DESIGN AND IMPLEMENT A LOW COST AUDIO AMPLIFIER

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Universiti Malaysia Sarawak
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APPROVAL SHEET

This project report attached here to, entitled "Design And Implement a Low Cost Audio Amplifier", prepared and submitted by Azman Bin Yaakub in partial fulfillment of the requirements for the degree of Bachelor of Engineering (Electronic Telecommunication) is hereby accepted.

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Dedicated to my
parents, brothers,
loves one & friends.

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ABSTRACT

Nowadays, the high performance stereo audio amplifiers have been widely used according to the technology. The system can be found in the market in various forms and advantages. The considerable advantages of this system are in terms of its output sound and application to many sound input sound devices. The purpose of this thesis is to build a low cost audio stereo amplifier. The characteristics and factors that have effect on the specifications of the system are discussed. Method and problems that arise in constructing it are also discussed.

ABSTRAK

Pada masa kini, system penguat audio stereo berkuasa tinggi yang berlandaskan teknologi terkini telah banyak digunakan. Sistem ini boleh didapati di pasaran dalam pelbagai bentuk dan kegunaan. Kelebihan yang dimiliki oleh setiap sistem dikira dari segi keluaran outputnya dan aplikasinya bersama dengan sistem lain. Tujuan laporan ini dibentangkan adalah untuk menerangkan pembuatan sebuah penguat audio stereo kos rendah. Kriteria-kriteria dan factor yang mempengaruhi spesifikasi sistem tersebut dibincang. Metod dan masalah yang dihadapi semasa pembuatannya juga dibincang.

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Preface

Power Amplifier

The purpose of power amplifier is to deliver a large amount of power to a load. The load here is most often a loudspeaker, which requires considerable power to convert electrical signals to sound waves.

For a power amplifier to work or to perform its function, it must be capable to dissipate large amount of power so that the heat produced from high current and voltage is released to the surroundings quickly. This is important to prevent temperature buildup before it disrupts the amplifier's operation.

That is why the author design the circuit in such a way that the surface areas is large to enhance heat transfer to the surroundings, even though the amplifier's component is bulky.

Power amplifiers are widely used in audio components such as radio, television audio receivers, stereos and recording-studio equipment. Power amplifiers are also used in electromechanical control system to drive electric motors. Examples are computer disk and tape drives, robotic manipulators, autopilots, antenna rotators, pumps and motorized valves, and process controllers of all kinds.

In this report, the author focused on only one thing, which is an audio power amplifier.

CHAPTER 1

INTRODUCTION

In this chapter explains about the basic concepts of amplifiers.

1.1 The characteristic Of Ideal Amplifier

An ideal operational amplifier has the following attributes:

1. It has infinite gain
2. It has infinite input impedance
3. It has zero output impedance

Although no real amplifier can satisfy these requirements, most modern amplifier has large gains and input impedances, small output impedances and negligibly small error results from assuming ideal characteristics.

A detailed study of the ideal amplifier is beneficial in terms of understanding how amplifiers are used as well as building some important theoretical concepts that have implications in many areas of electronics.

Figure 1.1 shows the standard symbol for an operational amplifier. Note that symbol shows (+) for noninverting and (-) for inverting.

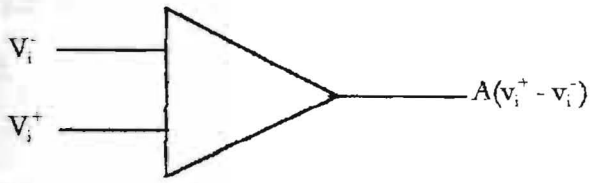

Figure 1.1 Symbol Of Operational

CHAPTER 2

In this chapter, we will discuss on classes of amplifiers which are Class A, Class B, Class C and Class D.

2.1 Audio Amplifier Classifications

2.1.1 Class-A Amplifiers

Class-A amplifiers are amplifiers that are biased and during operation, the output never saturates or cuts off. The characteristic of an amplifier in class A is its output remains in the active region during a complete cycle (one full period) of a sine wave input signal. Figure 2.1 shows a typical class-A amplifier and its input and output waveforms [1].

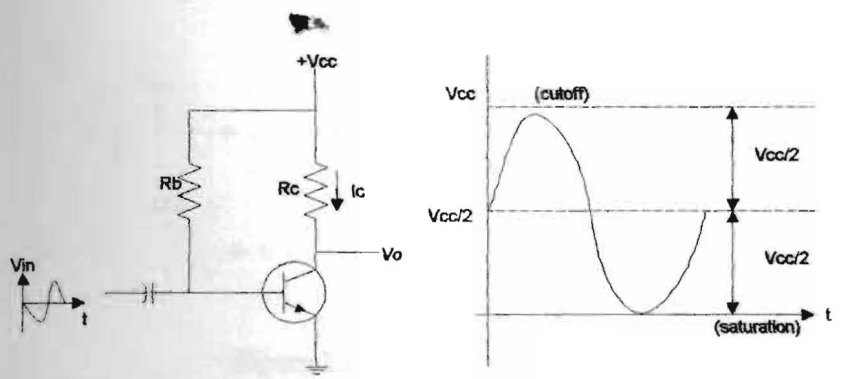

Figure 2.1 Class-A amplifier

The output of a class-A amplifier remains in the active region during a full period (360°) of the input sine wave. From the figure, the transistor output is biased midway between saturation and cutoff.

The efficiency of a power amplifier is defined to be

$$\eta = \frac{\text{average signal power delivered to load}}{\text{average power drawn from dc source(s)}}$$

Advantage of class-A amplifier is that it generally produces less signal distortion than some efficient classes of amplifiers. Disadvantage of class-A amplifier is it is less efficient, which is approximately only ¼ of total power consumed by the circuit is delivered to the load, under optimum conditions.

2.1.2 Class-B Amplifiers

Class-B amplifiers are amplifiers that their output current varies during only one half-cycle of a sine-wave input. It means that amplifier is active during a positive half-cycle or only during a negative half-cycle of the input. This operation is illustrated in Figure 2.2 [1].

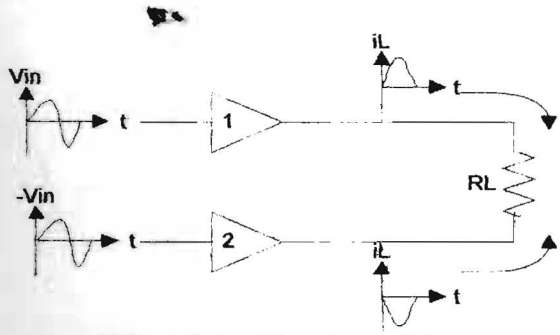

Figure 2.2 Class-B amplifier

The principle of class-B, a push-pull operation. Output amplifying device 1 drives current i_L through the load in one direction while 2 is cut off, and device 2 current in the opposite direction while 1 is cut off.

Class-B operation produce an output waveform that is only half-wave rectified and this makes it highly distorted. The waveform is not suitable for audio applications.

There are two transistors that are operated class B. One is to amplify positive signal and the other one is to amplify negative signal. The output is the combination of both of that waveform. These amplifiers are more efficient and more widely used in power applications than class-A amplifiers.

The advantage of class-B amplifier is that it can achieve higher efficiency than can be attained by class-A amplifier.

2.1.3 Class-C Amplifier

Class-C amplifiers are amplifiers that their output conducts load current during less than one-half cycle of an input sine wave. Figure 2.3 shows a class-C amplifier [1].

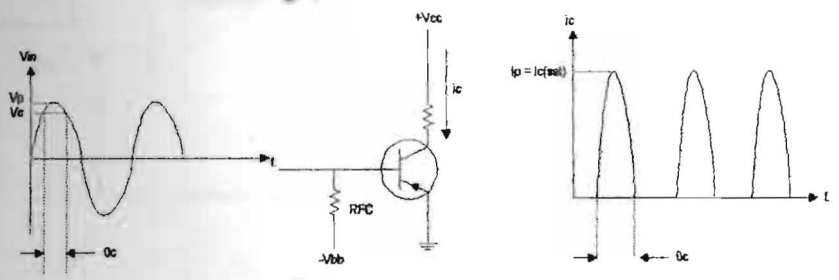

Figure 2.3 Class-C amplifier

A class-C amplifier has a highly distorted input. It could not be used in an application, such as an audio amplifier. Class-C amplifiers are used primarily in high-power, high frequency applications, such as radio-frequency transmitters.

The advantage of class-C amplifier is that it has a very high efficiency compared to other classes of amplifiers before this.

2.1.4 Class-D Amplifiers

Class-D amplifiers are amplifiers that their output is switched on and off. The output is in its linear range for zero time during each cycle of an input sine wave. The only time that a class-D output device is in its linear region is during that short interval. It required switching from saturation to cutoff, or vice versa.

Class-D amplifiers and its application is ideally suited for VMOS transistors. Figure 2.4 is the block diagram for Class-D [1].

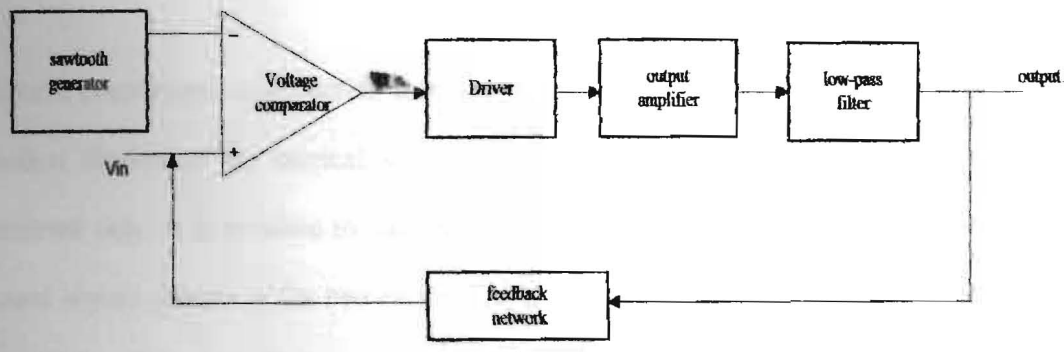

Figure 2.4 Class-D amplifier

A fundamental component of a class-D amplifier is a pulse-width modulator. It produces pulses, which are proportional to the amplifier's input signal. When the signal level is small, a series of narrow pulses is generated, and when the input level is large, a series of wide pulses is generated.

The advantage of a class-D amplifier is it has a very high efficiency, nearly 100%. The disadvantages are it needs a very good low-pass filter and a high-speed switching of heavy currents generates noise through electromagnetic coupling, called electromagnetic interference.

2.2 Stereo Sound

Since this project is to build an audio power amplifier then we must observe some facts about stereo sound techniques. The true meaning of stereo comes from the Greek word meaning 'solid', or 'three dimensional'. Since this audio power amplifier is using two channels with two loudspeakers then we are considering about a two-channel stereo reproduction principles.

In most cases stereo reproduction from two loudspeakers can only hope to achieve a modest illusion of the original sound field, since reproduction is from the front quadrant only. It is possible to state that the best illusion will be created when the sound signals present at the two ears are as similar as possible to those perceived in natural listening, even if one cannot entirely rationalize psychoacoustics mechanism involved.

It is possible to create this illusion using either time differences between the speaker outputs or using level differences between them. It is also possible to use a combination of the two, although there is a problem with 'time difference' stereo in that contradictions may arise between transient and continuous sounds. The main point to be considered with loudspeaker reproduction is that both ears receive the signals from both speakers, which is different from headphones where both ears only receives one signal channel.

The result of this is that the loudspeaker listener sited in a center seat receives at his left ear the signal from the left speaker first, followed by that from the right speaker, and at his right ear the signal from the right speaker first, followed by that from the left speaker, the time t being the time taken for the sound to travel the extra distance from the more distant speaker. Figure 2.5 shows the principal of two-channel stereo [2].

If the two speakers differ only in level and not in phase then it can be shown that the vector summation of the signals from the two speakers at each ear results in two signals, which, for a given frequency, differ in phase angle proportional to the relative amplitude of the two signals.

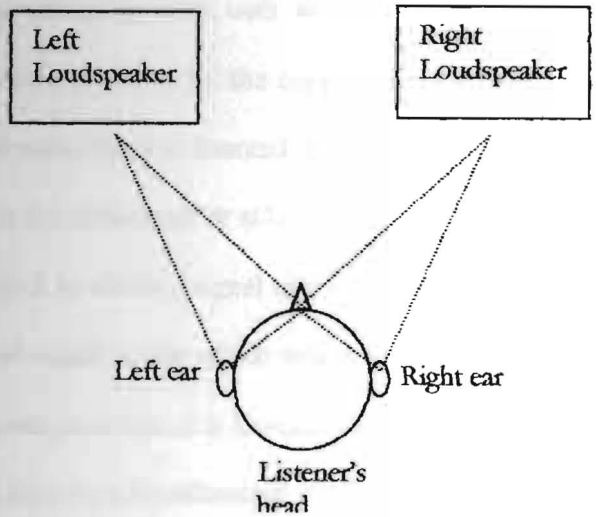

Figure 2.5 Two-channel stereo

2.2.1 Basic Stereo Signal Formats

In a basic stereo signal, we must consider about the nature of the stereo signal together with the definitions of the terms used to describe the various formats of stereo sound.

2.2.2 A, B, M and S signals.

In broadcasting technology, the left (L) channel is always refer as the 'A signal' and the right (R) channel as the 'B signal'. In some of the stereo microphones they are called 'X signal' and the 'Y signal' respectively. In a two channel stereo system the A signal feeds the left loudspeaker and the B signal feeds the right loudspeaker.

It is sometimes convenient to work with stereo signals in the so-called 'sum and difference format', since it allows for the control of image width and ambient signal balance. The sum or main signal is denoted by 'M', and is based on the addition of L and R signals, whilst the difference or side signal is denoted 'S' and is based on the subtraction of R from L to obtain a signal which represents the difference between the two channels. The M signal is that which would be heard by someone listening to a stereo program in mono, and thus, it is important in situations where the mono listener must be considered, such as in broadcasting.

2.2.3 Effects of misalignment on stereo signals.

Differences in level, frequency response and phase may arise between signals of a stereo pair, perhaps due to losses in cables, misalignment and performance limitations of equipment. It is important that these are kept to a minimum for a stereo work, as interchannel interference result in various audible side effects.

Differences will also result in poor mono compatibility. The differences and their side effects are discussed as below.

a) Frequency response and level.

A difference in level or frequency response between A and B channels will result in a stereo image biased towards the channel with the higher overall level or that with the better high-frequency response. Also, an A channel with excessive high-frequency response compared with that of the B channel will result in the clear movement of sibilant sound towards the A loudspeaker.

Level and response misalignment on MS signals will result in increased crosstalk between the equivalent A and B channels, such that if the S level is too low at any frequency the AB signal will become more monophonic (width narrower), and if it is too high the apparent stereo width will be increased.

b) Phase

Interchannel phase interference will affect one's perception of the positioning of sound source, and it will also affect mono compatibility. Phase differences between A and B channels will result in 'comb-filtering' effects in the derived N signal due to cancellation and addition of the two signals at certain frequencies where the signals are either out of or in phase.

c) Crosstalk

An interchannel level difference of only 18dB is required to give the impression of a signal being either fully left or fully right. Crosstalk between A and B signals is not therefore usually a major problem, since the performance of most audio is far in excess of these requirements. Excessive crosstalk between A and B signals will result in a narrower stereo image, while excessive crosstalk between M and S signals will result in a stereo image increasingly biased towards one side.

CHAPTER 3

COMPONENTS THAT SUPPORT THE AMPLIFIER

This chapter will be discussing on units that support the amplifier.

3.1 TRANSFORMER

Transformer is a device that can be used to increase or decrease the input voltage. Generally transformers are divided into two types, which are;

1. Step Up transformer
2. Step Down transformer

The functions of these transformers are simple. The function of step-up transformer is to increase input voltage and the function of step-down transformer is to decrease the input voltage. Other important function of transformer is to separate the amplifier circuit from the power source, as a voltage supply circuit. In this project, the author uses a step-down transformer to get a small amount of voltage.