AUDIO MIXER

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WAKNAGHAT

CERTIFICATE

This is to certify that the work entitled **"Audio Mixer"** submitted by **Puneet Garg – 081112, Harsh Vikram Singh – 081039, Arjun Puri – 081041** in fulfillment for the award of degree of Bachelors of Technology in Electronics and Communication Engineering of Jaypee University of Information Technology, has been carried out under my supervision.

This work has not been submitted partially or wholly to any other University or Institute for the award of this or any other degree or diploma.

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SUMMARY

In professional audio systems, a mixing console, or audio mixer, also called a sound board, is an electronic device for "mixing", routing and changing the level of timbre of an audio signal. A mixing console has various applications like studio recording, live shows, DJ mixing, etc. Most bands use a mixing console to combine musical instruments and vocals to the correct level. A modern mixing console looks like this :

Yamaha 2403 Audio Mixing Console

We are making an effort to build an audio mixer with best possible sound output, given the availability of components. Efforts have been made to reduce distortion and other unwanted noise. We are making a 3 input audio mixer, all of which are microphone inputs.

A project of soundboard is incomplete without the knowledge of acoustics. So we have made a conscious effort to learn about sound as much as possible.

CHAPTER 1

INTRODUCTION

An audio mixer can be defined as a circuit that takes a number of audio input signals and generates an audio output that is directly related to the inputs. The processor may, for example, simply invert the original signal and/or provide it with a fixed amount of voltage gain as in case of linear amplifier; or it may give an amount of gain that varies with signal frequencies, as in case of an active filter, etc.

A mixer can mix analog and digital sounds depending on the type of mixer. We are making 3 channel audio mixer in which there are three microphone inputs. The microphone inputs are used to process analog signals like voice signals.

A mixer circuit can be realized by using BJTs, MOSFETs, etc. , but the best known and the most versatile type is the audio signal processing IC operational amplifier. Thus we have used op-amp technology to design the circuits.

1.1 Applications

- Recording studios
- Public address systems
- Sound reinforcement systems
- Broadcasting, television
- Film post production

1.1 Examples

- To enable a signal that originated from three separate microphones (each being used by vocalist singing a duet, or by a musical instrument perhaps) to be heard through one set of speakers simultaneously. When used for live performances, the signal produced by the mixer will usually be sent directly to an amplifier, unless that particular mixer is being connected to powered speakers.
- Radio broadcasts use a mixing desk to select audio from different sources, such as cd players, telephones, remote feeds, pre-recorded advertisements.

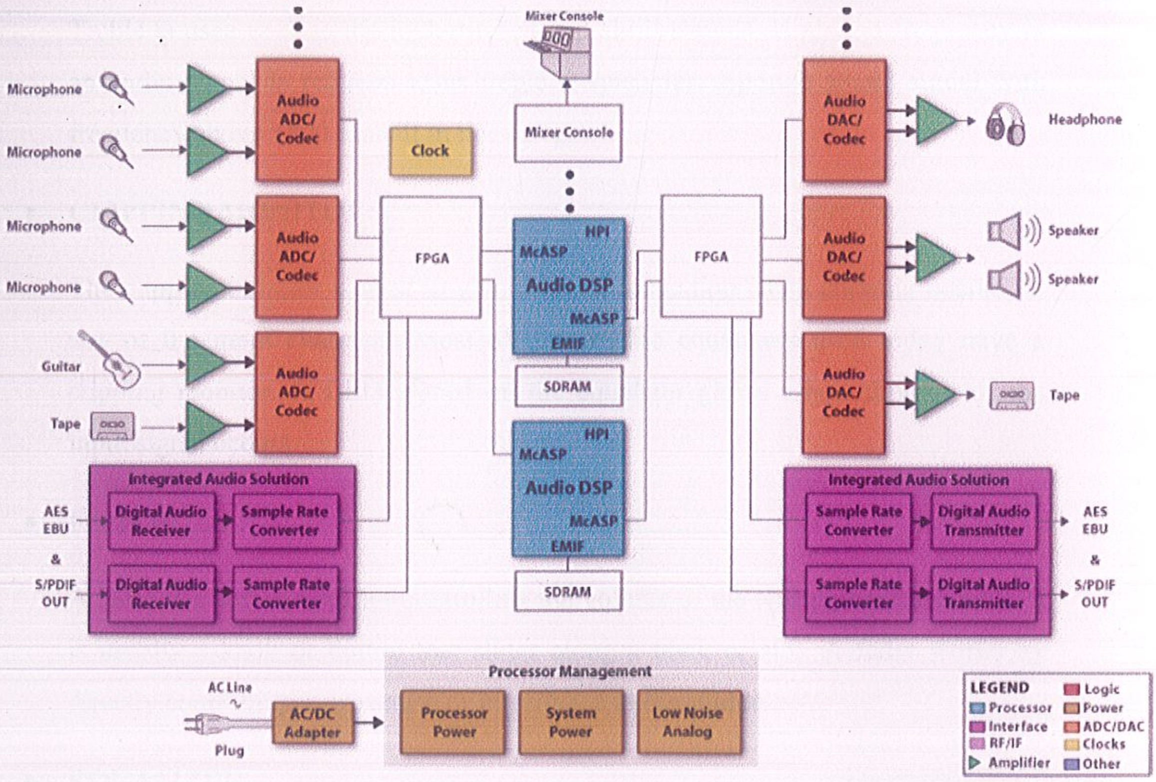


Fig. 1.1 Block Diagram of Professional Audio Mixer

1.2 Various Terms Associated with Audio Mixer

- **BASS:**

  Bass (pronounces as "base") is used to describe tones of low frequency range. A bass note produced from various instruments or singers usually comprises of sounds in the frequency range of 80 Hz – 500 Hz.

- **TREBLE:**

  Treble is used to describe high pitched and shrill sounds. High end frequency in an audio signal is referred to as treble frequencies. Audible sound signal with frequency above 1500 Hz fall in this category.

- **CLIPPING MONITOR:**

  The clipping monitor is used to give a visual indication when clipping occurs, in any of the input channels. Most of the graphic equalizers used today have a clipping monitor. A LED placed on the equalizer glows when clipping of the input signal occurs.

- **FADER:**

  A fader is a device used for controlling the volume of the channels individually. It is usually a knob or button that slides along a track or slot. A fader works by directly controlling a resistance or impedance to the source.

- **EQUALIZER:**

  An equalizer is a tone control system. Today, most systems have a graphic equalizer. It consists of a number of parallel connected, overlapping, narrow band, variable response filters that cover the entire audio spectrum, thus enabling an amplifier system's spectral response to be precisely adjusted to suit individual's needs.

- **TRANSDUCER:**

  A transducer is a device used for converting one form of energy into another. In an acoustical context, this usually means converting sound energy into electrical energy (or vice versa). For nearly all acoustic applications, some type of acoustic transducer is necessary. Acoustic transducers include loud speakers, microphones, hydrophones and sonar projectors. These devices convert an electric signal to or from a sound pressure wave.

1.3 Block Diagram of our Audio Mixer

We have divided our project construction into 4 stages. The block diagram given below consists of these four stages in order of connections made.

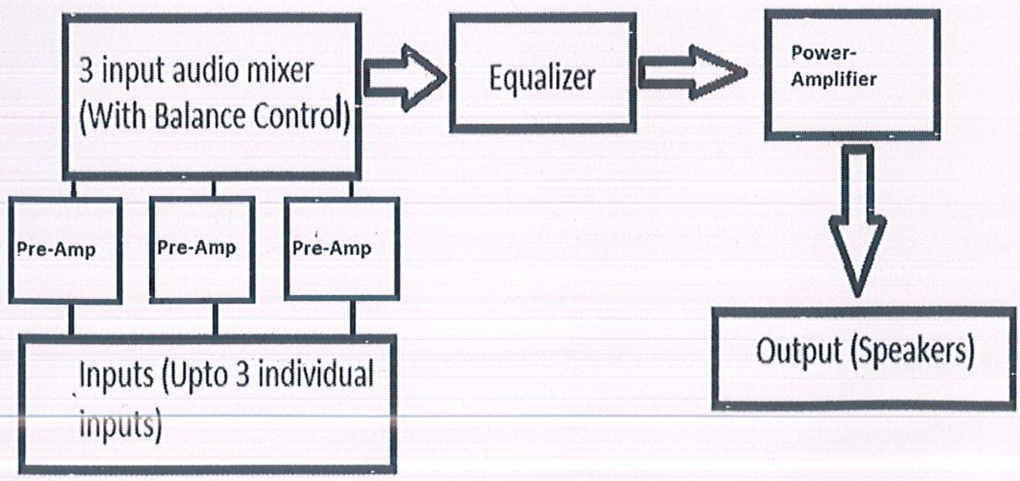


Fig. 1.2 BLOCK DIAGRAM OF AUDIO MIXER

- Pre-amplifier (used to amplify the input signal received from mic.)
- Audio Mixer
- Equalizer
- Power Amplifier

CHAPTER 2

ACOUSTICS

Sound is a very interesting phenomenon. The study of sound and its applications is called acoustics. It deals with the shape, design etc. of a room or theatre that make it good or bad for carrying sound. A sound engineer cannot work effectively without basic knowledge of electricity, acoustics and music. Only someone who is fully acquainted with all these aspects can use available sound equipment effectively and efficiently.

Owing to the rapid development in the field of electronics and electrical engineering in the past few decades or so, a number of distinct disciplines has arisen. One of these is electro-acoustics, which deals with the conversion, processing and reproduction of sound.

2.1 Some Important Terms

To understand the physics of sound it is mandatory to get familiar with a few terms. During the study of acoustics we come across many such terms. Few of these are:

- **REVERBERATIONS**

  Reverberations (reverb for short) are probably one of the most heavily used effects in music. The series of delayed and attenuated sound waves is what we call reverb and this is what creates the 'spaciousness' of a room. In reverb, for a short period after the direct sound, there is generally a set of well defined and directional reflections. These are directly related to the shape and size of the room, as well as the position of the source and listener in the room. These are **early reflections** (also called **'early echoes'**).

  After the early reflections, the rate of the arriving reflections increases greatly. These reflections are more random and difficult to relate to the physical characteristics of the room. This is called the **diffuse reverberation**, or the **late reflections.**

Another very important characteristic of reverberation is the correlation of the signals that reach your ears. In order to give a listener a real feeling of the 'spaciousness' of a big room, the sounds at each ear should be somewhat incoherent.

The reverberation of a room helps to keep the sound energy localized in the room, raising the sound pressure level and distributing the sound throughout it. If the direct sound from a source that reaches you is louder than the reflections, you are in the direct field. If, on the other hand, the sound pressure due to the reflected sound is greater than the direct sound, you are in **reverberation field**. The point at which the direct field and the reverberation field intensity are same is called the **critical distance**.

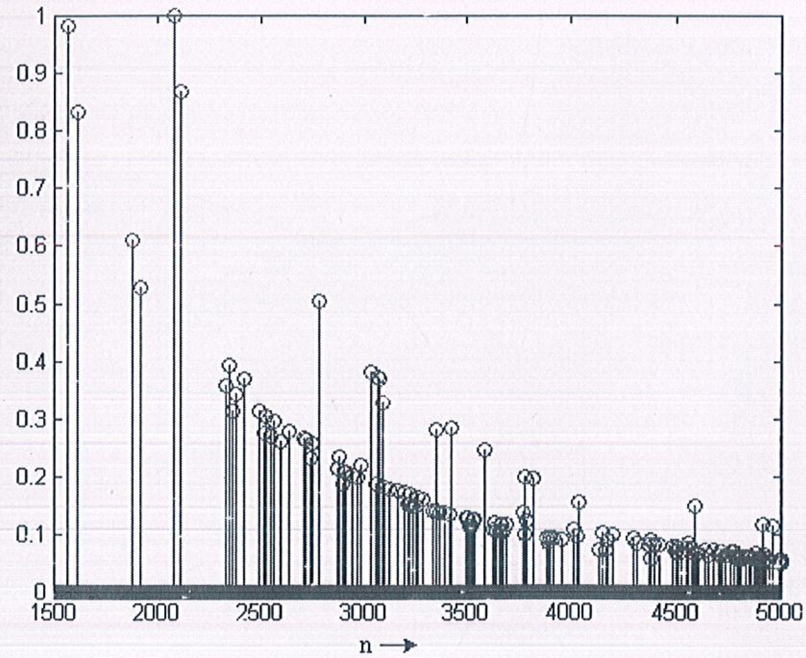


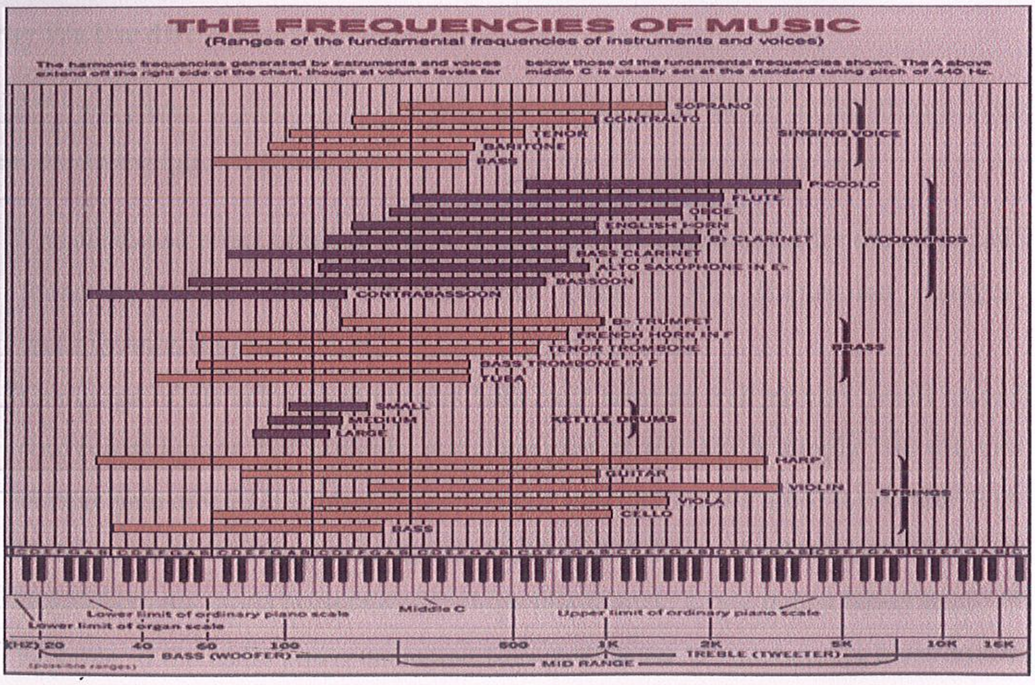
Fig. 2.1 IMPULSE RESPONSE OF A ROOM

If reverb is always around us, why do we add reverb to the recorded sounds?

Well, many tones we are listening to music, we are in environments with very little or poor reverb. Reverberations added enhance listening experience by distributing sound evenly across the room. That's why it is added.

- **SPECTRA:**

The richness of a sound or note produced by a musical instrument is sometimes described in terms of sum of a number of distinct frequencies. The lowest frequency is called the **fundamental frequency** and the pitch it produces is used to name the note. For example, in western music, instruments are normally tuned to A=440Hz. Other significant frequencies are called *overtones* of the fundamental frequencies, which may include harmonics and partials. Harmonics are whole number multiples of the fundamental frequency --- x2, x3, x4, etc. Partials are other overtones. Most western instruments produce harmonic sounds, but many instruments produce partials and harmonic tones, such as cymbals and other non-pitched instruments.



2.2 Unit Of Sound

The fundamental unit of sound is ***bel***, named after Alexander Graham Bell. 'Bel' is used to compare two powers on a logarithmic scale. For practical purposes 'bel' is quite large, therefore we use a smaller unit, named decibel (abbreviated as **dB).** Since one dB is one-tenth of one bel, we have

$$\text{Power Gain (in dB)} = 10 \log_{10}(P_1/P_2)$$

Where P_1 and P_2 represent the output and input powers of an amplifier respectively. The number of dB by which P_2 exceeds P_1, is giver by the above formula. Values of importance of gains are +3 dB and − 3 dB. These values correspond to a doubling and halving of a power, respectively.

Why We Use dB?

Usually we prefer expressing power gain (or voltage gain) in dB rather than a simple ratio due to the following reasons:

- The combined effect of many stages of amplifiers cascaded one after the other can be found by simply adding the individual gains in dB. (Logarithm changes a multiplication into an addition).
- The usage of dB helps us to express both very large as well as very small gains by convenient figures. For example, a voltage gain of 100,000 is simply 100 dB; and a voltage gain of 0.0001 (in fact, it represents a loss instead of a gain) is −80 dB.
- In many cases, the output of the amplifier is fed into a loudspeaker. Human ear response to the sound intensities on a logarithmic scale rather than a linear scale. For example, if the audio power increases from 2 W to 64 W, the hearing level does not increase by a factor of 64/2 = 32, but it increases by a factor of 6. Since $2^6 = 64$. Thus, use of dB is justified on psychological basis too.

2.3 Fundamental of Acoustics

The ultimate aim of each and every musical activity is the production of audible vibrations of the air. In general, mechanical vibrations of an elastic medium, such as air, are called sound. When these sound fluctuations lie in the audio band, they are called audible sound. The sound heard by an audience during a performance in a hall depends on how well acquainted the sound engineer is with the acoustics. The sound system installation is of lesser importance. Simple instruments and voice amplifiers, when the acoustics of hall taken into account, can give an equally well defined sound, as can be obtained with an extensive installation. This may sound odd but it can be explained by principles of sound engineering and transfer technology.

2.3.1 Vibrations and Waves

From a purely scientific point of view, sound is nothing but a wave motion, that is, a periodically changing of pressure in an elastic medium. This medium may be solid, liquid or gaseous. The distance between the particles of the medium becomes periodically greater or smaller. The particle itself vibrates around a more or less fixed position. Transportation of material does not take place. Only the sound is transported in the direction of propagation. In acoustics, it is important to know the wavelength λ of a sound vibration, given by:

$$\lambda = \frac{s}{f}$$

where s is the speed of sound in metres/second, f the frequency in hertz (Hz) and λ is wavelength in metres. The speed of sound, unlike that of light, is not constant. It depends on atmospheric pressure and temperature. At 0 °C, it is 331.6 m/s, and rises by 60 cm/s for each degree increase in temperature. Thus, at room temperature (20 °C or 68 °F), it is about 344 m/s.

CHAPTER 3
MICROPHONE AMPLIFIER

3.1 Condenser Microphones

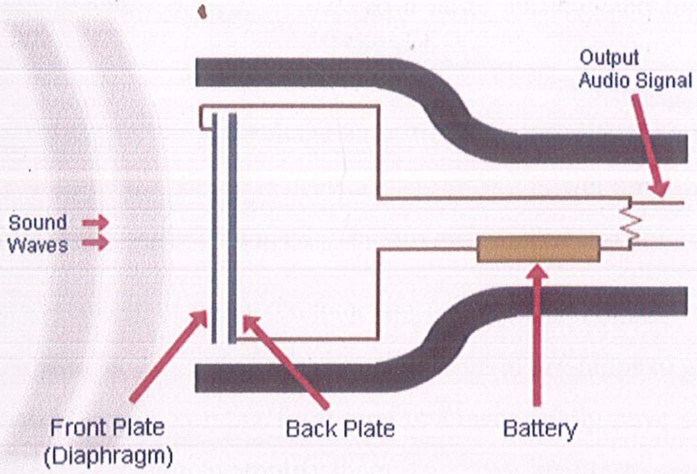
Condenser means *capacitor*, an electronic component which stores energy in the form of an electrostatic field. The term *condenser* is actually obsolete but has stuck as the name for this type of microphone, which uses a capacitor to convert acoustical energy into electrical energy.

Condenser microphones require power from a battery or external source. The resulting audio signal is stronger signal than that from a dynamic microphone. Condenser microphones also tend to be more sensitive and responsive than dynamic microphone, making them well-suited to capturing subtle nuances in a sound. They are not ideal for high-volume work, as their sensitivity makes them prone to distortion.

How Condenser Microphones Work

A capacitor has two plates with a voltage between them. In the condenser mic, one of these plates is made of very light material and acts as the diaphragm. The diaphragm vibrates when struck by sound waves, changing the distance between the two plates and therefore changing the capacitance. Specifically, when the plates are closer together, capacitance increases and a charge current occurs. When the plates are further apart, capacitance decreases and a discharge current occurs.

A voltage is required across the capacitor for this to work. This voltage is supplied either by a battery in the mic or by external phantom power.



The Electret Condenser Microphone

The electret condenser mic uses a special type of capacitor which has a permanent voltage built in during manufacture. This is somewhat like a permanent magnet, in that it doesn't require any external power for operation. However, good electret condenser mics usually include a pre-amplifier which does still require power.

Other than this difference, we can think of an electret condenser microphone as being the same as a normal condenser.

3.2 Amplifier

It is the name given in electronics to the circuit that amplifies a small voltage or current which may be alternating or steady. The small voltage or current is called input signal and the amplified voltage or current is called output signal.

Amplifiers are broadly divided into two basic categories:

1. Voltage amplifier
2. Power amplifier

- Voltage amplifier increases the voltage of the signal keeping the waveform unchanged.
- Power amplifier converts a large voltage to a large current, and the output voltage is made large.

For a microphone amplifier, we use voltage amplifiers. An amplifier's job is to take a weak signal and boost it enough so that it can drive a speaker or a power amplifier. This is because power amplifiers and tone controllers need to be driven by input signal with mean amplitudes

of tens and hundreds of millivolts which speech signals coming through condenser microphone do not possess. Some audio systems use several pre-amps to gradually build high voltage output signal. Signals coming from tape or tuner usually have suitable amplitudes. Thus they do not need a pre-amp to amplify them.

In our project, the output of microphones goes through the pre-amplifiers. Then it is fed to the summing circuit. Our aim is to make a device that amplifies the input speech signal appreciably. The input to this voltage amplifier is fed through a condenser microphone.

3.2.1 Design Considrations :

- What technique should be used to amplify this low voltage output of the condendenser microphone?
- The technique used to amplify should preserve the shape of the input signal.
- The voltage gain obtained should be stable.
- The output impedence should be very low.
- Minimum possible distortions should be there in the output signal.

Keeping in mind the above design considrations we looked for various op-amp ICs that have considrable voltage gain. These IC work in duel mode (differential input mode and single ended input mode); thus, varying the voltage gain according to the need.

We had used LM 741, the only readily available IC that is used as a pre-amp IC.

3.3 IC μ741

Table 3.1. ABSOLUTE MAXIMUM RATINGS OF μA741 IC :

Supply voltage, + V_{cc}	18 V
Supply Voltage, − V_{cc}	18 V
Differential Input Voltage, V_{id}	15 V
Input Voltage, V_I	15 V
Voltage between Offset Null and V_{cc}	15 V
Duration of Output Short-Circuit	Unlimited
Operating Free Air Temprature Range, T_A	0 to 70 ^{o}C

Table 3.2 Electrical characteristics

PARAMETER	TEST CONDITIONS	*TA**	μA741C			UNIT
			MIN	TYP	MAX	
V_{IO} (Input Offset Voltage)	$V_o = 0$	25 °C		1	6	mV
		Full Range			7.5	
$\Delta_{VIO(adj)}$ (Offset voltage adjust range)	$V_o = 0$	25 °C		±15		mV
I_{io} (Input offset current)	$V_o = 0$	25 °C		20	200	nA
		Full range			300	
I_{ib} (Input bias current)	$V_o = 0$	25 °C		80	500	nA
		Full range			300	
V_{icr} (Common mode input voltage range)		25 °C	±12	±13		V
		Full range	±12			
V_{om} (Maximum peak output voltage swing)	$R_L = 10\ k\Omega$	25 °C	±12	±14		V
	$R_L \geq 10\ k\Omega$	Full range	±12			
	$R_L = 2\ k\Omega$	25 °C	±10	±13		
	$R_L \geq 2\ k\Omega$	Full range	±10			
A_{vd} Large-signal	$R_L \geq 2\ k\Omega$	25 °C	20	200		V/mV

diferential						
A_V (Voltage amplification)	$V_o = \pm 10$ V	Full range	15			
r_i (Input resistance)		25 °C	0.3	2		MΩ
r_o (Output resistance)	$V_o = 0^{**}$	25 °C		75		Ω
C_i (Input capacitance)		25 °C		1.4		pF
CMMR (Common mode rejection ratio)	$V_{ic} = V_{icr\ min}$	25 °C	70	90		dB
		Full range	70			
K_{svs} (Supply voltage sensitivity) ($\Delta V_{IO}/\Delta V_{CC}$)	$V_{CC} = \pm 9$ V to ± 15 V	25 °C		30	150	μV/V
		Full range			150	
I_{os} (Short circuit output current)		25 °C		± 25	± 40	mA
I_{cc} (Supply current)	$V_o = 0$, No load	25 °C		1.7	2.8	mA
		Full range			3.3	
P_D (Total power dissipation)	$V_o = 0$, No load	25 °C		50	85	mA
		Full range			100	

*** All chracteristics are measured under open-loop conditions with common mode input voltage unless otherwise specified. Full range for the μA741C ic 70 °C.**

**** This typical value applies only at frequencies above a few hundred hertz because of the effects of the drift and thermal feedback.**

3.3.1 Schematics of IC-741

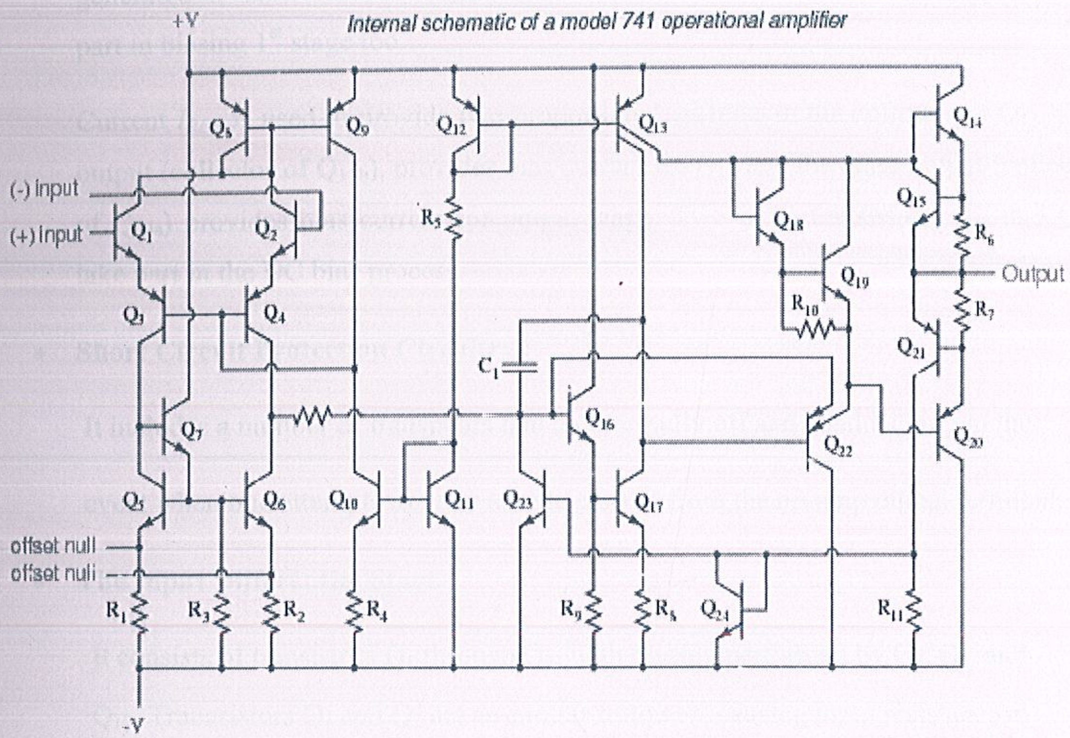


Fig. 3.1: Internal Structure of IC 741

The internal circuit of IC 741 can be explained in five stages, as follows:

- **Bias Circuit:**

  Reference current (I_{ref}) is generated in the branch at the extreme left, consisting of two diodes connected transistors Q_{11} and Q_{12} and R_4, bias current for 1st stage is generated in the collector of Q_{10}. Another current mirror formed by Q_8 and Q_9 takes part in biasing 1st stage too.

  Current I_{REF} is used to provide two proportional currents in the collector of Q_{13}. One output (collector of Q_{13b}), provides bias current for Q_{17} and the other output (collector of Q_{13a}) provides bias current for output stage. Two more transistors, Q_{18} and Q_{19}, take part in the DC bias process.

- **Short Circuit Protection Circuitry :**

  It includes a number of transistors that are normally off and conduct only in the event when one attempts to draw a large current from the op-amp output terminal.

- **The Input Differential Stage :**

  It consists of transistors Q_1 through Q_7, with biasing performed by Q_8, Q_9, and Q_{10}. Tranasistors Q_1 and Q_2 act as emitter followers causing input resistance to be high.

  delivering the input differential signal to the differential common base amplifier formed by Q_3 and Q_4.

  Transistors Q_5, Q_6 and Q_7, and resistors R_1, R_2 *and* R_3 form the load circuit of the input stage. This is an elaborate current mirror load circuit which not only provides a high resistance load but also converts the signal from differential to single ended form with no change in CMRR. The output is taken single-endedly at the collector of Q_6.

  In IC741, level shifting is done in the 1st stage using the lateral p-n-p transistors Q_3 and Q_4. It has an added advantage of protection of input stage transistors Q_1 and Q_2

against emitter-base junction breakdown.

- **Intermediate Single Ended High Gain Stage :**

It's composed of Q_{16}, Q_{17}, Q_{13b} and the two resistors R_8 and R_9. acts as an emitter folower, thus giving the second stage a high input resistance. This minimizes the loading on the input stage and avoids loss of gain.

Q_{17} acts as a common emitter amplifier with a 100 Ω resistor in the emitter. It's load is composed of the high output resistance of the p-n-p current source Q_{13b} in parallel with the input resistance of the output stage. Using a transistor current source as a load resistance (active load) enables one to obtain high gain without resorting to the use of large load resistance.

The output of the second stage is taken at the collector of Q_{17}. Capacitor C_c is connected in the feedback path of the second stage to provide fequency compensation using the Miller compensation technique.

- **Output Buffering Stage :**

The IC μA741 uses an efficient output circuit known as a class AB output stage. It consists of the complementary pair Q_{14} and Q_{20}, where Q_{20} is a substrate p-n-p. trnasistors Q_{18} and Q_{19} are fed by current source Q_{13A} and bias the output transistors Q_{14} and Q_{20}. Transistors Q_{23} acts as an emitter follower , thus minimizing the loading effect of the output stage on the second stage.

3.4 Working

A microphone converts the air pressure variations produced by a voice into a electrical variations of same frequency and amplitude,i.e.,it converts sound energy into electrical energy which is an ac signal here. While using the microphone its impedence should be kept in mind as bad impedence matching is one of the reasons for additional noises being introduced into the circuit. The impedence of the microphone used should be lesser than the impedence of the pre-amplifier used. This reduces distortions.

The ac signal then develops an alternating voltage across resistance R. The condenser microphone is of high quality but its voltage output is very low. Therefore, we require an amplifier to amplify the output signal of the condensor microphone. The output of condenser microphone is then fed into summing amplifier which is then fed into power amplifier.

3.4.1 Features

The amplifier we are using is a linear amplifier, i.e. the amplifier preserves the shape of the input signal. For instance, if the input signl is sinusoidal the output should also be sinusoidal. We are using an op-amp with single input supply. This is so because the negative feedback stabilizes the voltage and increase the resistance. Because the voltage feedback, the amplfier has very low output impedence and the negetive feedback also decreases the distortions and noise introduced by the amplifier in the output signal. Voltage feedback increases the bandwidth of the amplifier as well.

The main advantage of the inverting voltage is its ability to handle more than one input at a time.

3.4.2 Adjustable Inverted Gain

When the adjustable resistor is introduced at zero, the non inverting non inverting input grounded to get a max. input gain of $-{}^{R_f}/_{R_s}$. When the adjustable resistor is increased to R_f , equal voltages drive the non-inverting and inverting inputs. So because of common mode rejection, the output voltage is aprrox. zero. The circuit has an adjustable closed loop voltage gain from approx. 0 to $-{}^{R_f}/_{R_s}$.

[*CMMR*- It is the ratio of differential gain to common mode gain (${}^{A_d}/_{A_c}$). For ideal op-amp, the *CMMR* is infinity, but practically op-amp has *CMMR* of about 90dB.]

3.4.3 Design of Pre-amplifier

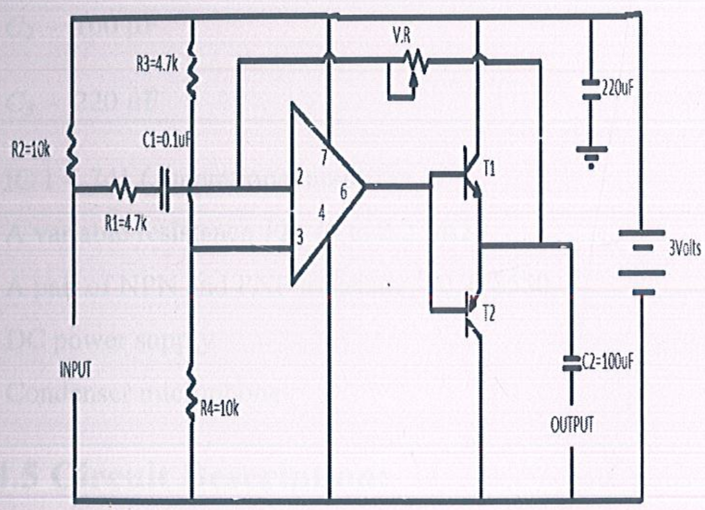


Fig. 3.2 Circuit Diagram of Pre-Amplifier

3.4.4 List of components used for amplifier

- **Resistances**

  R_1 - 4.7 kΩ

  R_2 - 10 kΩ

  R_3 - 4.7 kΩ

  R_4 - 10 kΩ

- **Capacitors**

  C_1 - 0.1 μF

  C_2 - 100 μF

  C_3 - 220 μF

- IC 1 - 741 (Operational amplifier)
- A variable resistance VR_1 (0 to 2.2 MΩ)
- A pair of NPN and PNP transistor, 8050/8550
- DC power supply
- Condenser microphone

3.4.5 Circuit Description:

The sound signals are picked up by the condenser microphone is converted into electrical variation which then passes through resistance R_1 and capacitor C_1 into pin 2 of the IC. These signals are amplified by the op-amp IC741 which is used in the inverting mode with a single supply. IC741 is triggered by the signal coming from the voltage divider using divider network of resistors R_3 and R_4 into pin 3. The gain of IC 1 can be set by varying the feedback through R_3 resistance. Here the output of IC coming out of pin 6 is further amplified by the push-pull amplifier using transistors 8050/8550. The microphone should be placed near the circuit with shielded wire to suppress hum. The output is then taken from emitter of two transistors and passes through capacitor C_2 which acts as filter. The output of the capacitor is then used to drive the power amplifier.

CHAPTER 4

SUMMING AMPLIFIER

4.1 Introduction

A summing amplifier is a voltage amplifier that combines all the audio input signals coming out of the pre amplifier stage. A summing amplifier acts as a mixer. This stage of the mixer can be a fixed gain stage or a variable gain stage. The output can be split into two depending upon the power amplifier. If the power amplifier can work in a dual mode i.e. has a stereo output, we split the output of the summing amplifier stage; else we take a single output to the next stage.

The voltage output of a summing amplifier is equal to the algebraic sum of two or more voltages each multiplied by the gain factor.

$$V_{op} = -\left\{ {R_f}/{R_1} + {R_f}/{R_2} + \cdots + {R_f}/{R_i} \right\}$$

Where R_f - resistance of negative feedback circuit

R_i - resistance of i^{th} input

The overall negative sign is unavoidable because we use the inverting input terminal. The adjustable resistors allow us to adjust the combined volume.

4.1.1 Functions of Summing Amplifier

- Mixing
- Fader(used for suppression of undesired channels)
- Master volume controller(controls volume of the entire system)

4.2 Diagram:

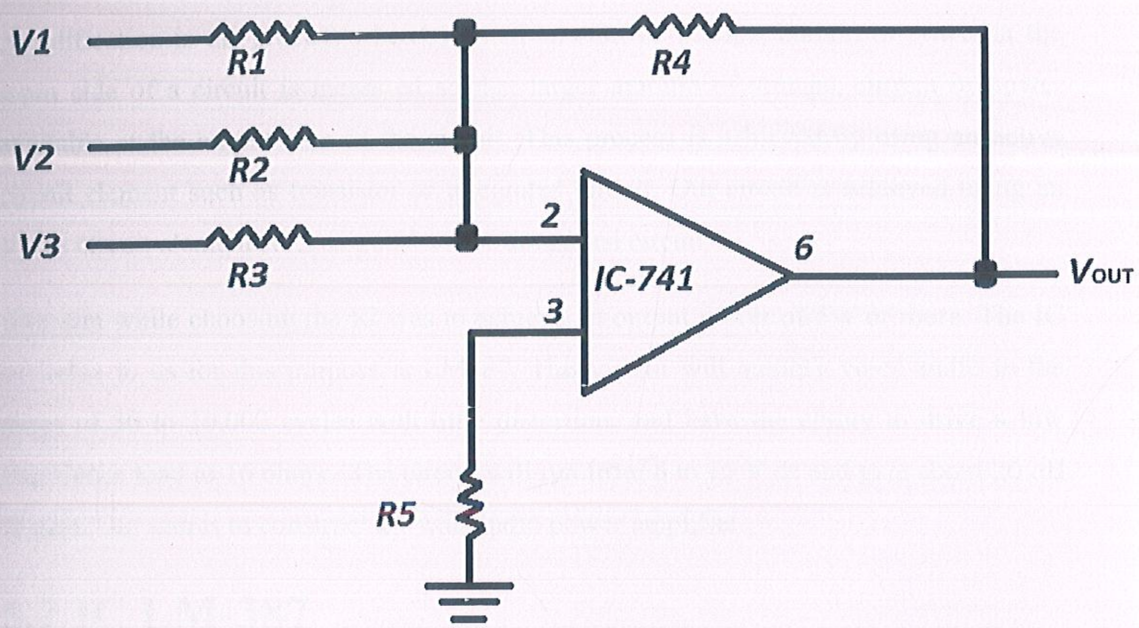


FIG 4.1. Circuit Diagram of Summing Amplifier

CHAPTER 5
POWER AMPLIFIER

5.1 Introduction

Amplification is the process whereby small amount of voltage, current or power at the input side of a circuit is increased so that larger amount of voltage, current or power available at the output side of the circuit. This process is achieved by using an active circuit element such as transistor or integrated circuit. Our circuit is achieved using an active circuit element such as transistor or integrated circuit.

Our aim while choosing the IC was to achieve an output power of 5W or more. The IC available to us for this purpose is LM387. This circuit will amplify voice audio in the range of 50 to 10,000 cycles with little distortion, and have the ability to drive a low impedance load to 16 ohms. The circuit will run from 6 to 15 V dc and give about 20 dB of gain. Our aim is to construct a 7 watt audio power amplifier.

5.2 IC LM-387

It offers:

- Higher output power.
- Low noise.
- Polarity inversion protection.
- Fortuitous open ground protection.
- High supply voltage rejection (20 dB minimum).

5.3 Concept:

To build an inexpensive audio amplifier with little parts count that is very reliable and easy to build and implement.

Table 5.1 Absolute Maximum Ratings

V_S	Supply voltage	20 V
I_o	Output peak current (non repetitive)	4 A
I_o	Output peak current (repetitive)	3 A
P_{tot}	Power dissipation at $T_{amb} \leq 80\ ^{o}C$	1 W
	$T_{tab} \leq 90\ ^{o}C$	5 W
$T_{stg,}\ T_j$	Storage and junction temperature	$-40\ ^{o}C$ to $150\ ^{o}C$

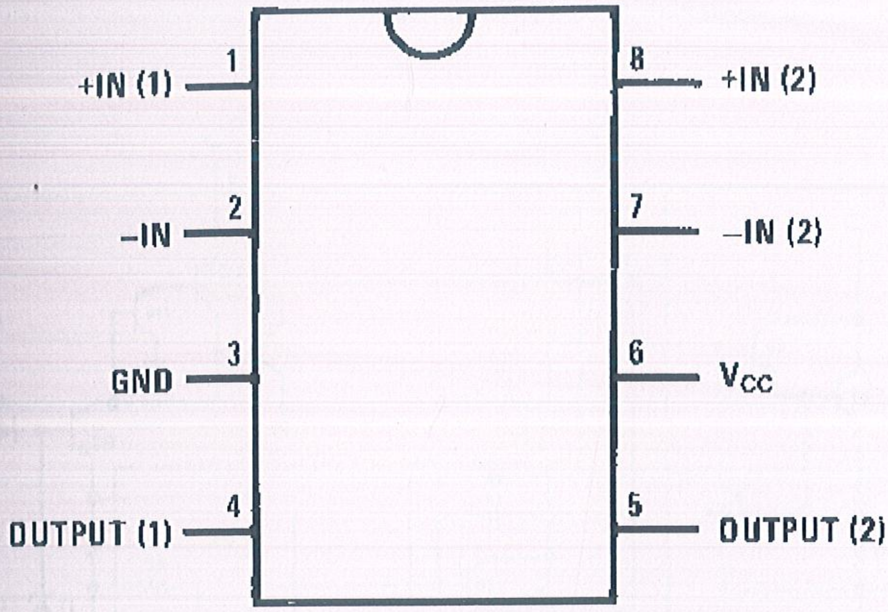


Fig 5.1 Pin Structure of LM387

The above figure shows the top shows the top view of LM-387.

5.2.2 Schematics of LM-387

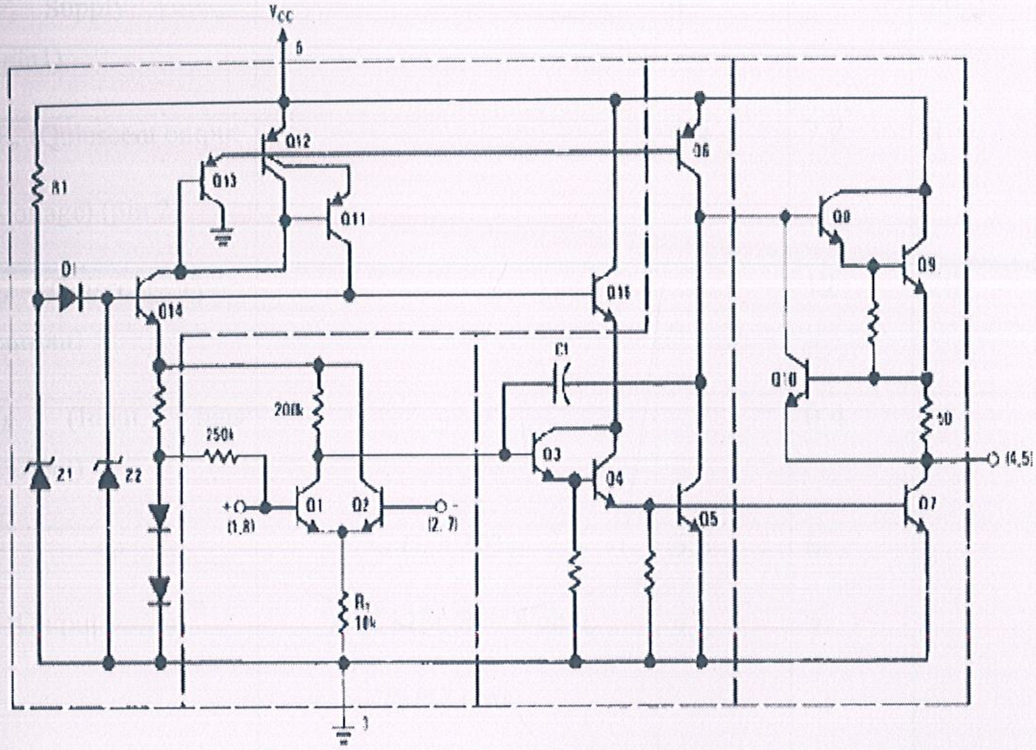


Fig. 5.3 Internal Structure of LM-387

TABLE 5.2 Electrical Characteristics (V_S = 14.4 V, T_{AMB} = 25 °C)

Parameter	Test conditions	Min	Typ.	Max.	Unit
V_s Supply voltage (pin1)		4		20	V
V_o (Quiescent output Voltage) (pin 2)		6.4	7.2	8	V
I_d (Quiscent drain current)			12	20	mA
I_b (Input bias current)			0.4		μA
P_0 (Output power)	D_L = 10% f = 1 kHz, R_L = 4 ohm, R_L = 2 ohm	5.5 5.5	6 7		W W
$V_{i(rms)}$ (Input saturation voltage)		220			mV
R_i (Input resistance)			5		MΩ

B (frequency Response) (−3dB)	$R_L = 4\ \Omega / 2\ \Omega$ $C = 820$ pF $C = 150$ pF	40 to 20,000 40 to 10,000			Hz Hz
D (Distortion)	$P_o = 50$ mW to 2.5 W $R_L = 4\ \Omega/2\ \Omega$ $f = 1$ kHz		0.3		%
G_{vo} (Voltage gain) (open loop)	$R_L = 4\ \Omega$ $f = 1$ kHz		80		dB
G_{VC} (Voltage gain) (Closed loop)	$R_L = 4\ \Omega / 2\ \Omega$ $F = 1$kHz		37	40	dB
E_{en} (Input noise Voltage)	$V_s = 16$ V $B(-3\text{dB}) = 40$ to 15000 Hz		80		pA
Efficiency	$P_o = 6$ W $R_L = 4\ \Omega$ $F = 1$ kHz		75		%
SVR	$R_L = 4\ \Omega$, $V_{ripple} = 1V_{rms}$	40	48		dB

5.4 Test and Application Circuit

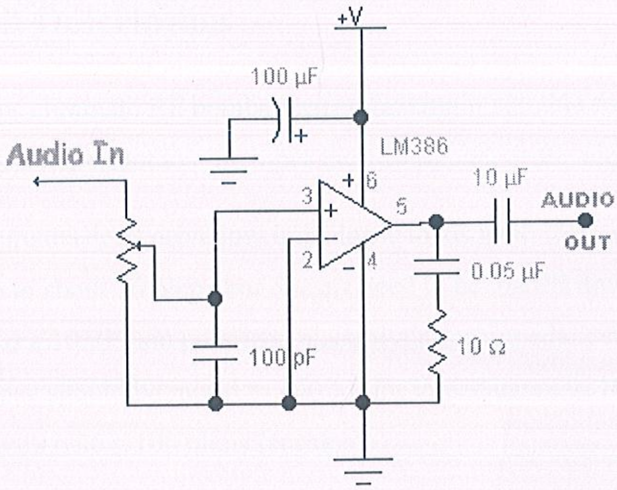


Fig 5.4 Circuit diagram

5.4.1 Components Used

- IC-1 LM-387
- Resistors
  - ❖ $R = 10\ \Omega$
- Capacitors
  - ❖ $C = 100\mu F/6.4V$
  - ❖ $C = 100pF$
  - ❖ $C = 10\mu F$

- $C = 0.05\ \mu F / 6.4\ \mu$

- Speakers (4Ω)

5.4.2 Circuit Description

The 100 pF helps eliminate RF bombardment, making it suitable for higher RF environments. It's important to remember the 100 µF cap or the circuit will oscillate.

Many op-amp circuits don't drive low impedance loads well, this circuit will handle a load impedance to about 16 ohms but doesn't need to be loaded down to that impedance. The output cap is a 10 µF non polarized electrolytic for impedance's to 600 ohms. The output capacitance should be raised to 100 µF for impedance's to 100 ohms and to 1000 µF for impedance's below 100 ohms (speaker.)

Operating bandwidth is from 50 Hz to 10 kHz. Over 10 kHz and the unit suffers from poor slew rate, causing distortion. Actually, for this purpose slew rate limitations work to our advantage as it helps make the amplifier less RF susceptible. Increased audio amplification can also be had with the addition of a feedback loop.

5.4.3 Characteristics of Amplifier

- Supply voltage = 16 V
- Maximum output power = 7 W
- Input sensitivity = 14.6 mV
- Input sensitivity (at maximum power) = 155 mV
- Zero signal current gain = 9.5 mV
- Supply current at maximum output power = 380 mV

- Frequency response
    - ❖ 1dB bandwidth = 40 Hz to 20 kHz
    - ❖ 3dB bandwidth = 20 Hz to 30 kHz
- Tone control response
    - ❖ Bass @ 100 Hz = 0 to + 8 dB
    - ❖ Treble @ 10 kHz = −13 dB to 0 dB
- Distortion at 3 W for various frequencies
    - ❖ 60 Hz 0.4%
    - ❖ 1 kHz 0.4%
    - ❖ 10 kHz 0.8%
- Input impedance

5.4.4 Tone Control

The frequency compensation normally required with such types of systems like treble cut and bass boost are catered in this circuit. While the required treble cut is achieved with a passive network at the input of the amplifier, the bass boosting is achieved in the active mode by judicious selection of RC network in the feedback loop of the amplifier.

CHAPTER 6

POWER SUPPLY

6.1 Introduction

In alternating current, the electron flow is alternate, that is the electron flow increases to maximum in one direction, decreases back to zero. It then increases in the other direction and then decreases to zero again. Direct current flows in one direction only. Rectifier converts alternating current to flow in one direction only. When the anode of the diode is positive with respect to its cathode, it is forward biased, allowing the current to flow. But when it's anode is negative with respect to the cathode, it is reverse biased and does not allow the current to flow. This unidirectional property of diode is used for rectification.

For the construction of the rectifier, the property of the capacitors to oppose any change in the voltage applied across them by storing energy in the electric field and of inductors to oppose any change in the current flowing through them by storing energy in the magnetic field of coli may be utilized. To remove pulsation of the dc current obtained from the rectifier, different types of combinations of capacitor, inductor and resistors may be used to increase action of the filtering.

6.2 Need for a Power Supply

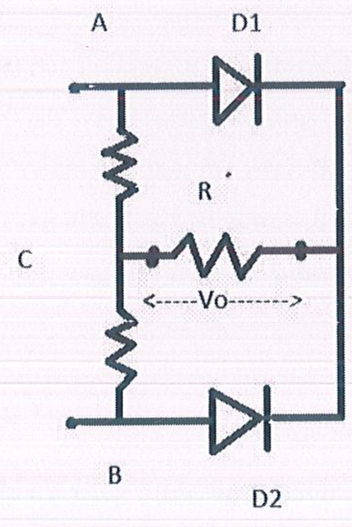
Power supply is a primary requirement for every circuit to work. Power supply can eliminate or replace the batteries of solid state electronic equipment and the equipment thus can be operated by 220 V ac mains instead of the batteries or dry cells. Now days, the use of commercial battery eliminator or power supply unit has become increasingly popular as power source for house-hold appliances like trans receiver, record player, cassette players etc.

6.3 Theory

Rectification is a process of rendering an alternating current or voltage into an unidirectional one. The component used for rectification is called rectifier. A half wave rectifier permits current to flow only during the positive half cycle of the applied ac voltage by eliminating the negative half cycle of the applied ac voltage. Thus the pulsating dc is obtained. To obtain smooth dc power, additional filter circuits are required. In our circuit, we have used a full wave rectifier. A full wave rectifier converts the positive half cycle as well as negative half cycle of applied ac voltage into dc.

6.3.1 Full Wave Rectifier

It is possible to rectify both alternations of the input voltage by using two diodes in the circuit arrangement. Assume 6.3 V rms (18 V P-P) is applied to the circuit. Assume further that two equal valued series connected resistors *R* are placed in parallel with the ac source.



The 18 Vp-p appears across the two resistors connected between points AC and CB, and point C is the electrical mid-point between A and B. Hence 9 Vp-p appears across each resistor. At any moment during a cycle of V_{in}, if point A is positive relative to C, point B

is negative relative to C. when A is negative to C, point B is positive relative to C. The voltage applied to the anode of each diode is equal but opposite in polarity at any given instant.

When A is positive relative to C, the anode of D_1 is positive with respect to its cathode. Hence D_1 will conduct but D_2 will not. During the second alternation, B is positive relative to C. the anode of D_2 is therefore positive with respect to its cathode and D_2 conducts while D_1 is cut off.

There is conduction by either D_1 or D_2 during the entire input voltage cycle. Since the two diodes have a common cathode load resistance R_L, the output voltage across R_L will result from the alternate conduction of D_1 and D_2. The output waveform V_{OUT} across R_L, therefore has no gaps as in the case of half wave rectifier.

The output of a full wave rectifier is also pulsating direct current. In the diagram, the two equal resistors R across the input voltage are necessary to provide a voltage mid-point C for circuit connection and zero reference. Note that the load resistor R_L is connected from the cathodes to this center reference point C.

An interesting fact about the output waveform V_{OUT} is that its peak amplitude is not 9 V as in the case of the half wave rectifier using the same power source, but is less than 4.5 Volt. The reason, of course, is that the peak positive voltage of A relative to C is 4.5 V, not 9 V, and part of the 4.5 V is lost across R. Though the full wave rectifier fills in the conduction gaps, it delivers less than half the peak output voltage that results from half wave rectification.

6.3.2 Filtration

The rectifier circuits we have discussed above deliver an output voltage that always has the same polarity but, however, this output is not suitable as dc power supply for solid state circuits. This is due to the pulsations or ripples of the output voltage. This smoothing is done by incorporating filter networks. The filter network consists of inductors and capacitors. Filter network circuits may be of four types in general:

- Inductive filter
- Capacitive filter
- Choke input filter
- Pi filter

The dc along with ac ripples from the rectifier circuit starts charging the capacitor C to about peak value. A small ripple is still present in the output of dc which may be reduced by adding additional filter network in series. In our project we have used a shunt capacitor filter.

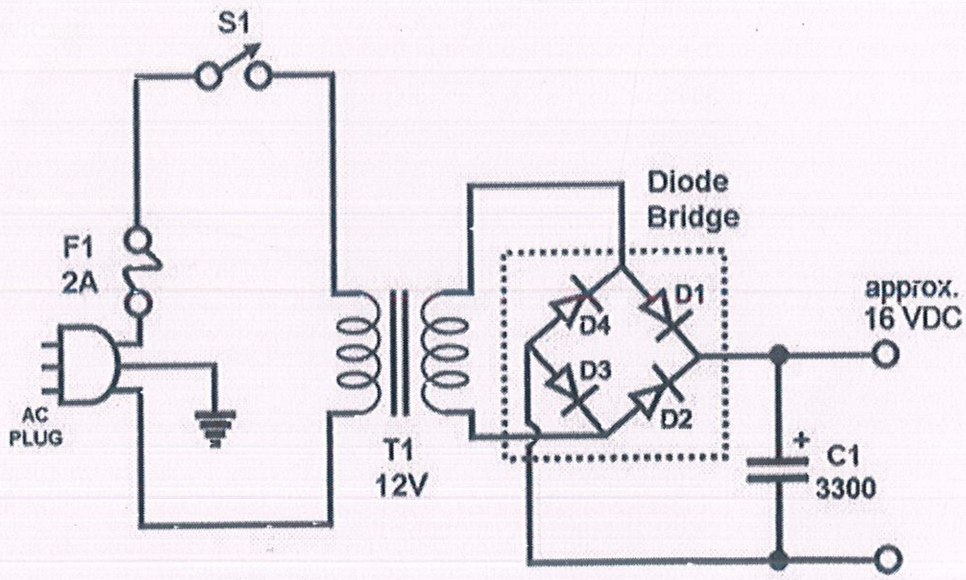


Fig. 6.1 Circuit Diagram of Power Supply

CONCLUSION AND RESULTS

While working on the project, we came across the world of acoustics and audio circuits. Theoretical study on the above mentioned topics was an easy task, but implementation of various audio circuits took us for a ride. The hurdles we went through were:

- The availability of components resulted into wild search.
- Circuit designing demanded our in depth skills.
- Practical examining whether a success or a failure, motivated us to do different and to do more.

But finally, after a lot of testing and choosing, we were able to successfully implement the hardware with near about desired result. Through this project, we gained immense knowledge.

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