Computer Life (CPL) ISSN: 1819-4818
DELIVERING QUALITY SCIENCE TO THE WORLD

Application System Design of Audio Power Amplifier on Display Based on APA2604C

Guiyi Xu, Mingyu Zhang [a]

Suzhou Vocational Institute of Industrial Technology, Suzhou 215104, China;

[a]zmy_hust@163.com

Abstract: This paper studied an application system that can realize quantitative adjustment of volume, LCD liquid crystal display, and amplification of audio signals. The design of this system is based on STC12C5A60S2 single-chip microcomputer, APA2604C audio power amplifier, and LCD1602 liquid crystal display. There are two levels of hardware and software. The hardware is designed by Protel DXP, and the software is compiled by Keil. This system can be divided into three modules: one is the main control system based on STC12C5A60S2 microcontroller, with ADC conversion, PWM control, reset, external crystal circuit and other functions; the other is based on APA2604C power amplifier, through the circuit design, input audio Noise reduction, explosion protection and gain amplification; the third is based on the LCD1602 liquid crystal display, real-time display of the system volume status, the end customer visual adjustment is more convenient, the system rear speakers output gain audio, the entire system is compact, cost-effective, can be directly It is transferred to the audio power amplifier circuit of the current mainstream liquid crystal display.
Keywords: STC12C5A60S2 microcontroller, APA2604C audio power amplifier, LCD1602 liquid crystal display, Audio amplifier circuit.

---

1. INTRODUCTION

Liquid crystal display (LCD) is thin film transistor liquid crystal display (TFT-LCD). It is widely used in the display industry, mainly used in mobile phones, notebook computers, TV sets, intelligent interactive systems, etc. With the accumulation of technology and the development of modernization, the traditional CRT displays have been gradually eliminated, and the market share of LCD has gradually increased steadily. Due to the higher color depth, faster refresh rate, higher contrast, greater brightness and faster response time in LCD parameters, the mainstream resolution of 1080p in the current market is 1080p. The household display has reached 5K level, and 8K display has appeared in large size. High resolution display is coming to 5g era. The wide-angle technology used in LCD has been able to reach 178 °angle of view, which makes the color of the displayed picture more vivid without distortion. HDR technology makes the image softer and smoother. With the popularity of portable home appliances, the progress of LCD integration has been further improved. At present, Intel launched an integrated computer card in 2018, which can be embedded in the display, making the PC set display integrated. The early use of audio is more, now the speaker is built into the display, which further improves its cost performance. The performance of the audio power amplifier (AMP) has an important impact on the product quality [1-6].

Traditional audio power amplifier refers to linear amplifier, such as a, B, AB class. Although this kind of AMP has excellent performance in linearity and harmonic distortion, it has large volume, high internal loss of energy and high power consumption, resulting in efficiency below 50%. This defect is that this kind of AMP is difficult to be widely used in portable electronic products. With the development of technology and market demand, the disadvantages of traditional amp, such as low efficiency and large volume, are becoming increasingly prominent. At this time, class D power amplifier with high efficiency, low energy consumption, high fidelity and small volume is more in line with the market demand. In order to meet people's pursuit of high fidelity audio, class D amplifier develops rapidly, because class D amp has been integrated into the chip and can be directly embedded into the system to amplify the input small amplitude audio limit with high sound quality, making it a new favorite in the portable audio market.

Since the beginning of the 21st century, consumer electronic products have been widely used in the world, and each generation of products has more abundant and humanized functions. For example, mobile phones, tablet computers, ultra-thin laptops, smart watches, monitors, audio and other forms of audio playback devices have entered the rapid development period. These devices interact with people through pictures and sounds, and the sounds of these products The frequency output is realized by amp. Portable electronic products are carried by people. Battery technology restricts the product's endurance and life. The traditional amp has low working efficiency, and most of the energy is converted into heat. Therefore, the product should be equipped with a heat sink and protection device. This design greatly increases the design cost and has certain security risks. Because the power transistor of class D amp does not work continuously, it is similar to the inverter of digital circuit, and the internal amplifier is in the switching state. Therefore, class D amp is also called digital power amplifier in the industry. When the switch tube is on, the saturation voltage drop is small, which makes its efficiency hum high, generally more than 80%, and with the increase of output power, the efficiency is higher.

The development of AMP has experienced several decades of development. People have made different attempts and improvements in the circuit structure and the devices used in the circuit. At present, the technology has become mature. According to the time span, the technology accumulation of AMP can be divided into four stages as follows:

In 1876, Bell invented the telephone, which means that the conversion of voice signal into electrical signal can be realized initially. Soon after, the electronic tube was invented in Fleming's laboratory. Since then, the electronic age has entered. The general electronic tube with low noise and high stability coefficient has excellent high fidelity effect on audio, and the working state of the device is very reliable. The electronic tube uses the interaction of vacuum electron flow and electric field to amplify the signal. The components need to be made by hand, and the parts are relatively small Large size, large volume and weight, many parts, internal parts in the work will lose energy, into heat, so the efficiency is low, the cost is relatively high.

Feedback technology appeared in the 20th century. This technology includes positive feedback and negative feedback, and the most commonly used time negative feedback technology in electricity. Through setting sampling points on the back end, feedback to the input end, and analyzing and processing according to the set function relationship, the technology achieves the control input end. Generally, when the polarity of the input signal changes, the initial data superimposed on the back end will tend to the center line. Based on this theory, when applied to audio amplifier circuit, the impact of front-end noise on the back-end is small, which can effectively improve the guarantee performance of audio.

In the 1950s, the point contact germanium transistor developed by Bell Laboratories came out. With the advent of the transistor, it can realize the functions of amplification, calculation, switching, rectification, voltage stabilization, modulation, etc. In the 1960s, integrated circuits began to develop rapidly. With the application of semiconductor technology and photolithography, the chip was highly integrated, the product performance increased exponentially, and the development of various consumer electronic functions and supporting software was rapid.

There are two series of transistors, one is bipolar type, which is controlled by current, including NPN and PNP; the other is field-effect type, which is controlled by voltage, with N and P channels. Because voltage control means high impedance, the control terminal has no effect on the system circuit. It can control the gate through high-frequency electrical model, which is applied in audio amplifier circuit, with low noise, low power consumption, high fidelity and high reliability High frequency characteristics, similar to the electronic tube, are indispensable in the current electronic field.

With the development of modernization and the popularity of various wearable portable electronic devices, the degree of integration is very high, and the quality and price is low. The display and audio technology used in human-computer interaction has been greatly improved. The problem of power supply in mobile devices is dwarfed by the over development of product functions, which puts forward higher requirements for components, high integration, low price and power consumption In the field of AMP, class D amp is one of the best. Under the high frequency sound source, it can gain effectively and output high quality sound.

In the continuous design and development process of audio amp efficiency, class D amp system has been discovered. This kind of audio amp has a great advantage, that is, it can keep high efficiency output and achieve high audio quality output at the same time. In the past, the general audio amp usually sends the signal directly to the audio amp for processing without processing, which requires that the device must work in the saturation area. The static power consumption of the audio amp in this structure is relatively large, and the efficiency is very low. To achieve a certain power output, it needs to have high energy consumption, not only the energy utilization rate is reduced, but also the system works under high temperature The power output is also limited.

The main content of this paper is to achieve the design of audio amplifier circuit, which can be embedded in the display to amplify the input small signal amplitude audio, and ensure the low distortion of the output signal. The system uses a DC regulated power supply, the main control chip is stc12c5a60s2-lqfp48 single-chip microcomputer, its performance is excellent, it has two PWM channels, eight ADC converters, through the acquisition of voltage changes, ADC conversion through the output of the single-chip PWM signal of different duty cycle, as DC voltage directly acts on the volume pin of ap12604c to control apa2604c Amp audio gain effect, and through the LCD1602 LCD display to display the current system volume change. The whole system has the advantages of low cost, small size, high efficiency, precise control and safety and reliability.

2. SYSTEM HARDWARE DESIGN

The first is 5V DC power supply. This design is powered by mobile phone charger, the specification is 5V / 2a, as the energy supply of each unit of the system. General DC power supply equipment consists of three parts: transformer, rectifier bridge and voltage regulator. If the back-end requires high voltage stability, the MOSFET can be used to adjust the back-end power. Generally, the voltage is sampled through the back-end output, and the signal is isolated and fed back to the feedback (FB) pin of the MOSFET with optical coupler, so as to achieve accurate adjustment of the output power.

Two decoupling capacitors should be connected to the 5V power supply. As shown in Figure 1, 100 μ F and 0.1 μ f are designed in parallel. This is because there are low resistance parts in the lead wire and components in the circuit, which will generate instantaneous current and partial voltage at the moment of power on, which will lead to system voltage fluctuation, which is easy to cause misjudgment of the back-end chip and has a great impact on EMC. There are only two kinds of general processing methods: one is short wide wiring and the other is changing compact high components. These methods are not recommended because of the space constraints of the substrate and design cost; the other is to add a decoupling capacitor cp10.1 μ f around the components to remove high-frequency noise in the power supply, and C303 100 μ f is used to stabilize voltage and filter low-frequency noise.

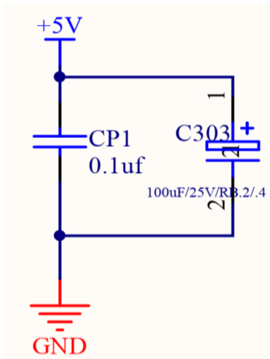

Fig. 1 Design of 5V DC power supply

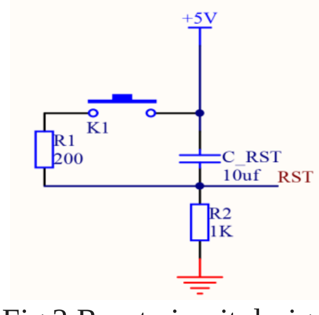

Fig.2 Reset circuit design

The second is reset circuit. STC12C5A60S2 microcontroller generally has five reset modes. This design uses external RST pin to reset. When the rst pin of MCU receives a certain width of reset pulse, that is, high level, the system will reset. When the level is low, the program will run from 0000H after reset, as shown in Figure 2.

This design first configures RST pin from I / O port to reset pin. When the button is on time, 5V power is supplied to RST pin, C_ The rst charging time is directly proportional to the value of RC, then the delay time is R (K Ω) * C (μ f) = 0.2 * 10ms = 2ms, the reset time should be greater than the time of two machine cycles, and the machine cycle is 12 * 1 / 22.1184m = 0.543 μ s, so the reset time of this design adopts 1rc, that is, 2ms. In this design, R1 can limit the charging current and control the reset time. R2 is designed as C_ Rst discharge.

STC12C5A60S2 series single-chip microcomputers have integrated R / C internal clock. However, due to the unified calibration of IC at the factory, the accuracy of the actual configuration circuit is not high. Generally, the external high-speed clock can achieve higher clock accuracy. As shown in the figure below, the load capacitance C102 is equal to C103, which forms a three-point capacitor oscillator with crystal Y1. This design uses a 22.1184MHz crystal oscillator. According to the theoretical estimation, the parasitic capacitance of crystal oscillator and PCB is about C0 = 10PF, then cy1-c0 = C102 * C103 / (C102 + C103), and calculate C102 = C103 = 20pF. Then, query the load capacitance from the data manual, and take 20pF to meet the design requirements. The design drawing is shown in Figure 3.

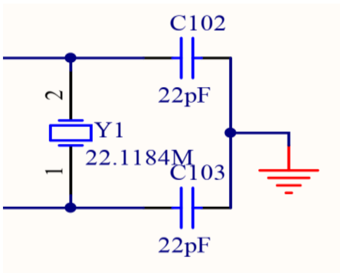

Fig.3 Circuit design of crystal oscillator

Because APA2604C AMP has many functions, excellent sound quality processing effect, large gain range, multiple control orders, easy to adjust and design, and has a high cost performance ratio, this paper selects this amp as an amplifier. In this design, the amp mainly amplifies the input audio signal, adjusts the output power through the PWM signal of STC12C5A60S2 single chip microcomputer, and finally outputs the audio after gain. The peripheral circuit design is shown in Figure 4.

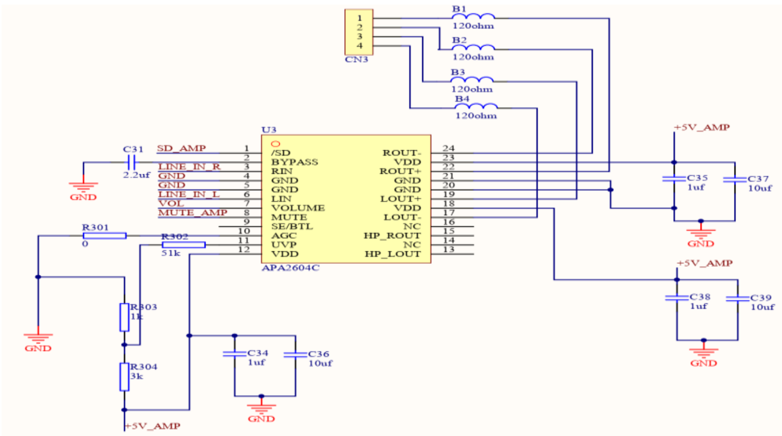

Fig.4 Apa2604c amp peripheral circuit design

In order to ensure the normal audio output of the whole frequency band, this paper briefly describes several design points of apa2604c amp peripheral circuit. Firstly, the pin SD controls the working state of the amp, and the IC will be started at high power level; bypass is the bias voltage of the amp, and the bypass capacitor is very important for the low noise design of the power supply side, and will affect the start-up time of the AMP (in direct proportion). Generally, this capacitor is larger than the filter capacitor at the input end to avoid. In this design, 2.2 μ f is selected, and the layout is as close to the pin as possible to increase the power Source stability. Volume is a PWM signal. The DC voltage generated by PWM with different duty cycle is different, so the internal gain setting is selected based on this. Mute is the mute function, which works normally at low level. The output is closed immediately at high level, and the back-end output is stable with 50% duty cycle. The smooth and static function built in the amp can prevent crackle. AGC output maximum power control, when the output reaches the maximum power, the amp will Automatically reduce the gain effect, prevent output clipping, so as to avoid damage to AMP and horn.

UVP is an undervoltage protection setting. This design collects the voltage of R303. The resistance value of R302 is much greater than that of R302 // r304. The turn off threshold of AMP UVP pin is 1.25V. Therefore, when the external voltage is lower than vlow, it will enter the closing mode. Vlow = [1.25 - (6 μ axr302)] x (R303 + r304) / R303 = 3.776v. Generally, considering the influence of hysteresis, the system is pulled to 4.8V or above to keep the system stable.

Because the power supply part of this amp needs to be fully decoupled, the parallel design of C134 = 1 μ F and C36 = 10 μ f is selected to prevent the oscillation and coupling on the audio line, so as to ensure low total harmonic distortion (THD). Since the capacitance will directly affect the low-frequency performance of the circuit, it is appropriate to choose C13 & C14 = 1 μ F. at the same time, it is necessary to consider the leakage path from the audio input source through the input network to the load, which will generate DC bias voltage at the input end. The audio is input to the left and right channels through Lin & Rin, and the audio output port is conducted through pins 17, 19, 22 and 24. Considering the short wiring from the amp to the horn, the ferrite bead filter can effectively absorb the high-frequency signal (about 1MHz or more), and the impedance is very low at low frequency. In this design, 120ohm magnetic beads are added to the four output channels of the output.

STC12C5A60S2 single chip microcomputer has 8-channel 10 bit ADC and two-way PWM. One channel is selected as the input of volume adjustment in 8-channel ADC. In this design, STC12C5A60S2 microcontroller as the main control equipment, knob rheostat and 20K resistance partial voltage. When adjusting the knob, the microcontroller collects the voltage through ADC, calculates a PWM duty cycle according to the voltage, and then provides the volume regulation signal for amp IC with PWM pin.

The p2.2 pin of STC12C5A60S2 single-chip microcomputer is defined as low-level. When the single-chip microcomputer works and the LED is on, it indicates that the system has been started, which is convenient for users to identify. The program memory of this IC is compiled independently. Flash is integrated into the IC. The main program is burned by RX and TX, which is convenient for debugging. The pin XTAL is the external high-speed clock, which can improve the long-distance stability of STC MCU. P0 is the data transmitted from single chip microcomputer to LCD1602. Under high frequency signal, 0 Ω resistance is connected in series as inductance and capacitance, which can effectively solve the EMC problem. At the same time, the zero even resistance current bearing capacity is weak. When short-circuit and over-current occur, the resistance will be fused first to protect the circuit. STC12C5A60S2 MCU communicates with LCD1602 through 11 pins. D0-d7 is the data bus used for signal output. E, RW and RS are used to control the display status of LCD1602. At the same time, it has reset circuit, mute, ADC sampling, PWM control and other functions. The main control circuit is shown in Figure 5.

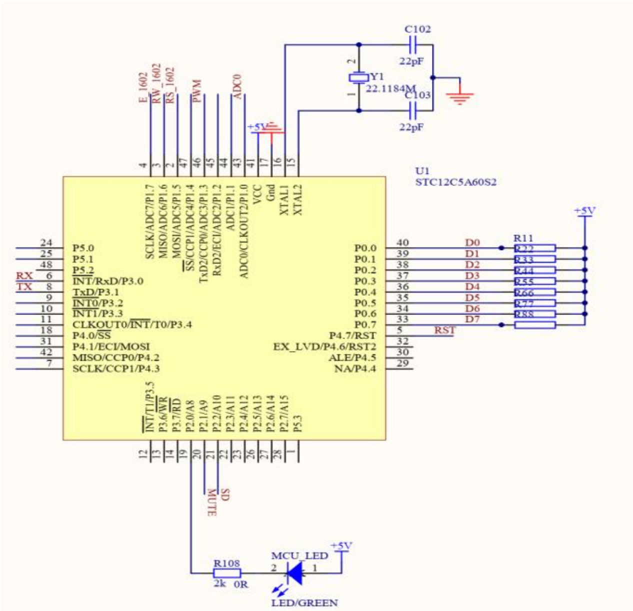

Fig.5 STC12C5A60S2 MCU main control circuit design diagram

LCD1602 LCD display can only display 32 characters, and the working voltage is DC 5V. LCD1602 through STC MCU P0 corresponding to d0-d7 channel input signal, LCD internal controller to process the signal, after processing the signal to the LCD driver for LCD display. LCD1602 is an external device. It takes a certain time to process and output signals. It needs to read busy signals for judgment. RS, R / W and e screen input signals before normal display. The design circuit is shown in Figure 6.

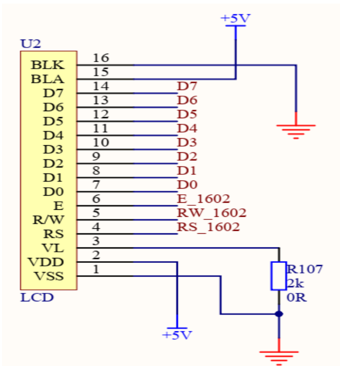

Fig.6 Circuit Design of LCD1602

P0.1-p0.7 can be used as input / output port (at this time, it is a quasi two-way port with 8-bit quasi bidirectional port, if there is pull-up resistor inside, there is no need to connect external pull-up resistor), or it can be used as address / data multiplexing bus (as low-8-bit address line a0-a7, data line d0-d7). VL pin can adjust the contrast of LCD. In the rated voltage, the higher the voltage is, the smaller the contrast will be. At that time, if the contrast is too high, it will produce residual image. This design uses 2K Ω to avoid such problems.

Because when adjusting the knob, the voltage change is a continuous analog quantity. In order to make STC microcontroller be able to recognize this kind of data, it is necessary to convert analog data into digital quantity, which uses analog / digital converter, namely ADC. The external crystal oscillator circuit used in this design is up to 22mhz, which is much larger than the human hearing range of 20-20KHz. The higher sampling frequency can effectively ensure the original sound quality. This design uses the knob rheostat with switch function. When adjusting the knob, first turn on the system power, then adjust the resistance value. The adjustable range of the variable resistor is 50K Ω. When it is selected to 0 Ω, the volume is zero, and when it is selected to 50K Ω, the output is 100 volume. Using linear change design, setting 0.5K Ω as a volume, the fluctuation range of the whole ADC sampling resistance is 20k-70k ohm, which consumes very little power. When the STC MCU receives the voltage signal, it starts AD conversion to digital signal, and then according to the conversion value in the program corresponding to the PWM value, it outputs to apa2604c amp to push the loudspeaker to play, as shown in Figure 7.

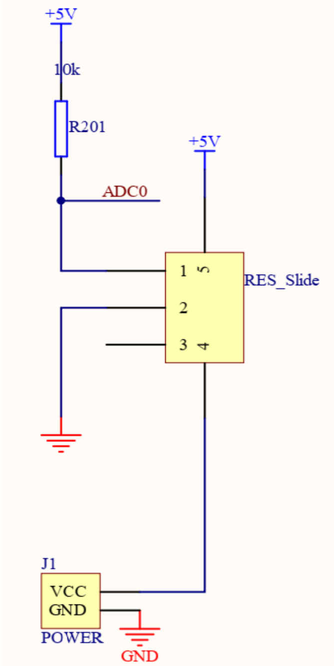

Fig.7 Volume Circuit Design

3. SUMMARY

This design uses tc12c5a60s2 single-chip microcomputer as the main controller of the audio amplification circuit system. The PWM signal is transmitted to apa2604c power amplifier to amplify the small amplitude audio signal. At the same time, it can display the volume in real time through data interaction with LCD1602. The main control circuit has the advantages of operation visualization, integration degree and high cost performance, and it is also the best state for debugging the peripheral circuit of the system. The components and chips used in this design are common components, and the system power supply design also conforms to the current design specifications, so it can be directly embedded into relevant equipment for application. In the design process, it is necessary to debug the hardware (filter, coupling, voltage divider, EMI, etc.) and software (including ADC, PWM, LCD, timer, amp initialization module) frequently to make the system have the optimal sound quality effect.

REFERENCES

[1]. Sun T , Gao Y , Gu Y . Studies on Information Hiding Method Based on Spread Spectrum Technology and Dual MP3 Audio Files[J]. 2010:412-416.
[2]. Biniak B , Cunningham C , Ivanov A , et al. SYSTEM FOR PROVIDING SECONDARY CONTENT BASED ON PRIMARY BROADCAST[J]. 2008.
[3]. Zhi-Ling B . The Design of Audio-visual Files Information Management System Based on C/S[J]. Journal of Minjiang University, 2006.
[4]. Wang D S , Zeng Y F , Wang J . Design of Intelligent Addressable Broadcast System Based on Zigbee[J]. Applied Mechanics & Materials, 2013, 385-386:1546-1549.
[5]. Sauer C , Roth-Berghofer T , Auricchio N , et al. Recommending Audio Mixing Workflows[C]// International Conference on Case-Based Reasoning. Springer, Berlin, Heidelberg, 2013.
[6]. Crispien, Kai, Petrie, Helen. Providing Access to Graphical-Based User Interfaces (GUIs) for Blind People: Using a Multimedia System Based on Spatial Audio Representation[J]. 1993.