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Implementation of a Wireless Controlled Device with RGB Driving Based on FFT

Zhang Hong, Tian Chunwei, Lin Gaoya and Wang Hongbin

School of Computer Science and Technology
Harbin University of Science &Technology

zh_e_mail@sina.com

Abstract

A novel device which can play audio files with LED lights whose RGB effects are based on frequency spectrum analysis to the audio signal while it is playing is introduced in this paper. Not only can the device bring us with songs, but also it presents colorful lights while we appreciate music. RGB light mixed principle and the relations between music and light colors which are through LED lights driven by PWM are investigated. The device is composed of SD card, U disk to store medium files, the audio decoder to produce audio signals, speakers etc., and the all components are under the control of a microprocessor. The proper sampling rate is chosen for frequency analysis, and the Fast Fourier Transform (FFT) method is optimized to suite for the application. File system is implanted to the microprocessor system which possesses wireless communication Xbee, whose functions such as medium file selection can be controlled remotely.

Keywords: *Audio Signal, FFT, File system, Microprocessor, RGB*

1. Introduction

There are lots of audio players, including smart speakers in market. The work introduced is to develop a novel audio player with LED lights whose colors and volume change along with the music playing. Spectrums of frequency corresponding to dual-channel audio signals are displayed in LCD, and at the same time the two groups of LED lights also flash corresponding to the two speakers. The Russian pianist Skriabin said that he could find light in music field, and there were colors in front of his eyes when listening to music every time. This phenomenon shows that when we listen to music, some relations between hearing and sight color could be set up. When music is being played, sampled audio signals in time domain are transformed into signals in frequency domain by FFT. Frequency spectrum analysis has been extensively applied in different fields, and Fast Fourier Transform (FFT) is an effective method in practical application. The FFT method is optimized to suite for the system with microprocessor embedded which is short of memory storage. It needs less storage because the FFT reverse positions are optimized, and also the calculating the butterfly operation is modified.

Features of audio signals can be extracted by analyzing their spectrum of frequency from which signal amplitude distribution over frequency is obtained. The scheme that light colors and illuminations map to frequency spectrum is investigated, in which RGB LED lights driven by PWM outputs. In order to present perfect effect, colors of lights are determined by the frequency spectrum of audio signals. Colors and intensities of LED lights will change exactly corresponding to the music which is being played in real time. When music is

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appreciated, the exhibition by both songs and lights simultaneously creates wonderful effects. With wireless module XBee, the device functions can easily be controlled remotely, and many devices can work together synchronously to form a larger distributed sound and light system which can be appreciated from far distance. An individual device can be controlled through an XBee module which is based on ZigBee protocol [1]. Lots of individual devices which merge into environment can be organized by XBee modules to play music together or along with some other pattern to produce a special effect.

2. System Design

The system consists of a microprocessor STC12C5A60S2, a SD card, a U disk, and a audio file decoder VS1003, two speakers, two groups of LED light RGB display, and a LCD display etc. The system overall block figure is shown as in Figure 1. The SD card can be accessed by the microprocessor directly through SPI bus, but U disk should be accessed via the interface chip CH375. The interface between CH375 and the microprocessor STC12C5A60S2 is parallel as shown in Figure 2, so is that of the LCD12864. The VS1003 are connected to the STC12C5A60S2 via SPI also. The interface between the VS1003 and the STC12C5A60S2 is as shown in Figure 3.

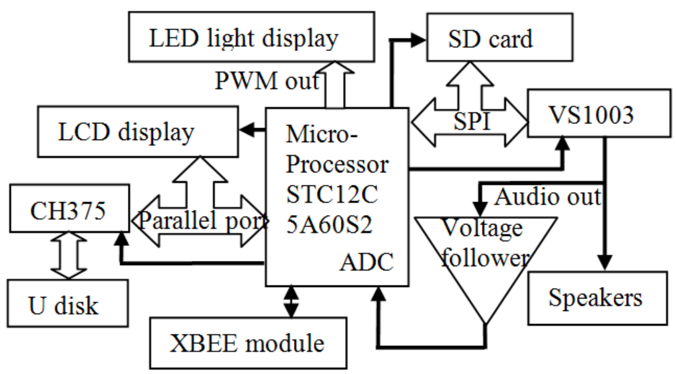

Figure 1. System overall block diagram

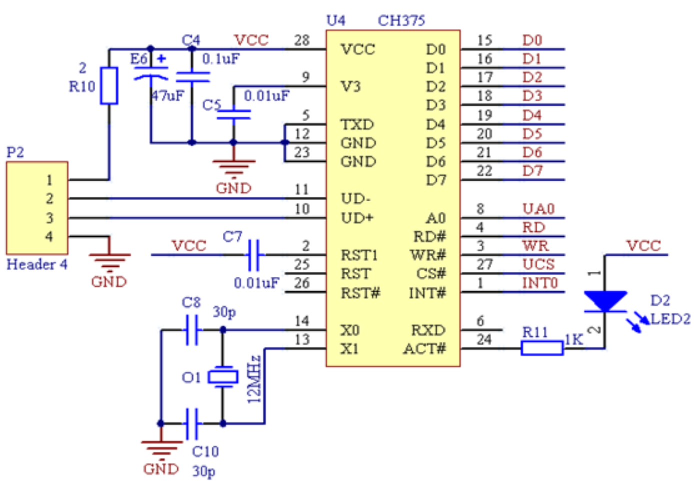

Figure 2. CH375 interaface diagram

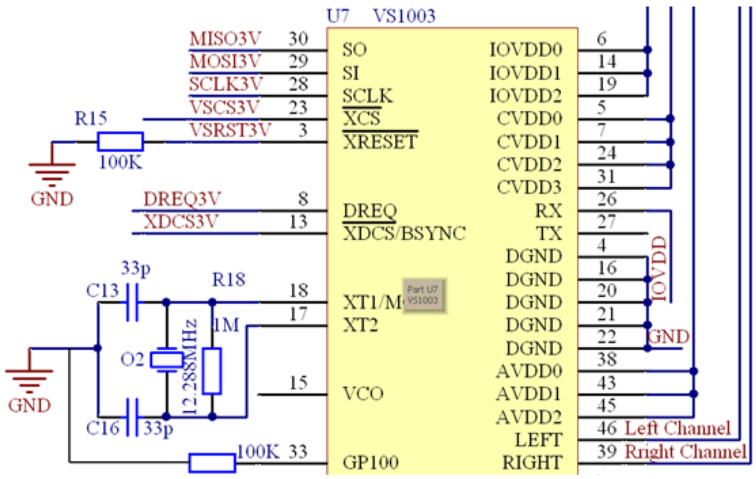

Figure 3. Diagram of interface for VS1003

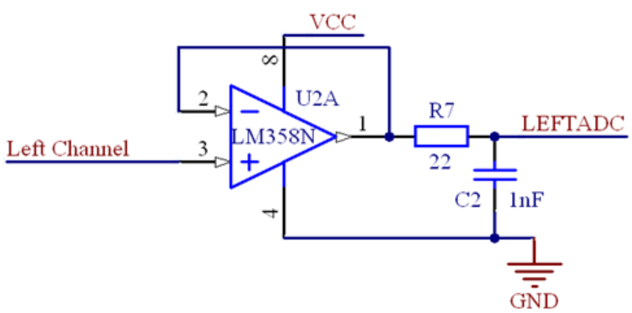

Figure 4. Voltage follower

The duel channel audio signal outputs of the VS1003 are left Channel and right channel signals. In order to play music fluently, data of audio files stored in the SD card or U disk should be read at a relatively constant speed by the microprocessor which then feeds the data to the VS1003 through the SPI. The VS1003 is blame for decoding the data whose form includes MP3/WMA formats, outputs duel-channel audio signals. The audio signals are sent into two ways, one way is to the double speakers, the other is to the two voltage followers whose output are sent to the two ADC ports of the microprocessor. The voltage follower for left channel audio signal is shown as in Figure 4. Aiming at decreasing noise interference, a RC filter is added to the output of the voltage follower which filters high frequency interference.

Sampled data of the two ADC ports are corresponding to the duel-channel audio signals that are the left and right channel respectively. ADC sampling rate is follow Naquist Sampling theorem, that if a sampling frequency is f_s, and the maximum frequency of a sampled signal is f_m, then $f_s \geqslant 2f_m$, so with the Sampled data, original signals can be recovered completely.

So the ADC ports for the left and right channel signals sample the analog signals at a fixed a sample rate f_s. Sampling points should be 2^n, such as 32, 64 or128, which constitute one group of sampled data. The interval between two groups of sampled data should take time for

FFT conversion and LED lights showing for watching that may lasts up to hundreds milliseconds. The sampled data corresponding to the duel audio signals are transformed into frequency domain by FFT algorithm separately. The frequency spectrums for the left and right channel signals can be shown on LCD display as bar figures. The characteristics of the audio signals can be extracted by analyzing their frequency spectrums.

Intensities and colors of LED light display are set up alone with amplitudes over frequency spectrums. The outputs of the two group LED light displays which are corresponding to the two channel audio signals are driven by the microprocessor PWM outputs whose duties are determined by RGB principle. RGB corresponds to the basic light colors red, green and blue. Each one group of LED lights is composed of red, green and blue lights. Different proportions of RGB mixes lights into different colors, which is implemented with PWM driving LED lights [2].

As it is described above, when the device works, a audio file is played, not only is the music outputted from the two speakers is appreciated by ears, but frequency spectrum bar figures corresponding to the two channels are displayed on the LCD display and at the same time, the two groups of LED RGB lights flashes corresponding to the two audio signal channels [3]. With the accompanying of colors, intensity of the LED lights presentation, the effect of music appreciations seems that the LED lights dance with rhythms of music and brings with wonderful feelings to people who are watching it.

3. Algorithm of Frequency Spectrum Analysis

For a signal in time domain $f(t)$, it's function of frequency spectrum $F(\omega)$ can be obtained by Fourier transform (FT). Discrete Fourier transform (DFT) should be used in digital system as data sampled are discrete values, a transform changes one domain value such as time in to frequency components and makes the signal processing far more easily than the time domain components [4]. As to a finite length sequence $x(n)$, it's DFT $X(k)$ is as (1)

$$X(k) = \sum_{n=0}^{N-1} x(n) W_N^{nk}, k = 0,1,\cdots,N-1 \tag{1}$$

Where $W_N^{nk} = e^{-j\frac{2\pi}{N}nk}$. But, when the number of sampled data increases, the transformation time will exponentially grow, so Fast Fourier Transform (FFT) is adopted in practical application. FFT has been applied for decades, and it is an effective method in audio signal analysis. FFT has advantages with less computational time by making use of the periodicity of W_N^r. The FFT method used in the embedded system should be optimized for less storage and time [5].

3.1. Reverse order analysis

In time signal sampling, for radix-2 FFT, input points are arranged in a binary bit reversal order, output points are in a normal order. Array $a[N]$ is used to store input points and medium arrangement. In order to prevent using more storage for data exchange, indexes of elements of the array is expressed in binary, and a table is setup for comparison between the index order and the index reverse order. To operating the reverse process by looking up the table needs less time and storage.

3.2. Butterfly computing algorithm

Data point N=2^M, the calculating process is divided into M levels, each level has N / 2 butterfly figures, The basic butterfly computation for level L is shown as in Figure 5. The computing formula is as (2) following the figure.

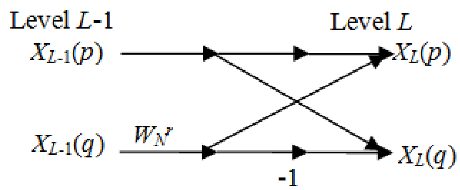

Figure 5. Basic computing of butterfly

$$\begin{aligned} X_L(p) &= X_{L-1}(p) + W_N^r X_{L-1}(q) \\ X_L(q) &= X_{L-1}(p) - W_N^r X_{L-1}(q) \end{aligned} \tag{2}$$

Assume q=p+B. The computed results contain real part and image part. The formula (2) can be written as (3) by using Euler's formula. The subscripts R in formula (3) means the real part, the formula corresponding to the image part is similar to (3).

$$\begin{aligned} X_R(p) = X_{(L-1)R}(p) + X_{(L-1)R}(p+B)\cos(2\pi r/N) \\ + X_{(L-1)L}(p+B)\sin(2\pi r/N) \end{aligned} \tag{3}$$

Formula (3) deals with sin and cos computation. A table containing values of sin, cos functions is set up for the formula computing, which can enhance the computing efficiency. There are four complex multiplication operations which under some special cases can be discarded in each butterfly computing. Figure 6 shows the FFT flow diagrams when N = 8. Rotate factors W_N^r possess repeatability. They take only one value $W_{N/4}^0$ in the first level, and two values in which one value is the same as that in the first level, and four values among which two values are the same as that in the second level. So some modification is considered when compiling the software for the special cases to avoid complex multiplication to reduce the computing time and storage.

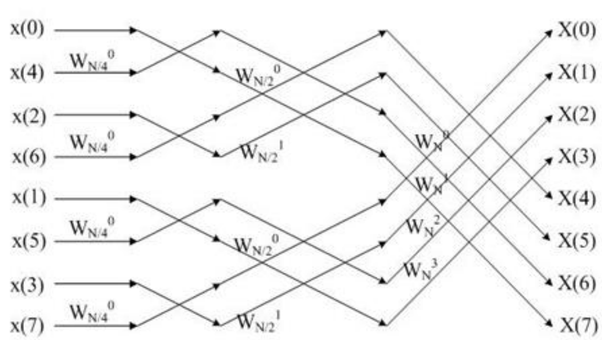

Figure 6. FFT butterfly flow diagram N=8

In level L, each butterfly has two input ports of data between them there are the interval of points $B=2^{L-1}$. Butterflies with an interval with points 2^{L} share the rotate factor. According to this rule, starting from an input port, from one level to the next, Operations for M levels operating can be performed. When a calculation is up to level L, different rotate factors has been derived step by step. When each rotate factor is coming out, the all butterflies of 2^{M-L} corresponding to the rotate factor have been finished. When a FFT program is compiled, triple loops are needed to carry out the operation.

4. Software Modules Analysis

The system program is developed in C programming language under environment Keil v4. The program contains functional modules such as LCD display, file system to operate SD card and U disk, audio decoder VS1003, FFT process, RGB LED lights driven by PWM outputs etc. Audio files which are usually large should be stored in SD, CF cards, or U disks. So to operate these kinds of memory is the key in the system development.

4.1. SD card operation via znFAT

To operate them conveniently, just as to read or save a file on disks on a computer, a file system should be adopted. To look for a file in a SD card, the operating upon the upper layer, does not care in which exactly sector where a file data locate, a file is indexed by its' file name. Like file system FAT32 of Windows [6], a file system znFAT that is suitable for systems with microprocessor embedded is applied in memory management. znFAT contains three layers as shown in Figure 7. The first layer is file operating functions for users' calling, the second layer is implementations of znFAT, and the third layer is driving program for storage devices. When it is in application, the first layer and the third layer are concerned, and the second layer which masks complex implementations of the file system. The main functions of znFAT shown in Figure 8 can satisfy most applications in file operations. Audio data are saved as files in storage such as SD cards. Operations on audio files rely on the functions of znFAT. The first step is to initialize a SD card to make sure the SD card exists, and then to initialize znFAT by call function znFAT_Init( ) to obtain the information of the SD card such as total size of the card capacity, byte numbers per sector, FATsector, next free cluster etc. To operate a file, the file should be opened first by call function znFAT_Open_File( ) with parameters such as the file name being conveyed. When a file is selected to play, after the file is opened, then the file data can be read out by calling function znFAT_ReadData( ), as len=znFAT_ReadData(&fileinfo,160*i,160, VSBuffer); if len equals to zero, it means the file ends, otherwise the file can be read continually. The first parameter of the function is file information structure &fileinfo, 160*i is the location from where data start to be read, the next 160, is the bytes of data read, VSBuffer is the buffer used to store the data read out.

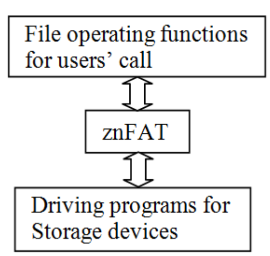

Figure 7. Structure of overall znFAT

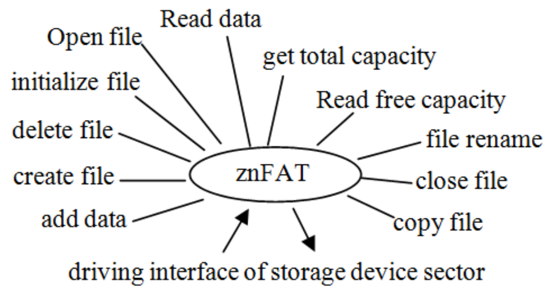

Figure 8. Functions of znFAT

4.2. VS1003 operation

A microprocessor communicates with a VS1003 through SPI serial bus. There are two ways for VS1003 SPI which are Serial Control Interface (SCI) and Serial Data Interface (SDI). VS1003 accepts commands by SCI and audio data by SDI from the microprocessor. Signals XCS, XDCS are synchronous signals for SCI and SDI respectively. SDI timing sequence is as shown in Figure 9. The VS1003 decoding rate should be kept consistent to the audio signal outputs. There is a data buffer with capacity of 512 bytes in VS1003, music correct playing needs to maintain the buffer not empty, so the data feeding to VS1003 should be 32bytes each time [7]. Before sending 32bytes, the microprocessor judges DREQ of VS1003, only when it is high, then to transfer the data.

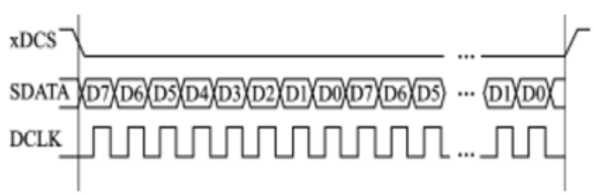

Figure 9. SDI timing sequence

5. Conclusions

Frequency spectrum analysis by the FFT method is applied in the system design, figures of frequency spectrum reflect time domain signals' change. If a signal changes rapidly, in frequency domain it has a wide frequency limit, similarly, if it changes smoothly in time domain, then it has a narrow frequency limit [8]. Figure 10 shows the FFT results corresponding to sample data with 64 points and 1024 points, at the same sampling frequency, the more sampling points, the higher the resolution, and the more densely populated spectral lines.

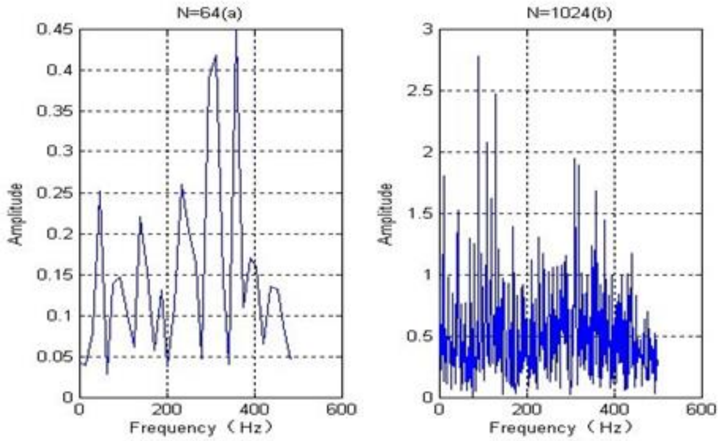

Figure 10. Frequency spectrum figures

The color and intensity of LED lights change according to frequency spectrum of an audio signal at a time. When music is played, audio signals spectrums are displayed and LED lights shining with the music rhythm, which will bring a new appreciation to our sense.

Acknowledgements

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