"I admit that I had read this report and in my opinion this report had fulfilled all scope and quality for the Bachelor Degree of Electronic Engineering (Industrial Electronic)"

Signature	:	
Supervisor	:	En. Norizan Bin Mohamad.
Date	:	25-3-05

ANTI-ECHO AMPLIFIER USING MICRO-CONTROLLER

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"I admit that all the article is from my own idea except for summarization each of it that I had explain the source"

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Specially dedicated to my beloved parents, family and fellow friends, who had encouraged and supported me in my entire journey of learning...

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ABSTRACT

This is the project of Anti-echo Amplifier Using Micro-controller. This project is purposely to eliminate or reduce the feedback from a microphone. A simple PA (public address) system consists of a microphone, an amplifier, and one or more speakers. Whenever those three components are being used, they have potential for feedback. Feedback is occurred when the sound from the speakers makes it back into the microphone and is re-amplified and sent through the speaker again. This process continues and the result is a loud noise out of the speaker. This phenomenon often happened in a Public Address (PA) System. So, this project attempts to reduce or eliminate the problem by introducing a delay between the microphone inputs to the speaker output by means of a micro-controller. This project consists of hardware and software. The part that consists for the hardware is electronic circuits especially in amplifier design. These electronic circuits are combined with the micro-controller circuit, PIC. The PIC program is the software part. It plays the most important role in this project where the main operations are to process the voice signal, record and playback the signal through Digital-to-Analog Converter. In recording process, the operations that involved are to read the data and store the data into RAM for a period of time. At this stage, the delay is created. As a result, this project is able to eliminate or reduces the loud echo caused by a feedback from microphone. When the microphone is placed near to the speaker, no echo sounds is produced.

ABSTRAK

Projek Anti Echo Amplifier Using Micro-controller ini dibina adalah bertujuan untuk megurangkan atau menghapuskan suapbalik yang terjadi dalam sistem audio. Sistem audio yang melibatkan penggunaan mikrofon, penguat dan speaker kebiasaannya berpotensi untuk terjadinya suapbalik. Supbalik terhasil apabila bunyi yang keluar daripada speaker masuk semula ke dalam mikrofon dan seterusnya dikuatkan oleh penguat dan keluar melalui speaker kembali. Kejadian yang berulang-ulang ini akan mewujudkan suatu bunyi yang bising melalui *speaker*. Itulah yang dinamakan suapbalik. Maka, projek ini dibina adalah bertujuan untuk menghapuskan suapbalik yang terjadi dengan cara mewujudkan lengah antara masukan mikrofon dengan keluaran *speaker*. Lengah ini akan dihasilkan dengan menggunakan PIC micro-controller. Projek ini melibatkan dua bahagian iaitu bahagian perkakasan dan perisian. Bahagian perkakasan melibatkan litar elektronik manakala perisian melibatkan proses membangunkan aturcara (*programming)* untuk fungsi PIC micro-controller. Lengah yang diwujudkan dalam aturcara untuk fungsi PIC micro-controller berupaya untuk mengurangkan atau menghapuskan fenomena suapbalik. Ini dapat dibuktikan apabila mikrofon di letakkan berhampiran dengan *speaker*, tiada bunyi bising terhasil. Sebagai kesimpulannya, projek ini telah berjaya mencapai objektif.

CONTENTS

LIST OF TABLES

LIST OF FIGURES

ABBREVIATIONS AND ACRONYMS

PA	-	Public Address
PIC	-	Peripheral Interface Controller
ADC, A/D	-	Analog-to-Digital Converter
DAC, D/A	-	Digital-to-Analog Converter
RAM	-	Read Access Memory
DC	-	Direct Current
SPI	-	Master Mode for Synchronous Serial Port
TI	-	Texas Instrument
DSP	-	Digital Speech Processing
PWM	-	Pulse Width Modulation
PDA	-	Personal Digital Assistance
LED	-	Light Emitting Diode
FET	-	Field-effect Transistor
I/O	-	Input/Output
IC	-	Integrated Circuit
CPU	-	Central Processing Unit
TAD	-	A/D Conversion Time per Bit
EPROM	-	Electrically Programmable Read Only Memory
PROM	-	Programmable Read Only Memory
ROM	-	Read Only Memory
MCLR	-	Memory Clear
IREF	-	Reference Current

LIST OF APPENDIXES

CHAPTER I

INTRODUCTION

This chapter will give an overall description of Anti-Echo Amplifier Using Micro-controller project. This chapter includes the project overview, project objectives, project scopes, methods of analysis and the thesis summary. Roughly, the flow of work from the beginning until the fulfillment of this project will be described in this chapter.

1.1 PROJECT OVERVIEW

Electronic systems in the context audio communication perform transmission, recording, playback, analysis, synthesis of voice signal and etc. Any system involving these processes are subject to a wide range of influences that may affect the quality of the system. This can include external interferences such as background noise, but it also can extend to echoic effect or nonlinear distortion introduced by analog electro acoustic devices or amplifiers. [1]

The difference of this project compared to other amplifier project is that instead of to amplify the input sound, this project is design to eliminate the noise sound that comes out from the speaker. This project will focus on the noise or echo that is caused by amplifier feedback. Hence, provide the solution to overcome the problem. In the next chapter, the caused of the feedback will be described and the project's methodology is highlighted.

1.2 PROBLEM STATEMENT

In Public Address system, a sequel or high-pitched noise will come out from the speaker. Someone will turn down the volume and the noise will stop. That noise is an indication that the amplifier (at least one stage of amplification) has begun oscillating. Oscillation is caused by a small part of the signal from the amplifier output being sent back to the input of the amplifier. This signal is amplified and again sent back to the input where it is amplified again. This process continues and the result is a loud noise out of the speaker. The process of sending part of the output signal is an amplifier back to the input of the input of amplifier is called feedback.

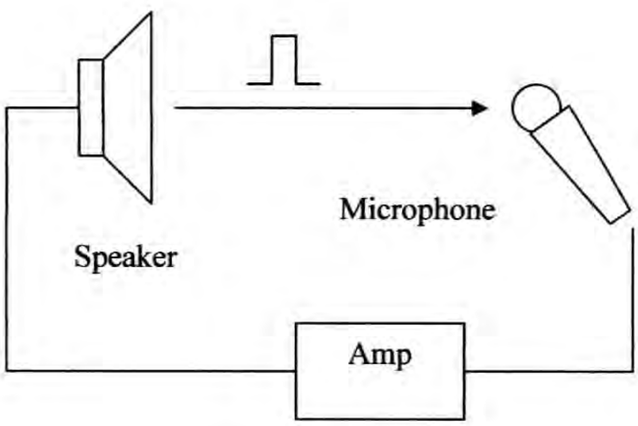

Figure 1.1: Feedback

Until now, there are only a few general rules that are used to avoid feedback such as pointing away the speakers from the microphone, using a unidirectional microphone, using equalizer to damp the frequencies where feedback is occurring, place the microphone close to the person who is speaking or performing,.

Apart from that, a noise gate can be used (automatically shuts off a signal when it gets below a certain threshold) or filter, lower the speaker output, so the mic doesn't pick it up., avoid aiming speakers directly at reflective surfaces such as walls, use direct injection feeds instead of microphones for musical instruments and use headset or in-ear monitors instead of speaker monitors. As can be seen, all of the rules must be set up manually. So, this project gives an alternative way to solve the problems.

1.3 PROJECT OBJECTIVES

This project will focus on the designing of a circuit which can eliminate or minimize the echo sounds. In order to complete the projects task, the most important things to do is to study the cause of the feedback. What is meant by feedback and how it is always occur in audio system will be investigated. After that, the objective is to find the best solution to reduce or eliminate the problems.

This project is split into two major components that are the circuit and the PIC program. From the PIC program, the next objectives are to study the micro-controller circuit, the function of the Analog-to Digital Converter (ADC) in the micro-controller circuit and how it works. The main objective from the study of micro-controller is to create a delay program which is actually the solution to minimize the echo. In order to do that, the PIC should be program so that the input signal from amplifier can be converted into digital signal, and being stored into the micro-controller RAM for several second thus provides the delay.

Other objectives of this project are to study and understand the function of amplifier, other digital circuitry and the micro-controller circuit. After that, the next objectives are to design the circuit and to analyze the system function of the circuit. Finally, a complete circuit of Anti-echo Amplifier Using Micro-controller will be build and the amplifier feedback can be reduced or eliminated.

1.4 PROJECT SCOPES

This project is convergent to audio system users which consist the using of a microphone and speaker. In other words, this project is built especially for PA system users. It is design in order to eliminate the feedback from microphone. The elimination will be done by introducing a delay which will be created using PIC micro-controller. Eventually, this project will provide a better performance of audio system, which is free from 'echoes'.

1.5 METHODS OF ANALYSIS

Since there is no similar project has been done previously, the only method to perform the task is to do a lot of revision from any source. The circuit is the combination of microphone pre-amplifier circuit, Micro-controller circuit, Digital to Analog Converter circuit and finally the amplifier circuit to drive the speaker. To ensure that the connection of the circuit is correct, the typical application of the component is referred from the components datasheet. Reference books, journals and other kind of articles are essential as a reference and as a comparison in developing this project. Also the guidance from supervisor and lecturers are important and from that, the ideas to design the circuit are created.

While for the micro-controller programming part, the PIC online tutorials, PIC project books, and samples of other project are being referred. From that, a revision has been made and the program is developed step-by-step. The graphical results from the analysis are performed in signal figure from both simulation and practical part.

1.6 THESIS SUMMARY

This thesis contains of five chapters which will describe this project in details. The first chapter is the introduction part where in this chapter, the overall descriptions of the project are highlighted. The objectives of this project, the scopes, and the methodology are described.

Second chapter of this thesis will be discussing about the research and analysis of this project. Each of the facts and the information from the analysis will be explained.

The third chapter will describe the method used in implement the project tasks. The techniques and methodology of this project is split into two major parts that is the hardware and software. In this chapter, each part will be described in depth.

The fourth chapter is the result and analysis of the project. In this chapter, all of the final results and the analysis that had been done will be stated clearly. The analysis is done by using certain equipment such as the oscilloscope, digital analyzer and etc.

The final chapter is the conclusion and the suggestion. A conclusion about the achievement of the projects objectives and the knowledge gained while doing this project is being stated in this chapter. The suggestion is made to improve the project operation for the next reference.

CHAPTER II

LITERATURE REVIEW AND PROJECT BACKGROUND

This chapter will described about the background of this project and literature review which includes the previous related project and the development of the technologies. For the project background, the history of an amplifier and how the feedback is occurred in most audio system are described. While for the literature review, the speech processing technique using PIC and the current noise cancellation technologies are explained.

2.1 PROJECT BACKGROUND

2.1.1 History of Amplifier

In the early days, power amplifiers used devices called *vacuum tubes*. Tubes are seldom used in amplifiers intended for DJ use (however tube amplifiers have a loyal following with musicians and hi-fi enthusiasts).

Modern amplifiers almost always use transistors instead of tubes; in the late 60's and early 70's, the term "solid state" was used. The signal path in a tube amplifier undergoes similar processing as the signal in a transistor amp, however the devices and voltages are quite different. Tubes are generally "high voltage low current" devices, where transistors are the opposite ("low voltage high current"). Tube amplifiers are generally not very efficient and tend to generate a lot of heat.

One of the biggest differences between a tube amplifier and a transistor amplifier is that an *audio output transformer* is almost always required in a tube amplifier. This is because the output impedance of a tube circuit is far too high to properly interface directly to a loudspeaker. High quality audio output transformers are difficult to design, and tend to be large, heavy, and expensive. Transistor amplifiers have numerous practical advantages as compared with tube amplifiers: they tend to be more efficient, smaller, and more rugged, no audio output transformer is required, and transistors do not require periodic.

A well designed tube amplifier can have excellent sound. Some people claim that tube amplifiers have their own particular "sound". This "sound" is a result of the way tubes behave when approaching their output limits (clipping). The onset of output overload in a tube amplifier tends to be much more gradual than that of a transistor amplifier.

A few big advantages that tube amplifiers have were necessarily given up when amplifiers went to transistors. First, tubes can withstand electrical abuse that would leave even the most robust transistor completely blown. Also, tube amplifiers use an output transformer to interface to the speaker; such a device provides an excellent buffer (protection to the speaker) in the case of internal malfunction. Modern amplifiers (with no output transformer) occasionally fail in a way that connects the full DC supply voltage to the speaker. If the amplifier does not have adequate protection circuitry built in, the result is often a melted woofer voice coil.

The Class of an amplifier refers to the design of the circuitry within the amp. There are many classes used for audio amps. The following is brief description of some of the more common amplifier classes:

Class A: Class A amplifiers have very low distortion (lowest distortion occurs when the volume is low) however they are very inefficient and are rarely used for high power designs. The distortion is low because the transistors in the amp are biased such that they are half "on" when the amp is idling. As a result, a lot of power is dissipated even when the amp has no music playing! Class A amps are often used for "signal" level circuits (where power is small) because they maintain low distortion. Distortion for class A amps increases as the signal approaches clipping, as the signal is reaching the limits of voltage swing for the circuit. Also, some class A amps have speakers connected via capacitive coupling.

Class B: Class B amplifiers are used in low cost, low quality designs. Class B amplifiers are a lot more efficient than class A amps, however they suffer from bad distortion when the signal level is low (the distortion is called "crossover distortion"). Class B is used most often where economy of design is needed. Before the advent of IC amplifiers, class B amplifiers were common in clock radio circuits, pocket transistor radios, or other applications where quality of sound is not that critical.

Class AB: Class AB is probably the most common amplifier class for home stereo and similar amplifiers. Class AB amps combine the good points of class A and B amps. They have the good efficiency of class B amps and distortion that is a lot closer to a class A amp. With such amplifiers, distortion is worst when the signal is low, and lowest when the signal is just reaching the point of clipping. Class AB amps (like class B) use pairs of transistors, both of them being biased slightly ON so that the crossover distortion (associated with Class B amps) is largely eliminated.