Design of 16 Channel R.F. Mixer

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Abstract: *An audio mixer amplifier is a device that translates a signal of one frequency band to another. It will accept many inputs at different frequencies and generate an output of the combination or sum of the frequencies. The mixer circuit provides good gain to weak audio signals. It can be used in front of an R.F. oscillator to make an R.F. receiver that is very sensitive to sound. Each input can be independently controlled by a variable resistor. There is also a provision for a balance control to fade out signal while simultaneously fading in the other.*
Key Words: *Audio Mixer, Frequency, Signal, Circuit.*

I.   Introduction

A.M. radio receivers can be broadly classified into two types viz., straight radio receiver and superhetrodyne radio receiver. The former was used in the early days of radio communication. However at present, all radio receivers are of superhetrodyne type.Straight radio receivers were used to reproduce radio waves into sound waves. The aerial receives radio waves from different broadcasting stations the signal is amplified decoded and theresulting audio signal is amplified and fed to the speaker for sound reproduction. The major limitation of straight radio receivers are (i) tuned circuits are used. This is because it is necessary to change the value ofvariable capacitors (gang capacitors) for tuning to the desired station, therefore, there is a considerable variation of charge between the closed and open positions of the variable capacitors. This changesthe sensitivity and selectivity of the radio receivers.(ii) There is too much interference of adjacent stations.

The shortcomings of straight radio receiver wave is overcome by the invention of superheterodyne receiver by Edwin H. Armstrong during the First World War. At present, all modern receivers utilise the superhetrodyne circuit. In this type of radio receiver, theselected radio frequency is converted to a fixed lower value, called intermediate frequency (IF). Thisis achieved by a special electronic circuit called mixer circuit. There is a local oscillator in the radio receiver itself. This oscillator produces high frequency waves. The selected radio frequency ismixed with the high frequency wave by the mixer circuit. In this process, beats are produced and themixer produces a frequency equal to the difference between local oscillator and radio wave frequency. The circuit is so designed that oscillator always produces a frequency 455kHz above the selected radio frequency. Therefore, the mixer will always produce an intermediatefrequency of 455 kHz regardless of the station to which the receiver is tuned. A frequency mixer is necessary to produce the intermediate frequency in a superheterodyne receiver.

Mixer

A frequency mixer is a 3-port electronic circuit. Two of the ports are "input" ports and the other port is an "output" port. The ideal mixer "mixes" the two inputsignals such that the output signal frequency is either the sum (or difference) of the inputs frequencies. In other words:

$$\text{fout} = f_{in1} \pm f_{in2} \qquad\qquad (1)$$

The nomenclature for the 3 mixer ports are the LocalOscillator (LO) port, the Radio Frequency (RF) port, andthe Intermediate Frequency (IF) port. The LO port istypically driven with either a sinusoidal continuous wave(CW) signal or a square wave signal. The choice to applya CW or square wave signal depends on the applicationand the mixer. Conceptually, the LO signal acts as the"gate" of the mixer in the sense that the mixer can beconsidered "ON" when the LO is a large voltage and"OFF" when the LO is a small voltage. The LO port isusually used as an input port.(Marki&Marki, 2010).

The other 2 ports of the mixer, the RF and IF, can beinterchanged as either the second input or the outputthe actual configuration depends on the application.When the desired output frequency is lower than thesecond input frequency, then the process is calleddownconversion and the RF is the input and the IF isthe output. The relationship between input and outputfrequencies is given by:

II. Design Methodology

In the design of audio mixer (16-channel), the following were taken into consideration:

(i)   The audio mixer: This has to do with the mixer itself which is a device that combines a number of audio signals at its input to produce an audio output. That is, it deals with audio signals only.

(ii)   The needs of the Mixer: This involves:

    (a)   The input devices for sound such as microphone, tape recorders, guitars, etc.

    (b)   The output devices such as amplifiers and loudspeakers.

    (c)   The power supply to operate it and also the internal power requirements of the components used.

(iii) Places of Application or Uses: These include conference halls, open fields or music industry.

(iv)   Cost Implication: This involves the cost of materials and that of labour and construction which must be as economical as possible

Considering the points raised above, one arrived at having an audio mixer that:

(i)   Is efficient in combining a number of audio signals to produce a fine audio output with means of filtering of noise that may accompany the input signal. So, filtering circuits are then incorporated with the use of resistors and capacitors (discussed later);

(ii)   Has input and output outlets for input and output devices - jackplug sockets were used to achieve this;

(iii) is able to use the common 220-240v a.c. mains as its power supply and also a small step-down transformer (240/9v) to take care of the internal power requirements for the components;

(iv)   is compact (not too big or heavy) and can be carried about easily as it is to be used with other heavy equipment;

(v)   Is economical and the components are readily available to come by in the market. Also, repairs (if faulty) must be simple.

THE DESIGN

The major design which results from the factors enumerated above was achieved by dividing the project into a number of stages. The stages are:

(a)   Power stage

(b)   Stereo stage

(c)   Auxiliary stage

(d)   Volume control stage

Power Stage

The stage involves the use of a 240/9V transformer, two 3300μF 35V capacitor and two IN5408 power diodes. The description and use of the components were discussed' earlier under review of components (Chapter Two). The diagram of the power stage is shown below.
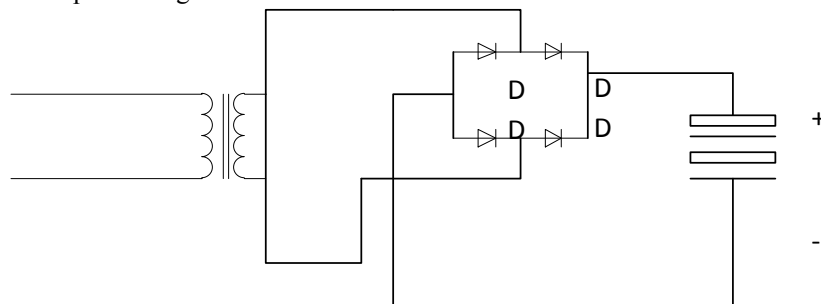

Figure 3.1: Diagram of Power Stage

D - Diode (1N5408)

C – Electrolytic capacitor (330μFf 35V)

S - Switch

To choose the mains transformer the voltage requirement of the mixer circuits and the components used were considered. The voltage requirements 3 - 12V. So the transformer must be one that will be able to produce a voltage of 12V or a little above bearing in. mind that the rating on a transformer is normally in Root Mean Square (RMS).

From the relation

Peak voltage Vp = $\sqrt{2}$ x R.M.S voltage, Vrms

$$V_P = \sqrt{2} \times V_{rms}$$

$$V_{rms} = \frac{V_P}{\sqrt{2}}$$

For $V_P = 12$

$V_{rms} = 8.485V$

Considering the rating of the available transformers, the above value (S.4S5V) can be approximated to 9V. Therefore, a 240/9V transformer was chosen. With a 9V transformer the peak voltage,

$$V_P = \sqrt{2} \times 9$$

$= 12.7V$

The 0.7V difference is still very normal when one considers the low voltage from the mains supply which may even be below 200V atimes.

For the diodes used for rectification the IN5408 was chosen. It has a peak reverse voltage of 150V.

For the capacitor the expected peak voltage is 12.7V which means that the peak - to - peak voltage will be

Vp_p=2x12.7V

$= 25.4V$

So, a capacitor whose rating will be more than 25.4V must be used.

For the capacitance of the capacitor we consider the following:

$$Capacitance = \frac{Period, T \times Peak\ Voltage}{Required\ Ripple\ Voltage \times Projected\ Load}$$

The period is that for charging and discharging of the capacitor which is 0.002s for full wave rectifier.

Period T = 0.002s

Peak voltage = 12.7V

Required ripple voltage = 5% peak - to - peak voltage.

= 5/100 x 25.4 = 1.27V

Projected load = 8Ω

$$Capacitance, C = \frac{0.002 \times 12.7}{1.27 \times 8} \qquad = 0.0025$$

C = 0.0025 x10^{-6} F

For better smoothening, a capacitance of higher value can be used. Also, considering the available capacitor in the market the 3300 micro farad capacitor was preferred. This gave rise to the use of 3300f-lF 35V capacitors.

Stereo Stage

This is the stage where the input signal is processed. The audio input signal is fed directly to the stereo stage, through the input devices. The audio signal is processed and filtered. The filtering is achieved by the use of capacitors and resistors.

The stereo stage involves the Gain, Frequency (Treble) and Mid (Bass). The Frequency and Mid are to perform the filtering job. At this juncture, it will be necessary to say something about filters.

Filters

The reactance of inductor and capacitors depend on the frequency of the a.c signal applied to them. That is why they are known as frequency selective components. By using various combinations of resistors, inductors and capacitors we can make circuits that have the property of passing or rejecting either low or high' frequencies or bands of frequencies. These frequency selective networks which act on the : amplitude and phase characteristic of the input a.c signals are called filters. Their performance is usually expressed in terms of how much attenuation a band of frequencies that passed through them experienced. The attenuation is expressed in terms of decibels (dB). Filters are circuits that are -able to select or reject signals according to their frequencies. They are classified according to the band of frequencies which is passed or stopped.

Tone Circuit

The circuit below was designed for the tone circuit (low and high pass filters), which performed the function of bass and treble respectively.
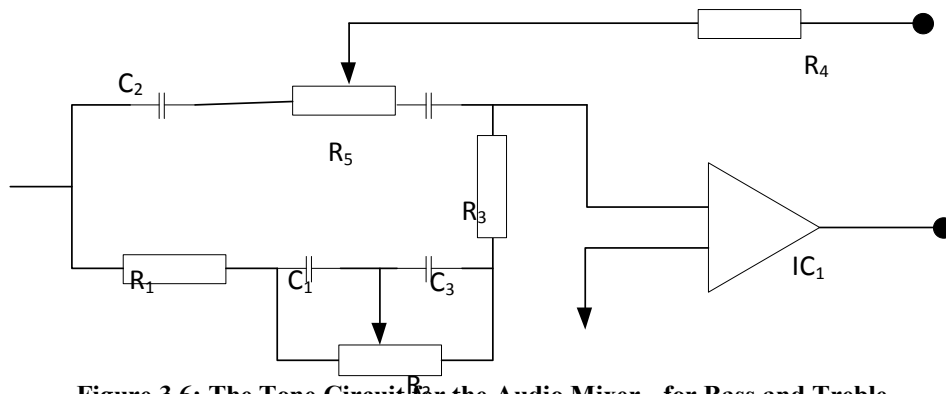
Figure 3.6: The Tone Circuit for the Audio Mixer - for Bass and Treble

Simply put the term audio means sound. The normal audio signal is of low level, that is, a signal of the range of frequencies which our ears can hear. The range of human hearing extends from 300 to 3400Hz.

The constructed audio mixer accepts audio inputs (up to 16) and combined them together to give an output. The output is monitored by the use of variable resistors (50KΩ).

The 50kΩ resistors were used as volume controls. In the stereo stage is the gain control (50KO variable resistor). The input signal from the input socket goes directly to the gain control which in turn feeds the Bass and Treble.

Note: Each channel of the stereo stage has its individual Gain, Treble and Bass controls.

In choosing components for the tone circuit, the Gain, Treble and Bass were considered. The Gain is to have a frequency range of 2 - 20 KHz that for treble is 200Hz - 2 KHz and the Bass is to have a frequency range of 20 - 200Hz.

The expression for frequency is given by

$$f = \frac{1}{2\pi RC}$$

For the Treble

R = 10KΩ = 10000Ω
C = 223 = 22 x 10^{-9} F.

and π = 3.142

$$f = \frac{1}{2\times 3.142 \times 10000 \times 10^{-9}}$$

= 723Hz.   This is a value between 200Hz and2Khz.

For the Bass

R = 10K = 10000 ohms
C = 104 = 100 x 10^{-9} F
π = 3.142

$$f = \frac{1}{2\times 3.142 \times 1000 \times 100 \times 10^{-9}}$$

= 159 Hz

This is a value between 20 - 200 Hz

For the Gain, considering circuit diagram of the stereo stage of the audio mixer the resistance R is 470 0 resistor in parallel with 1 OOK 0 resistor

From $\dfrac{1}{R} = \dfrac{1}{470} + \dfrac{1}{100000}$

$$R = \frac{470 \times 100000}{470 + 100000}$$

R   =   467.8 ohms.
C   =   473 = 47 X 10^{-9} F

$$f = \frac{1}{2\times 3.142 \times 467.8 \times 10^{-9}}$$

= 7238.7 Hz

This is a value between 2 - 20 KHz.

Varying the 50k ohms resistors (use) as the volume control, one can vary the frequencies thereby changing the tone (i.e. the output sound) as desired.

Auxiliary Stage

To improve the final output (at the loudspeaker), an auxiliary stage was designed for the mixer. The stage is fed directly from the output of the stereo stage via a resistor. Two auxiliary stages (A1 and A2) were provided.

Volume Control Stage

There are two major controls in this stage, master volume control and master auxiliary control. The master volume control is used to control all the volume at the same time. All the outputs of the stereo stage were looped together and connected to a master volume control (50KΩ variable resistor).

As done by the master volume control, the same is applicable to the master auxiliary control. The outputs of the auxiliary stage were looped together and connected to a master auxiliary control. The figure below shows the block diagram of the stereo stage with Gain, Frequency (Treble) and Mid (Bass).
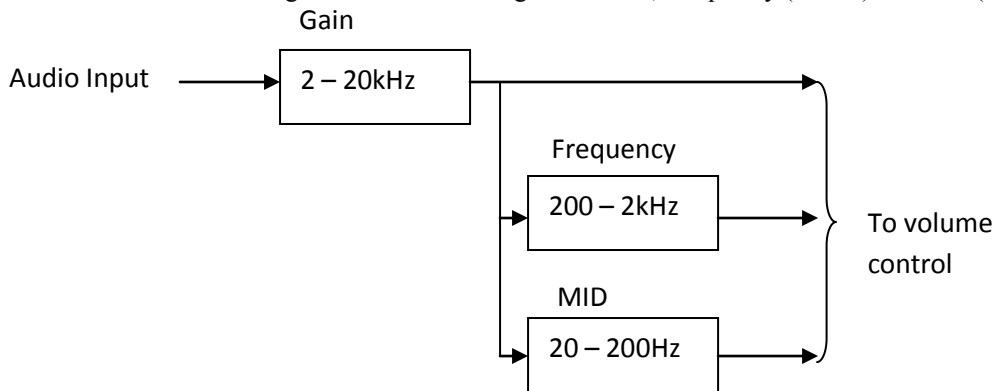
Figure 3.7: Block Diagram of the Stereo Stage

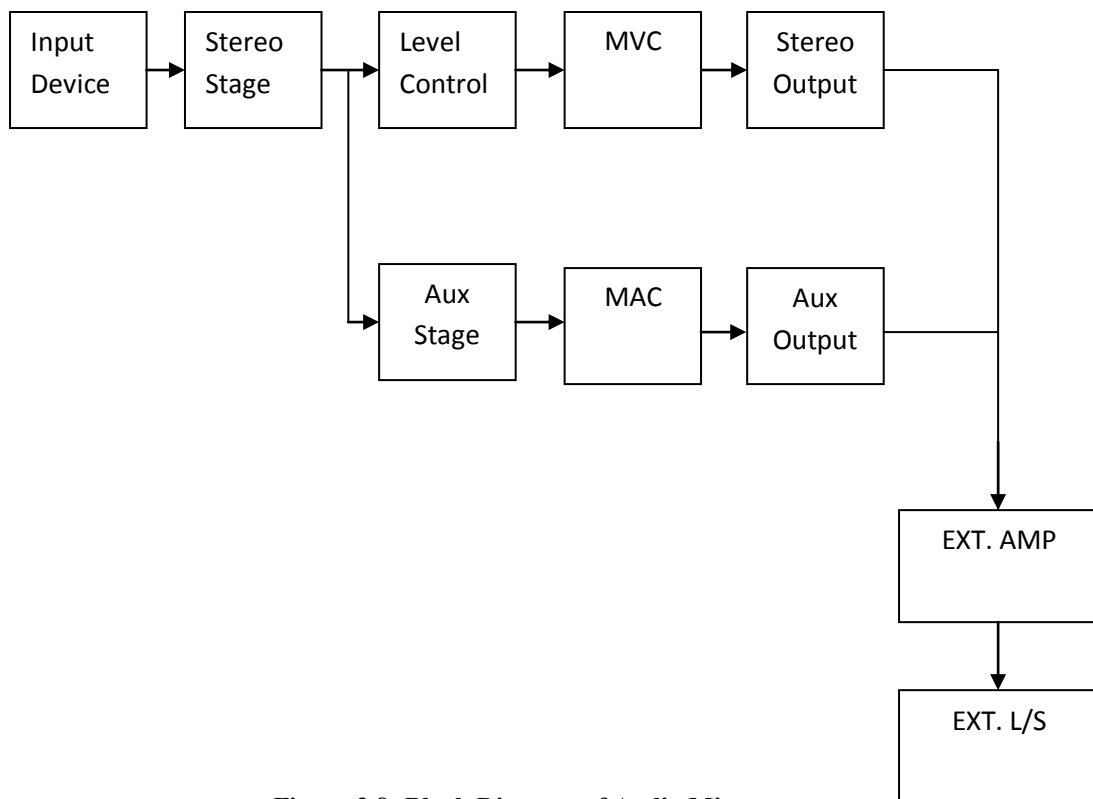A simple block diagram of the audio mixer is as shown below:

Figure 3.8: Block Diagram of Audio Mixer

Note

MVC   - Master Volume Control
MAC   - Master Auxiliary Control
Aux   - Auxiliary
Ext   - External
L/S   – Loudspeaker

III.   Conclusion

The 16 channel audio mixer is tested by connecting it to an external amplifier which is in turn connected to loudspeakers. The connection of the amplifier is to the output of the mixer. The heat generated is very low compared to that of an amplifier. As a result of this a wooden casing can be conveniently used as the case. It should be connected to an amplifier for heavy load.

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