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ARM 7 BASED MP3 PLAYER

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ABSTRACT
MP3 is a patented encoding format for digital audio which uses a form of lossy data compression. It is a common audio format for consumer audio streaming or storage, as well as a de facto standard of digital audio compression for the transfer and playback of music on most digital audio players. Since the MPEG-1 Layer III encoding technology is nowadays widely used it might be interesting to gain knowledge of how this powerful compression/decompression scheme actually functions. The MPEG-1 Layer III is capable of reducing the bit rate with a factor of 12 without almost any audible degradation. Arm7 lpc2148 is arm7tdmi-s core board microcontroller that uses 16/32-bit 64 pin (lqfp) microcontroller no.lpc2148 from Philips (nxp).The hardware system of lpc2148 includes the necessary devices within only one mcu has such as usb, adc, dac, timer/counter, pwm, capture, i2c, spi, uart, and etc.
Keywords – Arm 7, LPC2148, Mp3 Player, SD Card and SPI Protocol.

I. INTRODUCTION

Along with the enhancement of the people's requests of the portable music player, MP3 players in smaller size and better quality of tone win the general musical public's favor. At the beginning, MP3 files c an only be played by computers, with the development of the internet the MP3 player production is romoted. And along with the people's further understanding and higher demand of MP3 products, the MP3 products had a series of changes to be more exquisite, smaller and man machine; and because of the individual needs which is also getting stronger and stronger, function and product integration is also presented [1].

MP3 stands for MPEG Audio Layer III and it is a standard for audio compression that makes any music file smaller with little or no loss of sound quality. MP3 is part of MPEG, an acronym for Motion Pictures Expert Group, a family of standards for displaying video and audio using lossy compression. Standards set by the Industry Standards Organization or ISO, beginning in 1992 with the MPEG-1 standard. MPEG-1 is a video compression standard with low bandwidth. MPEGI/2 Audio layer-3 (MP3) is the most popular format for playback of high quality compressed audio for portable devices such as audio players and mobile phones. Typically these devices are based on either DSP or RlSC processors [2].

II. OVERVIEW OF ARM7 TECHNOLOGY

ARM was established as a joint venture between Arcon, Apple and VLSI. ARM is the industry'sleading provider of 16/32-bit embedded RISC microprocessor solutions. The company licenses its high performance, low cost, power efficient RICS processors, peripherals, and system-chipdesigns to leading international electronics companies. ARM designs microprocessor technology that lies at the heart of advanced digital products, from mobile phones and digital cameras to games consoles and automotive systems, and is leading intellectual property (IP) provider of high-performance, low-cost, power-efficient RISC processors, peripherals, and system-on-chip (SoC) designs through involvement with organizations such as the Virtual Socket Interface Alliance (VSIA) and Virtual Component Exchange (VCX).

The ARM7 is part of the Advanced RISC Machines (ARM) family of general purpose 32-bit microprocessors, which offer very low power consumption and price for high performance devices. The architecture is based on Reduced Instruction Set Computer (RISC) principles, and the instruction set and related decode mechanism are much simpler in comparison with micro programmed Complex Instruction Set Computers. This results in a high instruction throughput and impressive real-time interrupt response from a small and cost-effective chip.

The instruction set comprises eleven basic instruction types:
• Two of these make use of the on-chip arithmetic logic unit, barrel shifter and multiplier to perform high-speed operations on the data in a bank of 31 registers, each 32 bits wide;
• Three classes of instruction control data transfer between memory and the registers, one optimized for flexibility of addressing, another for rapid context switching and the third for swapping data;
• Three instructions control the flow and privilege level of execution; and
• Three types are dedicated to the control of external coprocessors which allow the functionality of the instruction set to be extended off-chip in an open and uniform way.

III. PROPOSED SYSTEM DESIGN

Proposed system is implemented using LPC2148 processor (of ARM7TDMI family) as shown in Fig 1. The processor is equipped with many peripherals, such as USB 2.0 controller, two multi-channel A/D converters and one D/A converter with 10 bit resolution, two 32 bit timers/counters, a PWM unit, two UART, two SPI and two I2C serial interfaces. From these rich possibilities, we have used the A/D converters, both timers/counters, the PWM unit, and both UART interfaces, a 2x16 characters LCD display, a trimmer, buzzer and four pushbuttons [3].

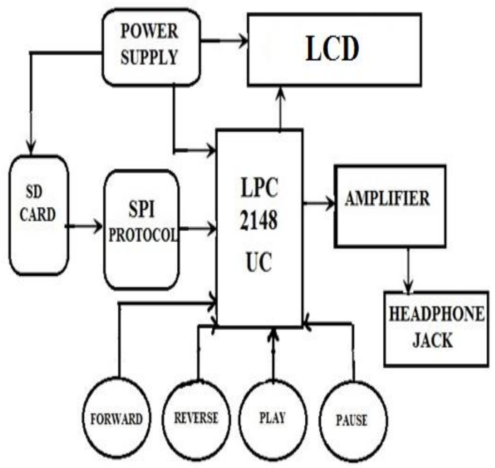

Figure 1 Block Diagram of MP3 Player

3.1 SD CARD USING SPI PROTOCOL

Secure Digital (SD) is a non-volatile memory card format for use in portable devices, such as mobile phones, digital cameras, GPS navigation devices, and tablet computers. The Secure Digital standard was introduced in 1999 as an evolutionary improvement over Multimedia Cards (MMC). The Secure Digital standard is maintained by the SD Card Association (SDA). SD technologies have been implemented in more than 400 brands across dozens of product categories and more than 8,000 models.

SD card consists of a 9-pin interface as shown in Fig 2, a card controller, a memory interface and a memory core. The 9-pin interface allows the exchange of data between a connected system and the card controller. The controller can read/write data from/to the memory core using the memory core interface. Communication with an SD card can be done in one of two modes: the SD mode or the SPI mode. By default, the SD card operates in the SD mode [3].

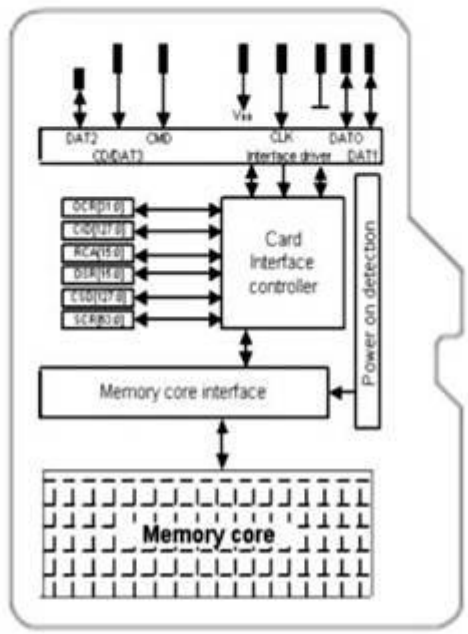

Figure 2 SD Card Architecture

Serial Peripheral Interface (SPI) communication was used to connect devices such as printers, cameras, scanners, etc. to a desktop computer; but it has largely been replaced by USB. SPI is still utilized as a communication means for some applications using displays, memory cards, sensors, etc. SPI runs using a master/slave set-up and can run in full duplex mode (i.e., signals can be transmitted between the master and the slave simultaneously). When multiple slaves are present, SPI requires no addressing to differentiate between these slaves. There is no standard communication protocol for SPI. SPI is used to control peripheral devices and has some advantages over I2C. Because of its simplicity and generality, it is being incorporated in various peripheral ICs. The number of signals of SPI, three or four wires, is larger than I2C's two wires, but the transfer rate can rise up to

20 Mbps or higher depends on device's ability (5 - 50 times faster than I2C). Therefore, often it is used in applications (ADC, DAC or communication between ICs) that require high data transfer rates.

3.2 LPC2148 MICROCONTROLLER

The LPC2148 microcontrollers are based on a 32/16 bit ARM7TDMI-S CPU with real-time emulation and embedded trace support, that combines the microcontroller with embedded high speed flash memory ranging from 32 kB to 512 kB. A 128-bit wide memory interface and unique accelerator architecture enable 32-bit code execution at the maximum clock rate. For critical code size applications, the alternative 16-bit Thumb mode reduces code by more than 30 % with minimal performance penalty.

Due to their tiny size and low power consumption, LPC2148 are ideal for applications where miniaturization is a key requirement, such as access control and point-of-sale. A blend of serial communications interfaces ranging from a USB 2.0 Full Speed device, multiple UARTs, SPI, SSP to I2Cs, and on-chip SRAM of 8 kB up to 40 kB, make these devices very well suited for communication gateways and protocol converters, soft modems, voice recognition and low end imaging, providing both large buffer size and high processing power [4, 5].

3.3 LM386 LOW VOLTAGE AUDIO POWER AMPLIFIER

The LM386 is a power amplifier designed for use in low voltage consumer applications. The gain is internally set to 20 to keep external part count low, but the addition of an external resistor and capacitor between pins 1 and 8 will increase the gain to any value from 20 to 200. The inputs are ground referenced while the output automatically biases to one-half the supply voltage as shown in Fig 3. The quiescent power drain is only 24 mill watts when operating from a 6 volt supply, making the LM386 ideal for battery operation [6].

Following are the features of the amplifier:

- Battery operation
- Minimum external parts
- Wide supply voltage range: 4V–12V or 5V–18V
- Low quiescent current drain: 4mA
- Voltage gains from 20 to 200
- Ground referenced input
- Self-centering output quiescent voltage
- Low distortion: 0.2% (AV = 20, VS = 6V, RL = 8W, PO = 125mW, f = 1 kHz)
- Available in 8 pin MSOP package

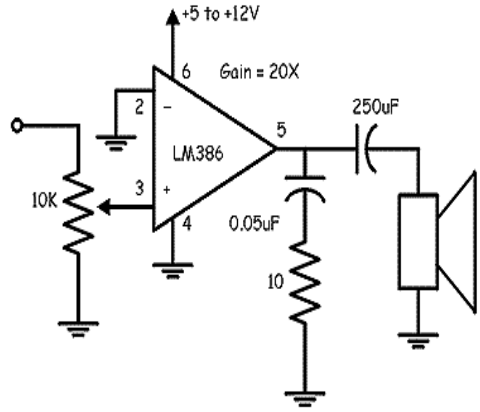

Figure 3 LM386 Audio Power Amplifier

IV. SOFTWARE DESIGN

The software design of MP3 player consists of its working principle with the encoding and decoding techniques used.

4.1 MP3 PLAYER WORKING PRINCIPLE

The basic principle of MP3 player is the MP3 or WMA digital music files or other formats in flash memory, via USB interface, are transmitted to decoder chip integrated in the main chip to decode turned into digital signals, which then are transformed into analog audio signals by the digital-to-analog conversions, and amplifies by the audio amplifier, and finally get output through the headphone [1].

The system generally works as follows: When a user wants to play MP3 music, if there are no songs in Flash memory in the system, they can via USB interface download music files from PC to a memory chip; if the system already has MP3 songs, the user can select MP3 file via the file name or the order of selection, then the main chip read the selected audio file from the Flash memory to the MP3 decoder chip, and decoded the audio file into a digital signal, the signal is sent according to external D/ A converter format which set by the users, via an external audio interface to A/D converter, an external A/D converter converts the digital signal into analog signals that human ear can accept, through the power amplifier and sent to headphones. During the playing course, the volume can be increased or decreased in the light of the choice and control. The process flow is shown in the Fig 4 [7].

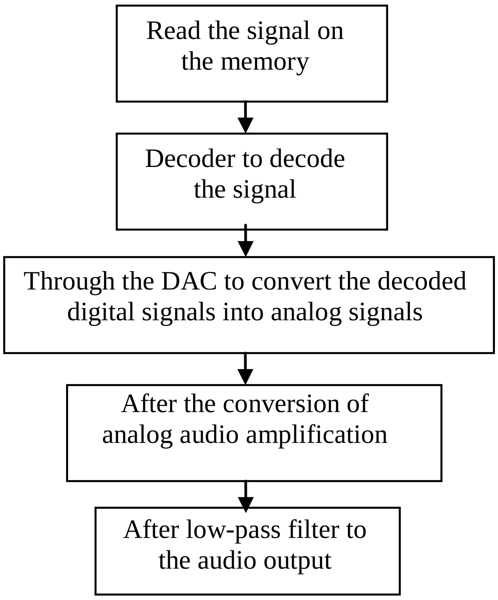

Figure 4 MP3 Flow Algorithm

4.2 MP3 ENCODING AND DECODING PROCESSES

The MP3 bit stream is a concatenation of a sequence of data “frames”. Each frame corresponds to two “granules” of audio where each granule is defined as precisely 576 consecutive audio samples. A granule, also called a “long block” may be sometimes divided into 3 “short blocks” of 192 samples each. The steps involved in decoding an MP3 frame are as follows:

1. Synchronization to the start of the frame and decoding of header information.
2. Decoding the side information including scale factor selection information, block splitting information, and table selection information.
3. Decoding the “main data” for both granules, including the Huffman bits for the transform coefficients, and scale factors. The main data may overflow into adjoining frames of the bit stream. Multiple frames of data may therefore need to be buffered to decode all the samples in both granules.

After the frame bits have been thus parsed the next step is to reconstruct the audio for each granule from the decoded bits using the following steps:

1. De-quantizing the transform coefficients from the main and side information. A non-linear transformation is applied to the decoded coefficients.
2. In the case of short blocks, the dequantized coefficients may be re-ordered and divided into three sets of coefficients, one per block.
3. In the case of certain stereo signals where the right and left channels may be jointly encoded, the transform coefficients are recast into left (L) and right (R) channel coefficients via a channel transformation.
4. An “alias reduction” step is applied for long blocks.
5. The Inverse Modified Discrete Cosine Transform (IMDCT) module is applied for coefficients corresponding to each of the 32 sub-bands in each channel.
6. Overlap-add mechanism is used on the IMDCT outputs generated in consecutive frames. Specifically the first half of the IMDCT outputs are overlapped and added with the second half of the IMDCT outputs generated in the corresponding sub-band in the previous granule.
7. The final step is the inverse poly-phase filter bank which combines the 32 sub band signals back into a full-bandwidth, time domain signal [1, 3].

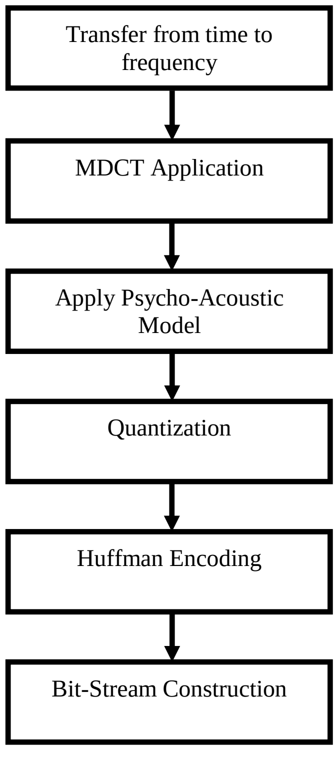

Figure 5 MP3 Encoding Process

V. CONCLUSION

MP3 technology has become very popular since its first edition release in 1993. The benefits of this technology are: The low data size (file size) is the biggest advantage. The compression ratio is not fixed which means that the user is free to compress the files to the desired size, by having a tradeoff between file size and audio quality.

Less compression will give better audio quality but larger file size. Due to the digital format of MP3 files, even if several copies of the same file are created, the audio quality will remain the same. This technique is known as serial duplication. The biggest disadvantage is the low audio quality. MP3 uses a "lossy" algorithm that deletes the "lesser audible" music content in order to reduce the file size, thus compromising on the music quality. Thus in future one can make a MP3 player using lossless algorithm which would give maximum efficiency and reduce the amount of data loss.

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REFERENCES

[1] Vijaykumar Gurkhe: *“Optimization of an mp3 decoder on the arm processor”,* Texas Instruments, Wind Tunnel Road, Bangalore, India-I7, TENCON 2003.

[2] Ying-Wen Bai and Chun-Lee Chiang: “*Design and Implementation of the Integration Applications for a Portable MP3 Player with a Bluetooth Hand-free/set”*, Department of Electronic Engineering, *Vol. 51, No. 3, AUGUST 2005.*

[3] Evan P. Ruzanski: “*Effects of MP3 encoding on the sounds of music”, MARCH/APRIL 2006.*

[4] Ying-Wen Bai and Fu-En Tsai: “*Design and Implementation of a Table-based GUI for MP3 Players”*, Department of Electronic Engineering*, Vol. 53, No. 2, MAY 2007*.

[5] Teodor Neagoe, Ernest Karjala, Logica Banica: “*Why ARM Processors are the Best Choice for Embedded Low-Power Applications”*, 2010 IEEE 16th International Symposium for Design and Technology in Electronic Packaging (SIITME).

[6] Jing Zhang, Xue-Mei Li, Wei Xu, He-Sheng Lv, Cheng-yong Li (School of Electronic Information and Automation, Chongqing University of Technology, Chongqing 400050, China), “*Design of MP3 player based on C51 MCU”, 2011.*

[7] Mengzhe Huang∗, Shengpei Wang∗, Hui Yang∗, Shuyuan Xiao∗ and Bei Wang: “*A sEMG-based MP3 Player Control System”*, Department of Automation, School of Information Science and Engineering East China University of Science and Technology, Shanghai, China 200237 †Key Laboratory of Advanced Control and Optimization for Chemical Processes (East China University of Science and Technology), Ministry of Education, 2013.