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award of the degree of Bachelor of Electronics Engineering (Computer Engineering)."

Signature	: ....
Name of Supervisor	: ....
Date	: ....

DESIGN OF A PORTABLE MP3 PLAYER

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This Report Is Submitted In Partial Fulfillment Of Requirements For The Bachelor Degree of Electronic Engineering (Computer Engineering)

Faculty of Electronics & Computer Engineering
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APRIL 2006

I declare that this thesis entitled "*Design of a Portable MP3 Player*" is the result of my own research except as cited in the references. The thesis has not been accepted for any degree and is not concurrently submitted in candidature of any other degree.

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Name : ....

Date : ....

ACKNOWLEDGEMENT

First of all, I would like to take this opportunity to express my deepest appreciation and gratitude to my project supervisor, Cik Hazura Haroon, who has contributed countless hours, interest, patience and effort in guiding me throughout the whole duration of the project. I am grateful for her valuable ideas and time that she has spend on me in suggesting the criteria of the design. Besides, thank you for her moral support for me to complete this project.

A part from that, great appreciation is also expressed to all the technicians and staff of Department of Computer Engineering for their technical support and valuable suggestion during this project. Appreciation is also addressed to Faculty of Electronics and Computer Engineering for providing the facilities and equipment needed for completing this project.

Finally, I would like to express my gratitude to all the helpful lecturers and course mates for their support, suggestion and understanding throughout the whole process of my project development.

ABSTRACT

This project is to design and physically create a portable MP3 player in the form of hardware and software. The portable MP3 player is an electronic device that can decode and play MP3 files. The MP3 format is a compression system for music. The MP3 player utilizes a memory device (Compact Flash) for storing MP3 files. The decoder (VS1001k) takes an MP3 file and converts it to an analog signal with the use of a digital-to-analog converter. After the file is converted, the signal is amplified through an amplifier so that the sound can be heard with the use of speakers or headphones. There are some user-interface buttons to allow the user to switch songs, pause, next, previous and adjust the volume. The microcontroller (PIC16F877) has the responsibility of coordinating all activities within the MP3 player. All devices send and receive information from the microcontroller. A high level programming language C is used to program in the microcontroller.

ABSTRAK

Project ini merekacipta sebuah pemain ‘MP3’ mudah alih dengan menggunakan komponen-komponen elektronik dan program. Pemain ‘MP3’ mudah alih ini adalah sebuah alat elektronik yang boleh mentafsir kod and memainkan fail ‘MP3’. Format ‘MP3’ ialah satu sistem pemampatan audio. Pemain ‘MP3’ ini menggunakan sebuah alat ingatan (Compact Flash) untuk menyimpan fail ‘MP3’. Pentafsir kod (VS1001k) pula akan mengambil fail ‘MP3’ dan menukarkan fail tersebut ke isyarat analog dengan menggunakan sebuah penukar digital ke analog. Selepas pertukaran fail, isyarat tersebut akan dikuatkan dengan menggunakan sebuah penguat bunyi supaya lagu yang dimainkan dapat didengar melalui alat pembesar suara atau alat pendengar. Pemain ‘MP3’ ini terdapat butang-butang yang membolehkan pengguna mengubah kekuatan suara, memainkan lagu seterusnya, lagu sebelumnya dan ‘pause’. Pengawal mikro (PIC16F877) pula bertanggungjawab untuk menyelaraskan segala aktiviti-aktiviti yang dijalankan dalam pemain ‘MP3’. Semua komponen akan menghantar dan menerima isyarat melalui pengawal mikro. Satu bahasa program C yang bertaraf tinggi digunakan untuk memprogramkan pengawal mikro.

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LIST OF TERMINOLOGIES

MP3	-	MPEG Audio Layer-3
MPEG	-	Moving Picture Experts Group
CD	-	Compact Disc
PIC	-	Peripheral Interface Controller
CF	-	Compact Flash
DVD	-	Digital Video Disk
HDTV	-	High Definition Television
DSS	-	Digital Spread Spectrum
ISO	-	International Organization for Standardization
PC	-	Personal Computer
FM	-	Frequency Modulation
LGPL	-	Lesser General Public License
ACM	-	Audio Compression Manager
VBR	-	Variable Bit Rate
CPU	-	Central Processing Unit
LCD	-	Liquid Crystal Display
USB	-	Universal Serial Bus
DSP	-	Digital Signal Processor
ADC	-	Analog to Digital Converter
DAC	-	Digital to Analog Converter
UV	-	Ultraviolet
RAM	-	Random Access Memory
ROM	-	Read Only Memory

CHAPTER 1

INTRODUCTION

MPEG Layer III (MP3) music compression format has become an extremely popular choice for digital audio compression. Its high compression ratio, and near CD quality sound make it a logical choice for storing and distributing music - especially over the internet, where space and bandwidth are important considerations. As a result of the MP3 popularity, a variety of portable MP3 players have entered into the market in an attempt to capitalize on the demand for portable, high quality music. Accordingly, a portable MP3 player based with microcontroller is designed and implemented similar to products currently available.

1.1 Objectives

a. To study the operations and functions of components in the portable MP3 player and the technology of MP3 player.
b. To design a portable MP3 player based PIC microcontroller.

1.2 Scope of Works

1.2.1 Hardware

Main Components/Equipments :

a. Power supply (Regulator)

b. Microcontroller (PIC16F877)

c. Decoder (VS1001K)

d. CF memory card (64MB)

e. Ear phone

f. Pushbuttons

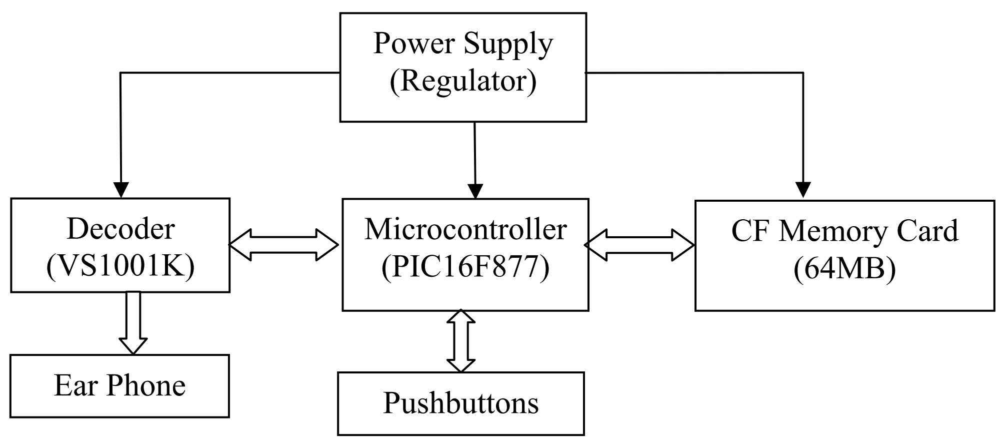

Figure 1.1: Block Diagram of a Portable MP3 Player

a) Power is supplied to the decoder, PIC microcontroller, compact flash (CF) memory card and all the pushbuttons such as play, pause, stop, next and previous.

b) PIC microcontroller is connected to decoder, compact flash (CF) memory card and all the pushbuttons to control them.

c) The decoder is connected to a set of headphones or speakers to play out music.

1.2.2 Software

a) A high level programming language, C is programmed into the PIC microcontroller.

b) Program C is written and used to control compact flash (CF) memory card, decoder and all the buttons.

CHAPTER 2

LITERATURE REVIEW

2.1 MP3

MPEG is the acronym for Moving Picture Experts Group. This group has developed compression systems used for video data. For example, DVD movies, HDTV broadcasts and DSS satellite systems use MPEG compression to fit video and movie data into smaller spaces. The MPEG compression system includes a subsystem to compress sound, called MPEG Audio Layer-3. We know it by its abbreviation, MP3 [1].

MP3 is a popular digital audio encoding and lossy compression format. It was designed to greatly reduce the amount of data required to represent audio, yet still sound like a faithful reproduction of the original uncompressed audio to most listeners. In popular usage, MP3 also refers to files of sound or music recordings stored in the MP3 format on computers [2].

2.1.1 History of MP3

MP3 was invented and standardized in 1991 by a team of engineers directed by the Fraunhofer Society in Germany.

The Moving Picture Experts Group (MPEG), the ISO (International Organization of Standardization) working group charged with developing compressed digital audio and video standards, concludes a first compression standard called MPEG-1 for use in video CDs (CD-I) . In its audio section, a generic family of three codec formats (Layer-1, -2, -3) is specified. Layer 3 is a more efficient codec and leads to its widespread adoption as a way to store music on the relatively small hard disk drives of the era's PCs and to transfer music files over the Internet through pokey 28.8kbps PC modems.

In 1995, MP3 gets its name. In an internal poll, Fraunhofer researchers unanimously vote for .mp3 as the file-name extension for MPEG Layer 3. MPEG Layer-3 is also selected as the audio format for the WorldSpace satellite digital audio broadcasting system.

In 1998**,** the era of MP3 portability began with the introduction of Diamond Multimedia's Rio in the U.S. and Saehan Information Systems's MPMAN in Korea. They are the first headphone stereos that used solid-state flash memory to store and play compressed MP3 music files, either downloaded from the Internet or "ripped" from a music CD. The ensuing popularity of MP3 portables led dozens of companies to offer compressed-music portables, and it led to the development of additional audio codecs for use in PCs and in portable devices.

In 2000**,** the U.S. suppliers launch the first headphone stereos equipped with hard drives and the first headphone CD players that play MP3-encoded 5-inch CDs [2].

2.1.2 The MP3 Format

The MP3 format is a compression system for music. This format helps to reduce the number of bytes in a song, without hurting the quality of the song's sound. The goal of the MP3 format is to compress a CD-quality song by a factor of 10 to 14, without losing the CD sound quality. A 32 megabyte (MB) song on a CD compresses down to about 3 MB on MP3. A song can be downloaded in minutes rather than hours, and 10 to 20 songs can be stored on an MP3 player using a relatively small amount of memory [1].

2.1.3 Quality of MP3 Audio

MP3 is able to provide a number of different options for its “bit rate”—that is, the number of bits of encoded data that are used to represent each second of audio. Typically rates chosen are between 128 and 320 Kbit per second. By contrast, uncompressed audio as stored on a compact disc has a bit rate of 1411.2 Kbit/s.

MP3 files encoded with a lower bit rate will generally play back at a lower quality. With too low a bit rate, “compression artifacts” (i.e., sounds that were not present in the original recording) may appear in the reproduction. A good demonstration of compression artifacts is provided by the sound of applause: it is hard to compress because it is random, therefore the failings of the encoder are more obvious, and are audible as ringing [2].

As well as the bit rate of the encoded file, the quality of MP3 files depends on the quality of the encoder and the difficulty of the signal being encoded. For average signals with good encoders, many listeners accept the MP3 bit rate of 128 Kbit/s and a sample rate of 44.1 KHz as near enough to compact disc quality for them, providing a compression ratio of approximately 11:1. MP3s compressed at this ratio will generate audio that is much truer to the original CD recording than will FM radio and cassette tapes. This is primarily due to fact that FM radio and cassette tapes are limited to a sample rate of approximately 22 KHz. MP3's could be compressed to 64 Kbit/s and 22 KHz to rival the quality of two inferior audio standards. However, listening tests show that with a bit of practice many listeners can reliably distinguish 128 Kbit/s MP3s from CD originals; in many cases reaching the point where they consider the MP3 audio to be of unacceptably low quality. Yet other listeners, and the same listeners in other environments (such as in a noisy moving vehicle or at a party) will consider the quality acceptable. Obviously, imperfections in an MP3 encode will be much less apparent on low-end computer speakers than on a good stereo system connected to a computer or -- especially -- using high-quality headphones.

Fraunhofer Gesellschaft (FhG) publish on their official webpage the following compression ratios and data rates for MPEG-1 Layer 1, 2 and 3, intended for comparison:

- Layer 1: 384 Kbit/s, compression 4:1
- Layer 2: 192...256 Kbit/s, compression 8:1...6:1
- Layer 3: 112...128 Kbit/s, compression 12:1...10:1

The differences between the layers are caused by the different psychoacoustic models used by them; the Layer 1 algorithm is typically substantially simpler, therefore a higher bit rate is needed for transparent encoding. However, as different encoders use different models, it is difficult to draw absolute comparisons of this kind [2].

Many people consider these quoted rates as being heavily skewed in favour of Layer 2 and Layer 3 recordings. They would contend that more realistic rates would be as follows:

- Layer 1: excellent at 384 Kbit/s

- Layer 2: excellent at 256...384 Kbit/s, very good at 224...256 Kbit/s, good at 192...224 Kbit/s
- Layer 3: excellent at 224...320 Kbit/s, very good at 192...224 Kbit/s, good at 128...192 Kbit/s

When comparing compression schemes, it is important to use encoders that are of equivalent quality. Tests may be biased against older formats in favour of new ones by using older encoders based on out-of-date technologies, or even buggy encoders for the old format. Due to the fact that their lossy encoding loses information, MP3 algorithms work hard to ensure that the parts lost cannot be detected by human listeners by modeling the general characteristics of human hearing (e.g., due to noise masking). Different encoders may achieve this with varying degrees of success [2].

A few possible encoders:

- LAME first created by Mike Cheng in early 1998. It is (in contrast to others) a fully LGPL'd MP3 encoder, with excellent speed and quality, rivaling even MP3's technological successors.
- Fraunhofer Gesellschaft: Some encoders are good, some have bugs.

Many early encoders that are no longer widely used:

- ISO dist10 reference code
- Xing
- BladeEnc
- ACM Producer Pro.

Good encoders produce acceptable quality at 128 to 160 Kibit/s and near-transparency at 160 to 192 Kbit/s, while low quality encoders may never reach transparency, not even at 320 Kbit/s. It is therefore misleading to speak of 128 Kbit/s or 192 Kbit/s quality, except in the context of a particular encoder or of the best available encoders. A 128 Kbit/s MP3 produced by a good encoder might sound better than a 192 Kbit/s MP3 file produced by a bad encoder [2].

2.1.4 Bit Rate

The bit rate is variable for MP3 files. The general rule is that more information is included from the original sound file when a higher bit rate is used, and thus the higher the quality during play back. In the early days of MP3 encoding, a fixed bit rate was used for the entire file.

Bit rates available in MPEG-1 Layer 3 are 32, 40, 48, 56, 64, 80, 96, 112, 128, 160, 192, 224, 256 and 320 Kbit/s, and the available sample frequencies are 32, 44.1 and 48 KHz. 44.1 KHz is almost always used (coincides with the sampling rate of compact discs), and 128 Kbit/s has become the de facto “good enough” standard.

Variable bit rates (VBR) are also possible. Audio in MP3 files are divided into frames (which have their own bit rate) so it is possible to change the bit rate dynamically as the file is encoded (although not originally implemented, VBR is in extensive use today). This technique makes it possible to use more bits for parts of the sound with higher dynamics (more sound movement) and fewer bits for parts with lower dynamics, further increasing quality and decreasing storage space. This method compares to a sound activated tape recorder that reduces tape consumption by not recording silence. Some encoders utilize this technique to a great extent [2].

Bitrates aren't quite the final arbiter of quality. The resolution of audio signal in general is in large part determined by the number of source samples per second stored in a given format. While bitrates are a measure of the *amount of data stored* for every second of audio, samplerates measure the *frequency with which the signal is stored*, and are measured in kiloHertz, or thousands of samples per second. The standard samplerate of CD audio is 44.1kHz, so this is the default samplerate used by most encoders, and found in most downloadable MP3 files. Audio professionals often work with 48kHz audio (and, more recently, 96kHz). Digital audio storage of lectures and plain speech is sometimes recorded as low as 8kHz. Streamed MP3 audio is often sent out at half, or even a quarter of the CD rate in order to compensate for slow Internet connection speeds [3].

While MP3 users cannot control the degree of lossiness specifically, as they might do with a JPEG image, they can control the number of bits per second to be devoted to data storage, which has a similar net result.

In the process of coding, the "irrelevant" portions of the signal are mapped against two factors: a mathematical model of human psychoacoustics (i.e., the masking requirements), and the bitrate, which is established at the time of encoding. The bitrate simply refers to the number of bits per second that should be devoted to storing the final product-the higher the bitrate, the greater the audio resolution of the final product, as shown in Figure 2.1. An easy way to visualize the effect of bitrate on audio quality is to think of an old, turn-of-the-century film. Old movies appear herky-jerky to us because fewer frames per second are being displayed, which means less data is distributed over a given time frame [4].

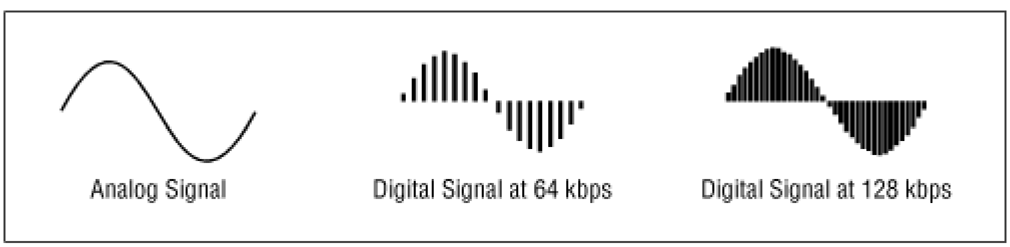

Figure 2.1: More bits per second means more audio resolution, pure and simple

For example, the current de facto standard is to encode MP3 at 128 kbps, or 128,000 bits per second. The codec takes the bitrate into consideration as it writes each frame to the bitstream. If the bitrate is low, the irrelevancy and redundancy criteria will be measured harshly, and more subtlety will be stripped out, resulting in a lower-quality product. If the bitrate is high, the codec will be applied with leniency, and the end result will sound better [4].

2.1.5 Encoding of MP3 Audio

The MPEG-1 standard does not include a precise specification for an MP3 encoder. The decoding algorithm and file format, as a contrast, are well defined. Implementers of the standard were supposed to devise their own algorithms suitable for removing parts of the information in the raw. During encoding 576 time domain samples are taken and is transformed to 576 frequency domain samples. If there is a transient 192 samples are taken instead of 576. This is done to limit the temporal spread of quantization noise accompanying the transient.

This is the domain of psychoacoustics: the study of human acoustic perception (in both the ear and in the brain).

As a result, there are many different MP3 encoders available, each producing files of differing quality. Comparisons are widely available, so it is easy for a prospective user of an encoder to research the best choice. It must be kept in mind that an encoder that is proficient at encoding at higher bitrates is not necessarily as good at other, lower bitrates [2].

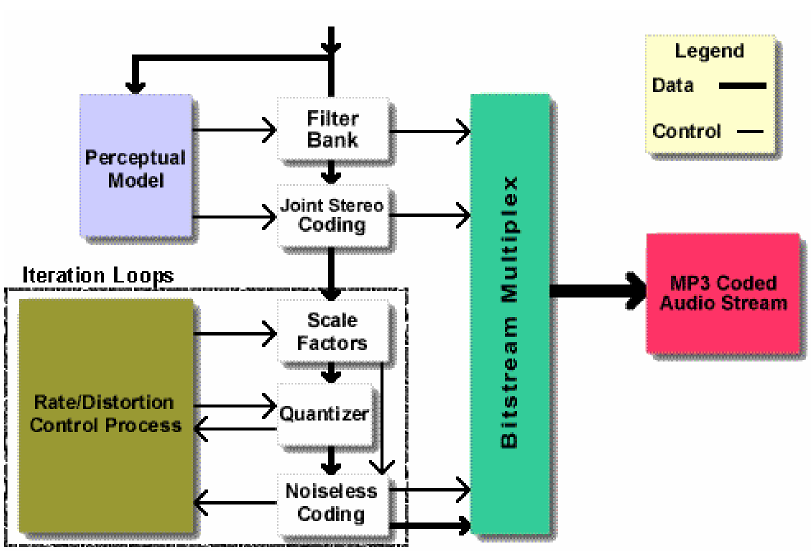

Figure 2.2: MP3 Encoder Flow Chart