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Design and simulation of four channel audio mixer

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Abstract

Every part that takes signal from performer to consumer must be near perfect. As building a mansion starts by laying a block, so is building a complex network of electronics system by laying a component. Audio mixer is a device that accepts input signals, combines them giving an output were all the signals are heard without distortion or interference. The major problem in designing an audio mixer is noise which should be at minimum level. The large signals flowing into a small area of an audio mixer must be strictly kept apart to avoid cross talk. This is best done by virtual earth mixing bus. The designed mixer was grouped into modules for easier design approach, and Computer Aided Design (CAD) software was used to simulate it. The simulation aided in monitoring the design stages so that any mistake in the circuit can be corrected and the designer is safe from the prize of a new component. The printed circuit board was also designed using CAD, "Easily Applicable Graphical Layout Editor (EAGLE)." This is a modern approach to electronic construction. From the results obtained, there was a slight variation between the determined and measured values. The measured values were less than the calculated values. This was due to the tolerance of the components used especially the 5% resistors used. This work was undertaken in threefold: the design, simulation and construction. The three approaches were successfully achieved.

Keyword: Audio mixer, Simulation, Signal, CAD, Noise and Complex network

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INTRODUCTION

An audio mixer is an electronics instrument that accepts a large number of signals. Through a small space and keeps them strictly apart until the operator chooses to mix them. A well design audio mixer accepts all the signals to its input and reproduces them at its output where all the signals are heard without distortion or interference. Different features are added to audio mixer to; route, change level, timbre and/or dynamics of the audio signals.

An audio mixer should have the following basic features:

- Preamplifier: this section consists of circuits that amplify the weak voltage from the microphone to a higher more useable level. Most microphones must be used in conjunction with microphone preamplifier to function effectively. the preamp consist of a trim or gain control which controls the amount of amplification or attenuation needed to bring the signal to a nominal level for processing.

- The equalization stage; this section separately attenuate or boost a range of frequencies (e.g. bass, mid-range, treble).

- The pan-pot: this split the signals into two sections which are treated as left and right. This gives smooth panning without unwanted level change.

- Mix-routing: this is the point where all the signals are combined before feeding the mixer. Normally, all the signals on the left sides are joined together to form the left channel, and those on the right sides to form the right channels.

- The final mixing is done by virtual-earth mixing technique which is normally an inverted op-amp mode.

The objective of this research is to design an audio mixer, simulates its performance using Computer Aided Design (CAD) software and to construct it, aiming bridging the gap between the theory and the practical aspect of audio mixer design.

This research work after construction will be of great importance in live performance and recording of music. The design stages will aid in a better understanding of building complex system from cascaded simple circuits. The components values were obtained by Equations, simulation was done using student version of circuit maker (CM60S). The board on which the components were assembled was designed using (CAD) Easily, Applicable Graphical Layout Editor (EAGLE). The threefold approach to this research was carried out and achieved successfully.

Components Design

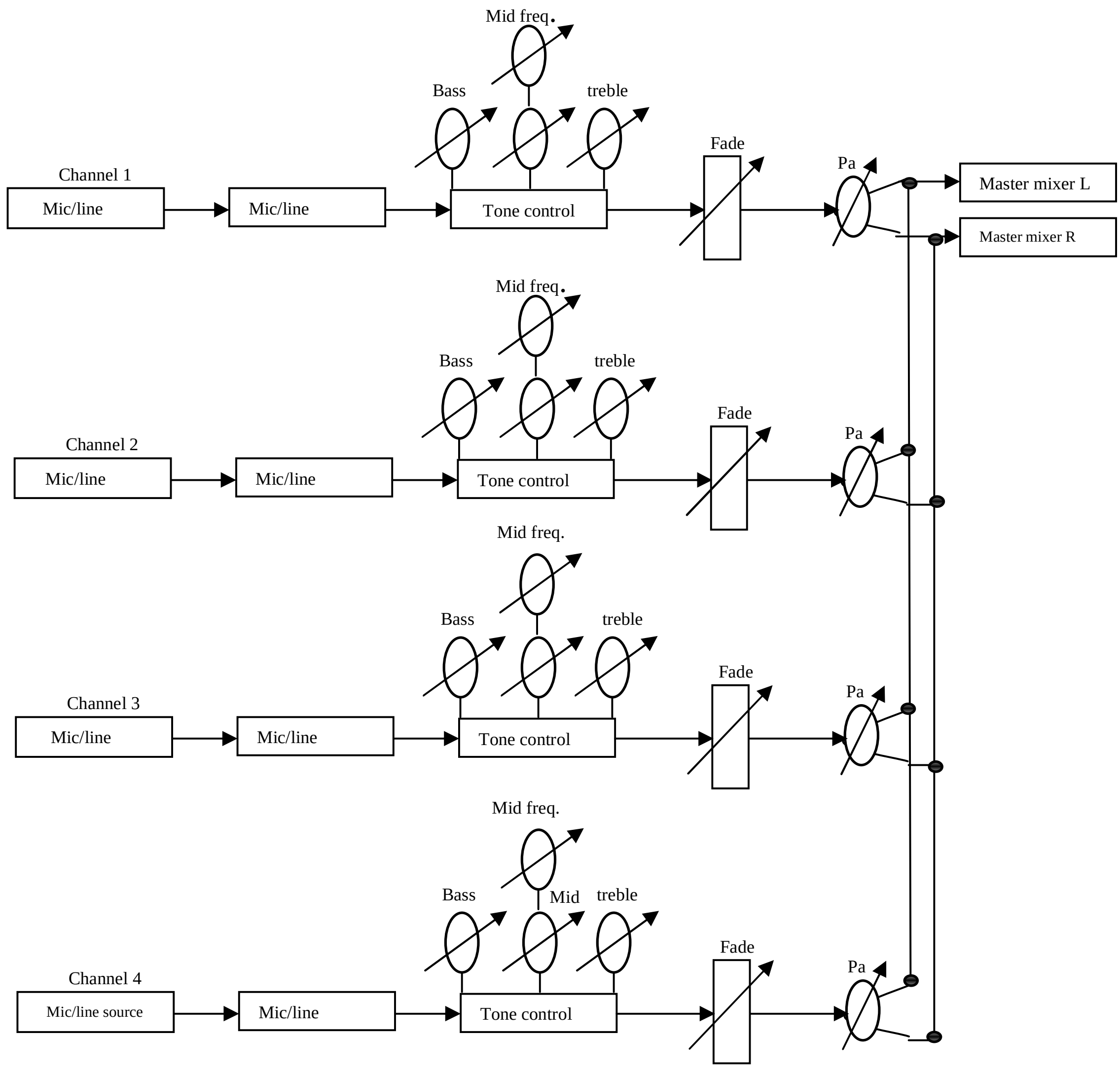

Fig.1 Block diagram of the four channels audio mixer

To simplify the design, the whole art splitter into four modules and will be treated accordingly.

Microphone Preamplifier Design

The microphone preamplifier modules the most important part of mixer design because it is at this section that noise is generated apart from the power line noise.

The preamp module has two input connectors the microphone input and the line input connectors, 10dB attenuation pad, variable gain control, peak detector, active tone control fader and pan-pot [1].

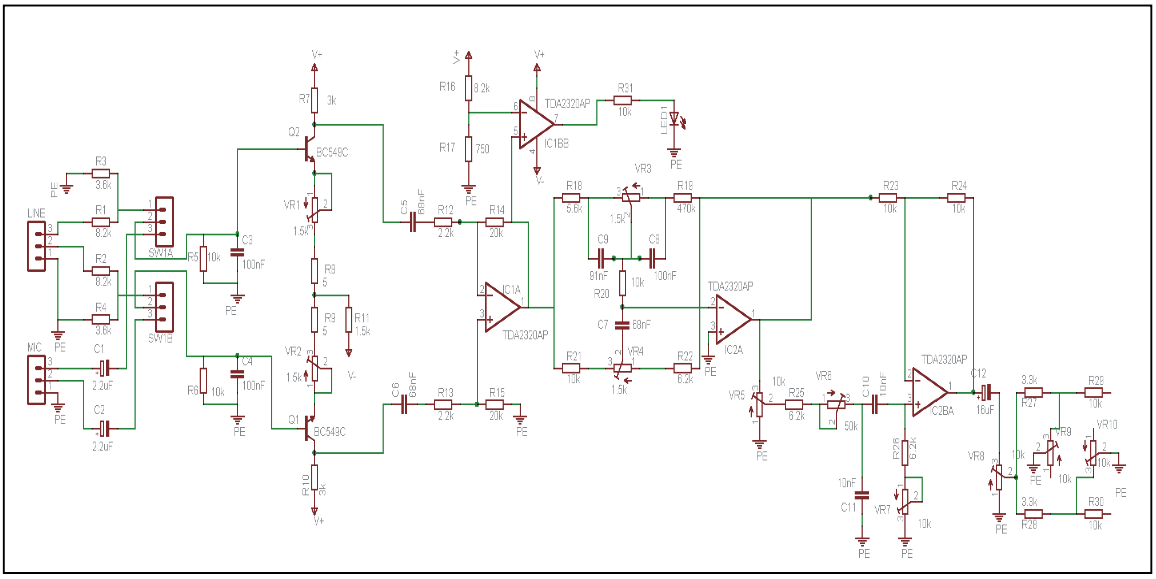

Fig.2 preamplifier module

Signal from microphone or line is selected by the switch sw_1. The signal from the line might be large or large enough to cause clipping if allowed directly into the preamp, so a 10dB attenuation network is introduced to reduce the signal amplitude.

From Fig.2 above, the same current flows through R_1 and R_3, R_2 and R_4. Let the voltage from the line be V_1 and that to the switch be V_2.

From Ohms law,

$$V_1 = I(R_1 + R_3) = I(R_2 + R_4) \qquad 1$$

$$V_2 = I(R_3) = I(R_4) \qquad 2$$

Attenuation, the inverse of gain is the ratio of input signal to output signal [15].

$$\frac{V_1}{V_2} = \frac{R_1 + R_3}{R_3} = \frac{R_2 + R_4}{R_4} \qquad 3$$

Since the chosen attenuation is 10dB,

$$10dB = 20\log\left(\frac{V_1}{V_2}\right) \qquad 4$$

$$= 3.1623$$

Choosing $R_1 = R_2$ to be 8.2k for easier selection of R_3 and R_4.
Using 3, R_3 =3.752k (3.6k nearest preferred value.) with this new value, attenuation is 10.29dB.

Preamp Design

Assuming that the base terminal of the transistors are at ground potential with a small base current making the drop across R_5 and R_6 close to zero volts assuming also that the transistors are turned ON with their bases at zero volts, then the emitter must be at 0.7V with respect to the base. This condition is required to forward-bias the base-emitter junction and turns the transistor ON. This satisfies the biasing requirement for NPN transistors [12].

Choosing V_{CC} = 12V

$V_{OUT} = 0.5V_{CC}$ 5

From 5, $V_{OUT} = 6V$

For BC549C to conduct, I_C = 2mA, V_{CE} = 5V and V_{BE} (ON) =0.6V [16].

The following assumptions simplify the design.

- The base leads of the transistors are at ground potential and since the base current is small, it makes the voltage drop across R_5 and R_6 at zero potential.
- The base-emitter must be at -0.7V (-0.6V for BC549). This condition is essential to turn ON the transistors [11]
-

$V_{EE} = V_{R_{11}} + V_{(V_{R1}+R_7)} - V_{BE}$ 6

Choosing $R_{12} = R_{13} = R_C$ to allow the current of 2mA

$V_{OUT} = I_C R_C$ 7

Using 7,

R_C = 3k.

$I_E \approx I_C$ 8

For symmetry,

$I_T = 2I_E = 2I_C$ 9

$VR_{11} = 2I_T R_{11}$ 10

Using 10 in 6,

R_{11} = 1.5k

The minimum current gain for BC549C is 420 [14].

$\beta = \frac{I_C}{I_B}$ 11

$I_B = 4.76\mu A$

This current flows in each of the base resistors R_5 and R_6. Base voltage of 50mV is enough for microphones [5]

$R_5 = R_6 = R_B$

$R_B = \frac{V_B}{I_B}$ 12

Using 11 in 12,

R_B = 10.5k (10k nearest preferred value). The A_C resistance of the emitter (re) is

$$re = \frac{25mV}{I_E} \qquad 13$$

Using 8 in 13,

$re = 12.5\Omega$

The minimum gain occurs when VR_1 and VR_2 are at minimum, and is given as

$$\frac{V_{OUT}}{V_{IN}} = \frac{-R_{12}}{re + R_8} = \frac{-R_{13}}{re + R_9} \qquad 14$$

$R_{12} = R_{13}$, $R_8 = R_9$

Choosing a maximum gain of 100 (40dB) for low noise and using it in 14,

$VR_1 = VR_2 = 1.5k$

The common mode gain is given by

$$CMMR = \frac{A_d}{Acm} \qquad 16$$

Using 14 and 15 in 16, CMMR = 33.98dB

Differential amplifier design

From Fig.1 above, U1 has a balance input and a single ended output. To minimize Johnson noise and degrading effects in op-amp, R_{14} –R_{17} should be chosen from $(100\Omega - 20k\Omega)$ [3].

The gain of the deferential amplifier is chosen to be 10(20dB) to further amplifier the weak signal from the transistor preamplifier.

The gain for U1 is given by [8].

$$Gain = \frac{R_{16}}{R_{14}} + 1 \qquad 17$$

Choosing $R_{14} = R_{15} = 2.2k$, for a gain of 10 and using 17,

$R_{16} = R_{17} = 20k$.

C_5, C_6 set the frequency at unity which blocks DC components of the signal from the collector.

The frequency is given by

$$F_1 = \frac{1}{2\pi R_{14} C_5} \qquad 18$$

For a unit frequency, C_5 =72nF (68nF nearest preferred value). From (18)

Clipping monitor design

The clipping monitor is used to give a visual indication of when clipping occur in any of the input channels. The circuit used is a comparator which switches on an LED when a voltage at the non-inverted input exceeds the reference voltage at the inverted input.

The reference voltage is given by [4]

$$V_{ref} = \frac{R_{32} V_{CC}}{R_{31} + R_{32}} \qquad 19$$

On personal experiment with LM324, general purpose quad op- amp in amplification mode, voltage above 11.2V causes clipping. Therefore I chose reference voltage of 11V and using it in 18, after choosing R_{31} to be 8.2k

$R_{32} = 745\Omega$, (750Ω) Chosen,

With this value, clipping occur when input signal exceed 10.99V

Tone control design

The tone control is a Baxandall active filter (low and high pass filter).

The filter uses a frequency dependent feedback network that provides boost/ cut for high and low frequencies independently without interaction between the low pass (bass) and the high pass (treble) adjustment [17].

When centered, there is neither gain nor loss and so the output gives a flat pass-band with the op-amp acting only as buffer.

The gain for the low pass is given by

$$A_V = \frac{R_{18} + R_{19}}{VR_1} \qquad 20$$

For a gain of 20dB and choosing R_{18} to be 5.6k, VR_1 to be 50k based on availability, R_{19} = 490k (470k nearest preferred value). The gain with the new value is 19.56dB
The gain for treble is given by

$$A_V = \frac{R_{18} + 2R_{20}}{R_{21}} \qquad 21$$

For a gain of 20dB and choosing R_{21} to be 2700, R_{20} = 10.7k (10k nearest preferred value). The gain with the new value is 19 54dB. At lo frequency, the capacitor C_9 is open isolating the high pass from the low pass filters. At maximum boost, the capacitor C_7 is shorted and the -3dB turnover frequency is given by

$$F_2 = \frac{1}{2\pi VR_1 C_8} \qquad 22$$

Choosing maximum boost to be 30Hz, C_8 =106nF (100nF nearest preferred value)
F_2 with this new value is 31.83Hz. The frequency at maximum cut is given by

$$F_3 = \frac{1}{2\pi R_{18} C_7} \qquad 23$$

In this case, C_8 is shorted. Choosing the maximum cut to be 300Hz. C_7 = 95nF (91nF nearest preferred value).

The maximum cut with the new capacitor is 312.27Hz.
At high frequency, C_7, C_8 are shorted, causing VR_1 to be zero.

The frequency at maximum boost is given by

$$F_4 = \frac{1}{2\pi R_{22} C_9} \qquad 24$$

Choosing F_4 to be 2.5 kHz and C_9 10nF for easier selection of R_{22}. From 23, R_{22} = 6.4 kHz (6.2k nearest preferred value). The new frequency is 2.57 kHz.

The frequency at maximum cut is given by

$$F_5 = \frac{1}{2\pi(R_{19} + R_{22} + 2R_{20})C_9} \qquad 25$$

F_5 =32.32.07 Hz.
U_3 forms a band-pass filter. The gain is given by

$$A_V = \frac{R_{23}}{R_{24}} + 1 \qquad 26$$

The resistors in Equation 26 are choosing to be 10k given a gain of 2. The band pass is chosen to give a narrow and wider band width. VR_3 alters the pitch of the signal.

The band pass filter consists of two sections; the low pass and the high pass sections.

While the frequency of the high pass section increases, those of the low pass section decreases giving a narrow band width. When the reverses occur, a wider band width is obtained. This technique can be achieved using dual pot connected such that as the resistance of the other pot increases the other pot's resistance decreases. For the high pass section, the frequency is given by

$$F_6 = \frac{1}{2\pi(R_{26} + VR_5)C_{10}} \qquad 27$$

Choosing the lower section of the band pass to be 160Hz, two octaves lower than the 640Hz center frequency, [5] and VR_4 to be 50k, and C_{10} to be 10nF for easier selection of R_{26}.
Using 27, R_{26} =49.45k (47k nearest preferred value). The new frequency is 164.1Hz

The maximum frequency is obtain whenVR_5 is minimum. From 26, it was computed as
F_7 =338.6Hz.
For the low pass section, the frequency is given by

$$F_8 = \frac{1}{2\pi(R_{25} + VR_4)C_{11}} \qquad 28$$

Maximum frequency occurs when VR_4 is at minimum. The frequency at this point is chosen to be 2.56 kHz, two octaves higher than the 640Hz center frequency.
Using 28, R_{25} is 6.22k (6.2k nearest preferred value). The new frequency is 2.57 kHz

The minimum frequency for the low pass section is obtained when VR_4 is at maximum. This frequency F_9 is obtained from 27.

F_9 = 283.16Hz.
Therefore the narrow bandwidth is

$$F_7 = F_9 \qquad 29$$
$$= 55.4\text{Hz.}$$

The wider band width is

$$F_8 = F_6 \qquad 30$$
$$= 2.4\text{ kHz.}$$

The mid frequency is therefore variable from 55.4HZ to 2.4 kHz.
VR_6 is the fader (volume control) for the particular channel which controls the pitch from minimum to maximum.
C_{12} and VR_6 forms a high pass filter for attenuating lower frequency ripples.

Choosing a cut-off frequency of 1Hz [8] and VR_6 10k base on availability,
$C_{12} = 15.9\mu F$
$C_{12} = 16\mu F$, then the new frequency is 0.99Hz. R_{27}, R_{29}, VR_6 forms a dual T-network. When centered, the pan-pot is at 25k. Choosing R_{27} to be 3.3k and R_{29} to be 10k, the net resistance is

$$R_{NET} = \frac{R_{27} \times R_{29}}{\frac{1}{2}(VR_6)} \qquad 31$$
$$= 1.32\text{k.}$$

This is the input resistance to the master mixer.
Master Mixer/Headphone Amplifier Design

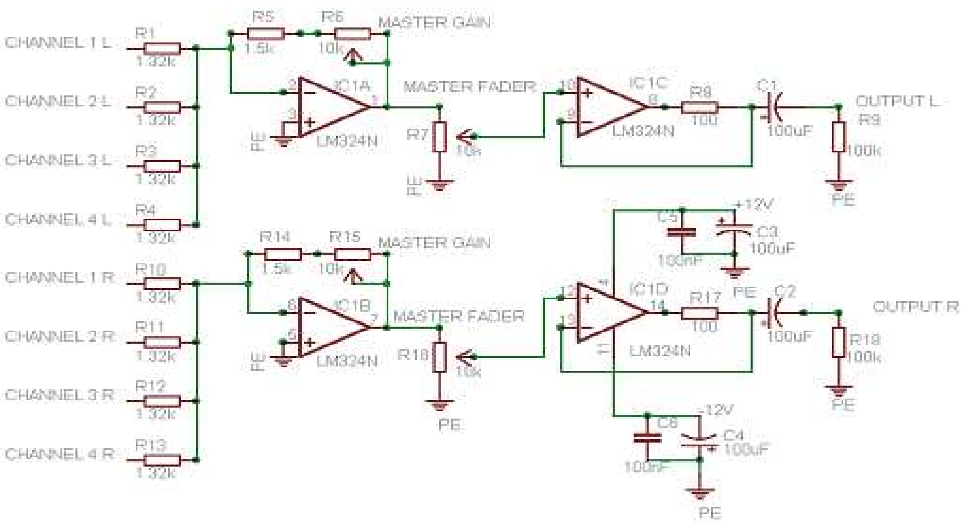

Fig.3: Master mixer

The master mixer in Fig.2 above is a summing amplifier which is base on the inverting buffer (virtual-earth).

When signal is applied to the input, it flows through the 1.32k input resistance. The op-amp will tend to maintain equal voltages at its two inputs. Since the non-inverted input is at ground potential, the inverted input has to stay at same potential thus virtual earth.

When the four inputs are loaded, the op-amp gives an output that is the sum of the four input signals. If the signals have same amplitude and polarity, they are simply added.

$$R_{NET} = \frac{R_{33}}{1.32k}$$

$$= 1.14$$

Close to one but inverted. When all the signals are fed, the op-amp sees an input resistance of $4 \times 1.32k$ (5.28k).

The mixer attenuates at this level so VR_8 can be varied to increase the gain. This multiplies the noise to this section by 4. [5], to allow for more gain, we chose VR_8 to be 50k. The inverted amplifier stage is noisier than the non-inverted stage. [2] to minimize noise, a non-inverted operational amplifier is introduced with a gain of 1. A high pass filter is also introduced to minimize low noise. VR_9 is a dual pot, acting as the fader for both the left and right master mixers.

Headphone amplifier design

The head phone amplifier is based on the TDA2320 amplifier offering very low distortion, no pop sound and high quality performance. It is capable of 0.4W into 8Ω

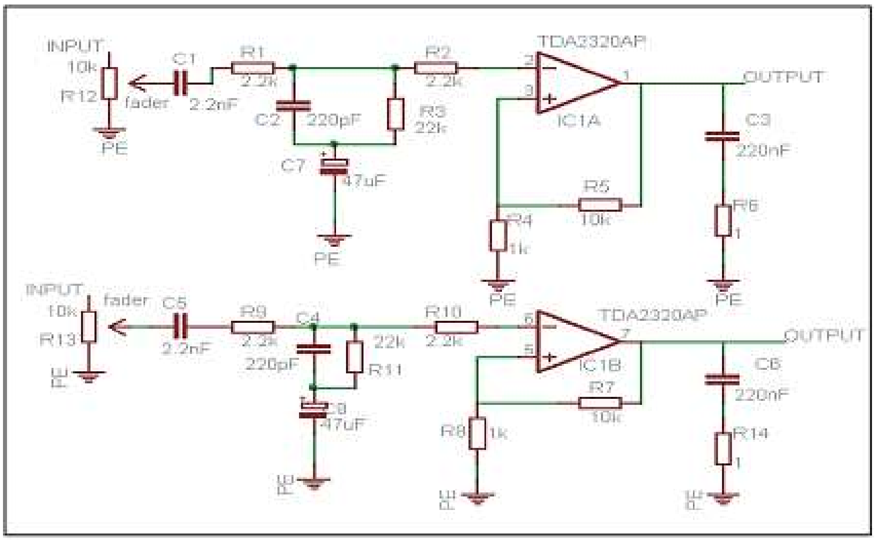

Fig.4: head phone amplifier

R_{42}, R_{43}, R_{44} forms a passive band pass filter. The gain is set by R_{45} and R_{46}. Using 26, the gain for the head phone amplifier is 11.

Power supply module

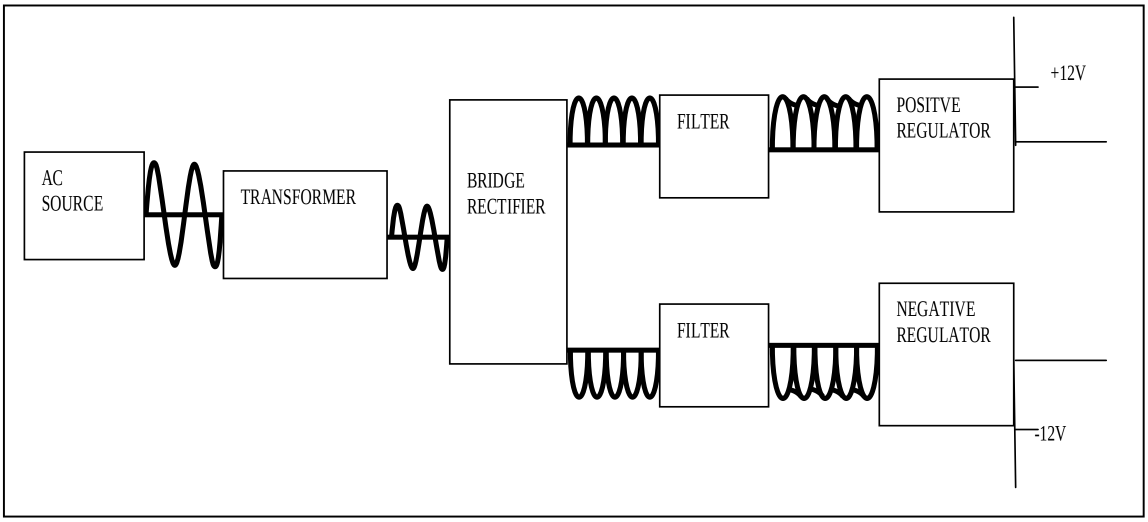

Fig.5: block diagram of bipolar power supply

Most electronic devices require DC voltage to operates, batteries are good example of such, but operating time is limited unless recharged or replaced. Since the most convenient and economical source of power is the domestic AC supply. It is of greater advantage to convert it from alternating form usually 220VAC to a DC voltage.

Power supply design

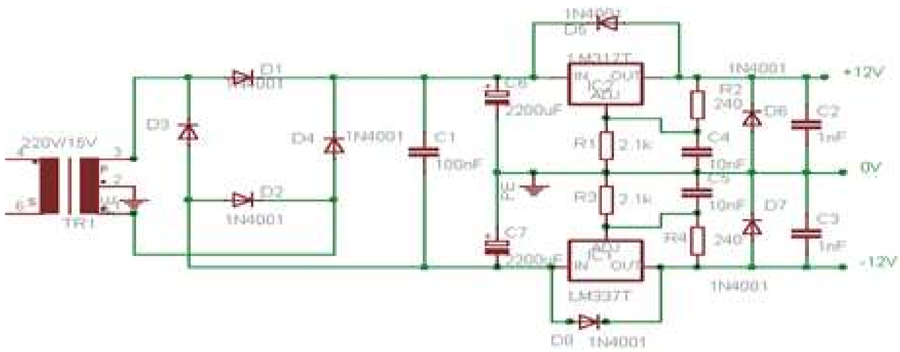

Fig.6: power supply circuit

In designing a rectified power supply, two stages are most important; the

The block diagram below shows the stages involved in converting AC to a smooth and stable bipolar DC source. [6]

From Fig.4 above, when an alternating source is applied to the power board at 220V AC, the transformer steps it to down to 15V AC. it is next fed to the rectifier which converts it to a pulsated DC source. A filter is introduced which tends to smoothen the output. A voltage regulator is finally introduced, it stabilizes and further smoothen the output voltage.

DC analysis stage which gives the size of the transformer and the AC analysis stage which gives the size of the capacitor.[3], the design specification are,

$I_L F_L = 800mA$

$V_{DC} F_L = \pm 12V$

$R_0 = 8\Omega$

$Ripple \angle 5\%$

$V_{DC}N_L = V_{DC}F_L + I_{FL}R_0$ 32

$V_{DC}N_L = 18.4V$

$$E_{r.m.s.} = \frac{V_{DC} N_L}{1.4} \quad 33$$

= 13.14V

Rectifier diodes should be chosen such that the average current should be more than that demanded by the circuit to avoid over heating of the diodes [3], the average current is given by

$I_{av} \geq I_L F_L$ 34

$I_{av} \geq 800mA$

The peak-inverse-voltage (PIV) that is, the reverse-bias voltage which if exceeded the diode will be destroyed. It should be greater than.

$V_{DC}N_L + 20\% \ V_{DC}N_L$ 35

$PIV \angle 22.08V$

The diode 1N4001 has I_{av} of 1A and PIV of 50V [11]. Choosing a capacitor with ripple less than 5%, at full load,

$$V_{r.m.s.} = \frac{(\%ripple)V_{DC} F_L}{100} \quad 36$$

= 0.6V

$\Delta V_0 = 3.5 \times 0.6$

= 2.1V

For full wave rectification, the capacitance C is given by

$$C = \frac{I_L}{200 \Delta V_0} \quad 37$$

$= 1900 \mu F$

Allowing for 20% tolerance, [11]

$C = 2200 \mu F$

The working voltage of the capacitor is given by 35, and is V_W =22.08V (35V) safe to cover 20% tolerance.

Dual regulator design using LM317T and LM337T

The voltage regulator used in this design is based on the LM317T and LM337T positive and negative variable regulators respectively. The dual regulations means positive regulation from

LM317T and negative regulation from LM337Tthe dual (bipolar) ±12V regulators are capable of delivering 1.5A from 1.25V to 32V. [17]

Design specification /procedure

$V_{OUT} = \pm 12V$

For proper regulation, there should be a minimum drop-out voltage of 3V across the input output terminal of the regulator [9].

A nominal 1.25V should be left between the output and the adjustment terminal. The resistors $R_1$and R_2 or R_3 and R_4 are connected between these terminals to conduct a current given by [8]

$$I = \frac{V_{ref}}{R_1} \quad 38$$

With R_1= R_4= 240Ω, I = 1.5A.

This current also flow through R_2 and R_4 causing a voltage drop given by

$$VR_2 = R_2 \times 5.2mA \quad 39$$

Using Kirchhoff's voltage rule, the output voltage of the regulator is given by

$$V_{OUT} = V_{ref} + IR_2 \quad 40$$

From 35 and 37, R_2 and R_4 are 2.067k (2.0k nearest preferred value)

The diodes D_5 and D_8 protect the regulator from input short while D_6 and D_7 protect from output short [10]

Method of Simulation

Simulation

The term simulation is used in different ways by different people. As used here, simulation is defined as the process of creating a model (i.e. an abstract representation or facsimile) of an existing or proposed system (e.g. a project, a business, a mine, a water shed, the organs in the body etc.) in other to identify and understand those factors which control the system and/or to predict (forecast) the future behavior of the system. [7]

A Computer Aided Design (CAD) software version of circuit maker (CM60S) was used to simulate the designed circuits. The simulation aided in monitoring the design stages in case of mistake or future failure.

The software used can allow only 50 components on its work-station; therefore the circuits are split into four parts so as to be accommodated.

The Fig. 7 below show the simulated circuit with their output waveforms and frequency response respectively on the right hand side.

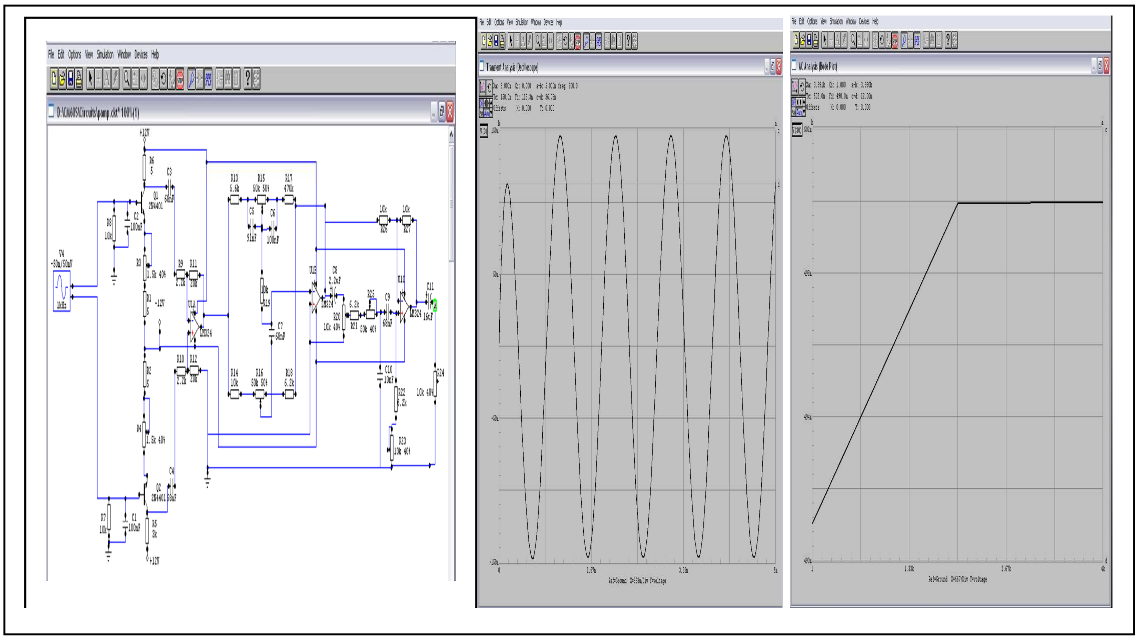

Fig.7: Simulated preamplifier/tune control

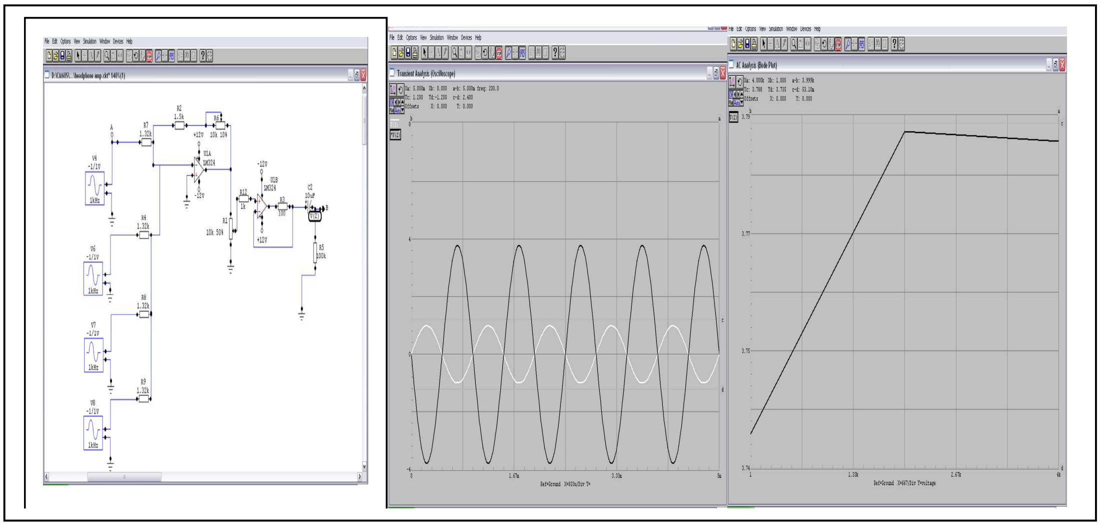

Fig.8: Simulated master mixer

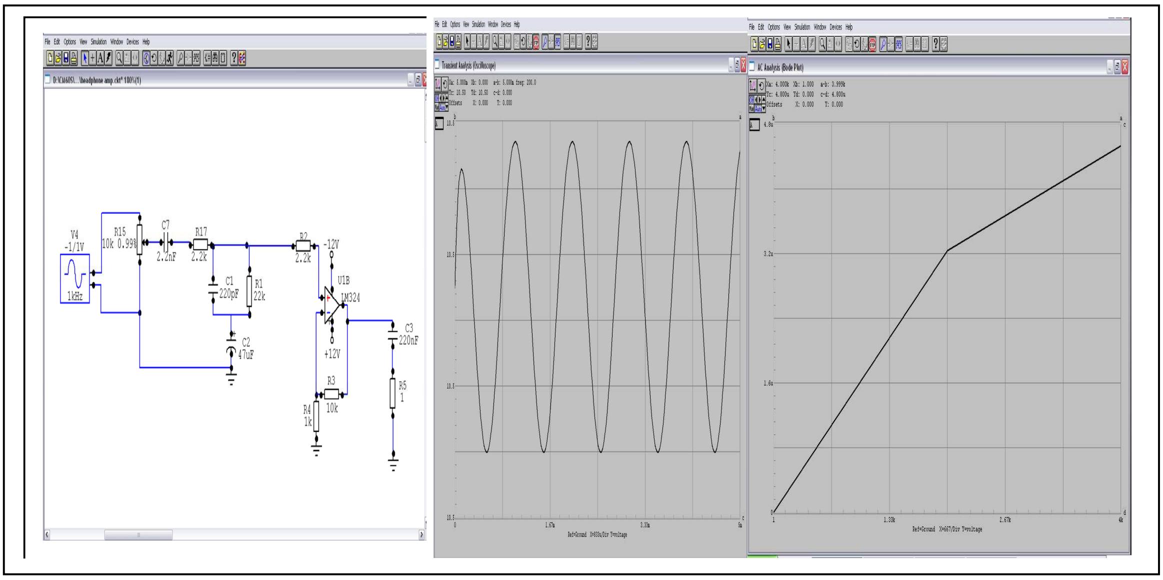

Fig.9: Simulated headphone amplifier

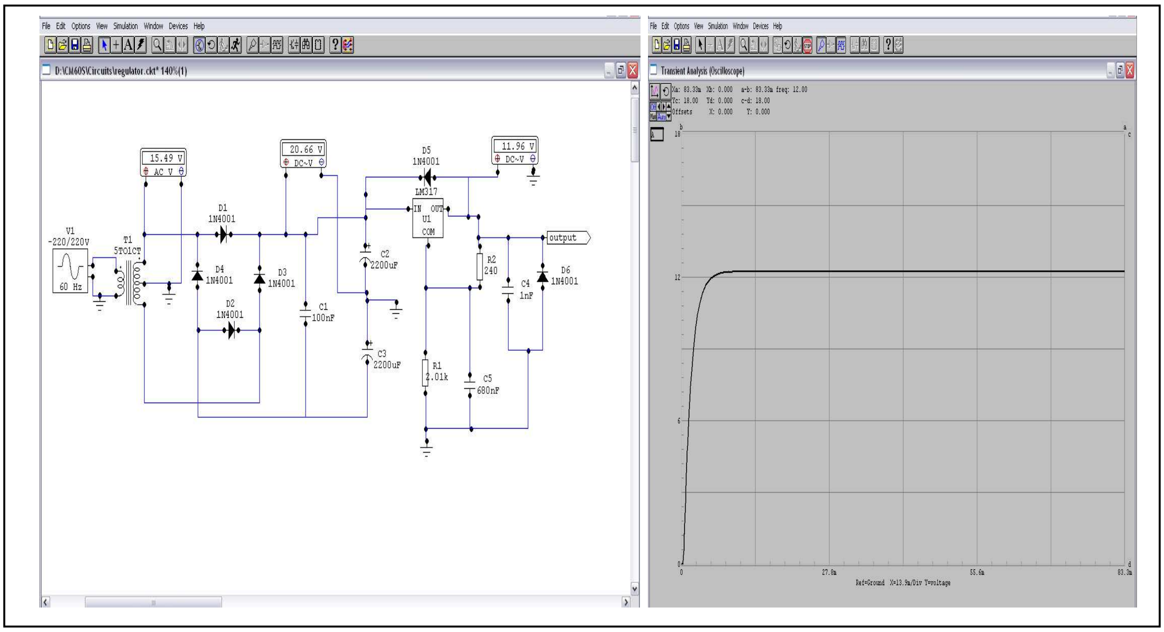

Fig.10: Simulated regulator

RESULTS

Simulated results

From the simulated results of Fig.7, it is evident that circuit reproduced its input without distortion. The bode analysis indicates that the gain is constant at frequency above 2.001 kHz. The simulated master mixer and headphone amplifier all shows faithful reproduction with variation in the frequency display indicating real behavior. The simulated regulator shows a clean straight line indicating noise free output and the DC output is 11.99V

Table 1: Comparison between theoretical and measured values

Modules	Theoretical values (dB)		Measured values (dB)	
	Input voltage	Output voltage	Input voltage	Output voltage
Preamp at minimum gain	0	0	0	-1.21
At maximum gain	0	40	0	36.7
Power supply	-	11.99	-	11.86
Master mixer at minimum gain	0	2	0	1.56
At maximum gain	0	20	0	18.2

CONCLUSION

From the simulated results above, it is clearly shown that the circuits' waveforms and frequency display are almost close to theoretical values. Theories and principles when put together gives a model .this work were under-taken in threefold; the design, simulation and construction. Conclusively, the work was successfully constructed and tested and found to be working satisfactorily.

Recommendation

This audio mixer can therefore be used for live performance and as an aid in learning circuit design and construction. Input channels can be extended to 64, but power source should be design to handle up to 4A. Graphic, or parametric, or state variable, or stereo equalizer filters can be added to the master mixer to add quality and style. Modern digital effects can be added as well.

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