the +15 – 0 – -15v were connected to IC Regulator 7815 and 7915 respectively. The DC voltage +9, and +5 were achieved with the 7809, 7805 regulators. Finally, connecting buses with socket were used to run the power supply to various section of the system.

3.2.2 CONSTRUCTION OF THE DISCOMIXER

The Disco mixer was grouped into 5 modules. The first three modules consist of the 3 channel stereo mixer, the fourth module consists of the C.D and Tape input channels and the fifth module consists the adder and three tone control circuit.

The First Three Module

Each of these modules had the microphone jack plug mounted to the Vero board; this was connected to the input of the decoupling stages. The output of the speech filter was connected to the pre-amp and tore control. The output of the pre-amp and tone control was then connected to the stereo balance and volume control. This is illustrated in Fig. 2.2.

The Fourth Module

The fourth module had the AV jack plugs, stereo version mounted on the Vero board. The AV jack plugs were connected to the volume control (fader).

The Fifth Module

The fifth module had on it the adder circuit and the 3 tone control circuit. All the output from the mixer section i.e. output of the stereo balance and volume control were connected to the left and right adders via a bus wire. Each input of the adder had a resistor with the same value. The resistors were connected together to the input of the op-amp. The output of each of the op-amp i.e. the left and right channel was connected to the input of the 3 way stereo tone control.

The output of the three-way control serves as line out at the back panel of the system and also as the input of the power amplifier.

3.2.3 CONSTRUCTION OF THE AMPLIFIER

The 100W amplifiers both left and right were constructed on a single module. The power amplifier transistors were mounted directly on the veroboard, which served as a link between the veroboard and the heat sink. The heat sink in turn was mounted directly on the back panel of the system.

3.2.4 CONSTRUCTION OF THE DRIVE LEVEL INDICATOR.

The drive level display was constructed on a separate veroboard, which served as a module. The LED arrays were mounted on the veroboard vertically with equal distance and equal height from each other. It was then fixed on the perforated part of the front panel.

3.2.5 CONSTRUCTION OF THE SPEAKER.

The speaker section, which forms part of the case of the whole system, was located below the amplifier/mixer with a slab separating them. The tweeter, mid-range and woofer for each channel were linked to he 3-way network unit. The speaker was positioned appropriately to allow for maximum multidirectional propagation of sound.

3.2.6 CONSTRUCTION OF CASE

The case was made up of metal sheet, wood, and leather materials. The frame of the system was constructed with wooden box but the upper part (both front and back) panel was made of metal sheet. The plywood used for construction was ¾ inch plywood. The metal sheet was gauge 24 aluminum sheet. A good description of the case with its dimensions and parts are shown in fig 3.00

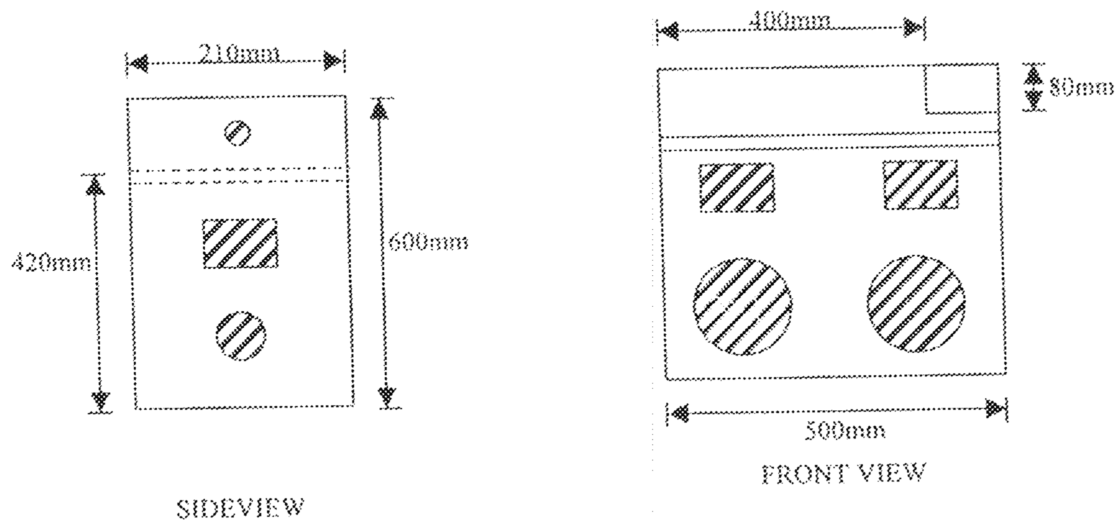
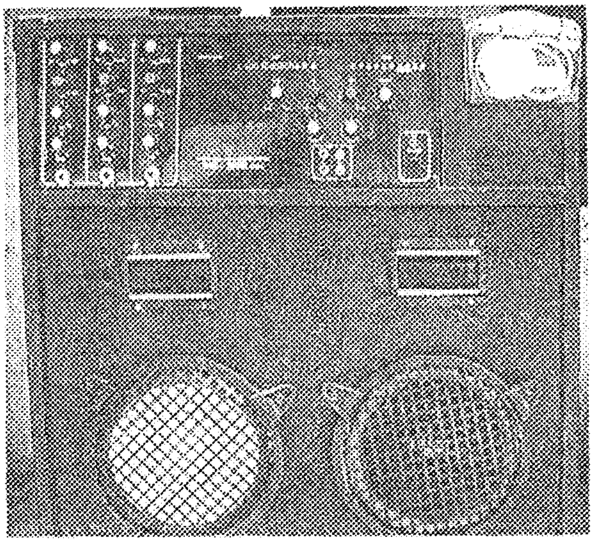
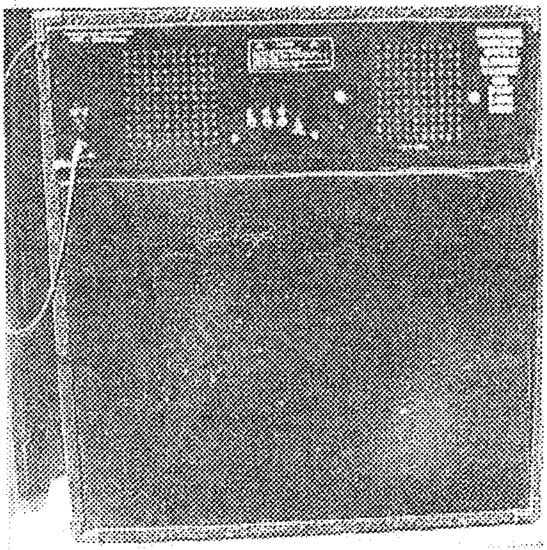
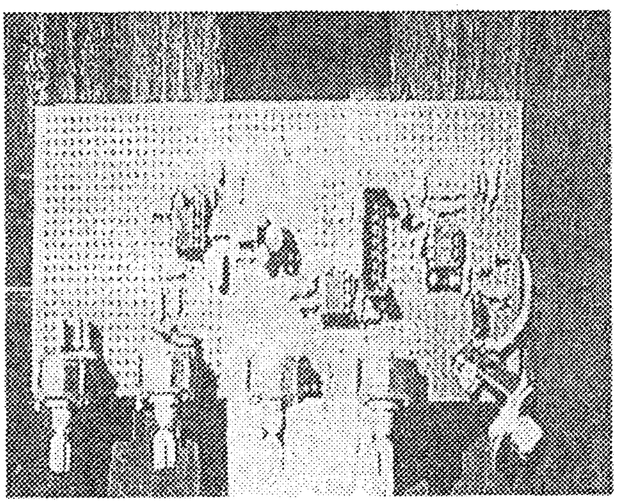


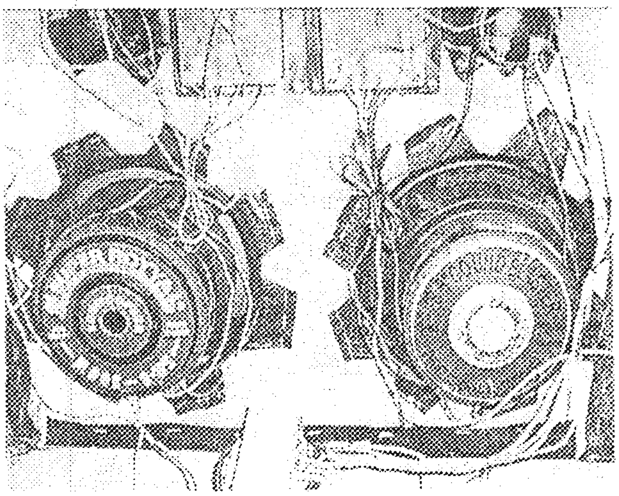
Fig. 3.00 Dimension of case

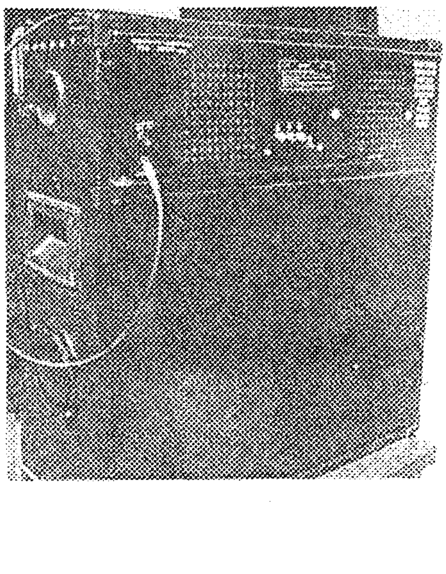
The picture of the pas-200W is shown for proper view of the case.











3.2.7 CONSTRUCTION PRECAUTIONS

The construction of each of the modules on the veroboard was carefully, neatly arranged and soldered. Each module is made up of socket able Bus for easy interfacing with other modules. This allows for maintenance of some of the modules with out affecting the others.

The modules were properly arranged in the case for easy maintenance and inspection of components in case of any fault. Even dissipation of heat within the system was put into consideration.

3.3 TESTING.

After the construction of each unit in a module, test was carried out using different methods and means. For the op-amp based circuits, they were powered with the +15v-0-15v and the output of the op-amp was measured using a volumeter. After that, a signal generator generating a frequency on 1KHz was fed into the inputs and the output was connected to the impedance miniature speaker. This method stated above was used for all the circuits in the Disco-mixer, stereo done control, and amplifier stage. Also to test the drive led indicator, the signal from signal generator was fed into the Drive display input terminals. The signal level was varied on the signal generator and the display LEDs were observed and monitored. The tape recorder was played and the treble, mid-range and bass control knob were turned clockwise and anticlockwise respectively and the response was observed from the speaker. Also, microphone with cord and cordless were used to test the mixer and the treble and bass controls were adjusted according. The responses from the speaker were also observed.

3.4 RESULTS AND DISCUSSION.

After due test were carried out as explained in section 4.1, the following results were observed as explained below;

At the output of the op-amp 0V was displaced on the DMM, which shows to some extent that the connection were properly done. When input signal were fed into each units and modules as the case may be, the response were loud and free of noise or noticeable oscillation. The responses were also amplified. After the testing of the three-way tone control by playing the tape recorder, the effect of the bass midrange and treble were very impressive. Also the Drive level indicator responded as desired when the volume control was increased and decreased as the case may be.

Further more the response of the P.A.S with the microphone was loud and clear. It was free of hum and unwanted noise.

3.5 sPROBLEMS FACED

During the construction of the project, certain components were difficult to get in the market. As a result components from shacks or scraps were used. Also there were components, which could not be found either in the market or from shacks and scraps. As a result of this I had to improvise by making use of what was available, though this did not deter the work. Necessary equipments such as Cathode Ray Oscilloscope (C R O), wattmeter, and spectrum analyzers were not at my disposal. A crude method of testing had to be adopted.

CHAPTER FOUR

CONCLUSION AND RECOMMENDATION

4.1 CONCLUSION

Construction of the P.A.S 200w stereo is a good and marketable prototype that integrates a Disco-mixer, Amplifier, Speakers and Tape Player/recorder with maximum cost effectiveness.

This actually reduces the cost of acquisition of a good public address system that is free of noise, loud and clear. It also eliminates the possibility of inter connection of too many wires that would have been as a result of separate units but all has been integrated into one and makes it portable.

Finally the aim and objective of the project was achieved and the project was successful. I therefore suggest it as a prototype that could be marketable.

4.2 RECOMMENDATION

I recommend that using a digital filter instead of analogue in the design and construction should modify the P.A.S 200w stereo. Also, the speaker unit should be detachable from the whole system without affecting the framework of the system. A C.D player could also be incorporated into the system to allow for multi-system. The position of the tape player/recorder could be placed in an enclosure to avoid been exposed to frictional damage.

It should be expanded from the 3-channel mixer to 6-channel mixer. Also, effect could be incorporated into the mixer so that it offers what any imported mixer offers. A line transformer also could be incorporated into the system.

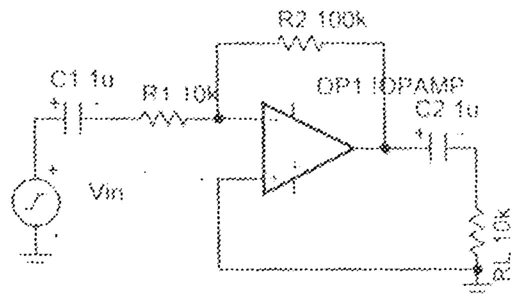
Finally parametric or graphical equalizer should also be incorporated to allow for more eradication of discolouration of sound.

REFERENCES

1. R. Southeimer, (2001), "Designing Audio Circuits", Singapore: Tech. Publications PTE CTD.
2. P.H. Wire field Hill, (1995), "The art of electronics", U.K., Cambridge University Press.
3. Philips, (2000) "ECG Replacement amide", U.S.A, SRUBS.
4. A.P. Machine, (1999), "Machine Electronic Principles", U.S.A.: Mc GRAW – Hill.
5. Luc. Baert, (1998), "Audio And Compact Disc Technology", Great Britain: Focal Press.

APPENDIX A

The formula below can be used to determine the gain, bandwidth and cutoff frequencies of the Op-Amp used in the various design of Chapter two.



Gain, $Av = \frac{-R1}{R2}$

Bandwidth, $f2_{(cl)} = \frac{f_{unity}}{R2/R+1}$

Cutoff frequency, $fc = \frac{1}{2\pi R1C1}$


4.5V