A Low Frequency Power Amplifier Design Based on Output CapacitorLess Circuit

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Abstract: This low-frequency power amplifier utilizes the AT89S51 microcontroller as the main control chip. The system can be divided into several modules: power amplification circuit module, pre-amplification module, band-stop filtering module, protection circuit module, volume control module, power supply voltage, AC voltage, and AC current testing module, A/D conversion module, power module, and single-chip microcomputer subsystem control module. We chose class AB audio power amplifiers. To ensure circuit stability and higher fidelity, we selected the TDA2030 integrated chip to build the OCL power amplifier circuit, preventing efficiency from being too low. Through the control of single-chip microcomputer subsystem, it can measure and display output power, the supply power of the DC power source, and overall efficiency. This system utilizes electronic switches connected with band-stop filters, with the band-stop frequency ranging from 40～60Hz.

Keywords: Power; amplification; TDA2030; AT89S51.

1. Introduction

Audio power amplifiers constitute a mature field in technology, where relentless efforts have been made over decades, resulting in significant progress in circuit technology, components, and conceptual understanding. Reflecting on the developmental journey of power amplifiers can be intriguing for audio enthusiasts[1][2].

The history of audio technology can be divided into four stages: tubes, transistors, integrated circuits, and field-effect transistors. In 1906, American Lee De Forest invented the vacuum triode, marking the inception of human electroacoustic technology. In 1927, the Bell Labs introduced negative feedback technology, propelling audio technology into a new era. Representative amplifiers such as the "Williamson" amplifier successfully employed negative feedback technology, significantly reducing distortion. By the 1950s, the development of vacuum tube amplifiers had reached its peak, with various tube amplifiers emerging[3]. The sweet and mellow sound of vacuum tube amplifiers still remains favored by audiophiles.

The advent of transistors in the 1960s opened up a broader soundscape for audio enthusiasts. Transistor amplifiers offered delicate and captivating sound, lower distortion, wider frequency response, and dynamic range. In the early 1960s, the United States introduced a new member in audio technology—integrated circuits. By the early 1970s, integrated circuits gradually gained recognition in the audio industry for their high quality, affordability, compact size, and versatile functions. Till now, thick-film audio integrated circuits and operational amplifier integrated circuits are widely used in audio circuits.

In the mid-1970s, Japan produced the first field-effect power transistor. Due to its characteristics of pure and sweet sound akin to vacuum tubes, as well as a dynamic range of up to 90dB and THD<0.01% (at 100kHz), field-effect transistors quickly gained popularity in the audio world. Many amplifiers today utilizes field-effect transistors as final stage outputs.

In the 1980s, digital amplifiers emerged as the favorite of a new generation. As we are now in the information age, coupled with technical training and guidance from major brands, the installation technical level in large cities does not differ much, but there are not many high-level installers. The vast majority only stay at the basic installation level, merely learning many superficial decoration techniques, without focusing on sound quality and sound field, deviating from the ultimate goal of audio[4].

2. Comparisons and Selection of Power Amplifiers

2.1. Class A Audio Power Amplifiers

Class A audio power amplifiers operate in a good linear state and do not produce switch distortion or crossover distortion, as long as biasing and dynamic range control are properly managed. From a distortion perspective alone, they represent a high-quality amplification circuit[5]. However, Class A amplification circuits have relatively low efficiency, typically around 30% in practical operation. Especially noteworthy is the significant heat generated by the output transistors serving as Class A power amplifiers. Moreover, they have high requirements for stability, circuit design, and operational feasibility.

2.2. Class B Audio Power Amplifiers

Class B audio power amplifier circuits have a static operating current set near zero at the transistor base current. When there is an excitation signal, only one branch conducts, resulting in relatively high efficiency. However, these amplifiers suffer from high crossover distortion, making it challenging to meet the requirements. Therefore, this approach is abandoned.

2.3. Class AB Audio Power Amplifiers

Class AB Audio Power Amplifiers can be considered as a combination of Class A and Class B audio power amplifiers[6]. Their static operating current lies between

Class A and Class B, with relatively low operating current and high efficiency. Even with small signal inputs, they maintain high efficiency. In this type of amplifier, when the output level is below the set value, the output power transistors operate in a slightly conducting state, resulting in very low crossover distortion.

2.4. Class D Audio Power Amplifiers

Class D audio power amplifiers are a type of switch-mode audio power amplifier where the output power transistors operate in a switching state, greatly reducing power consumption and significantly increasing efficiency. However, to control distortion rate, the circuit's sampling frequency needs to be relatively high, which inevitably results in some electromagnetic radiation, potentially interfering with other electronic devices. If fidelity is a primary concern, at least four inductors are required in the LC filter, which increases costs[1].

In comprehensive comparison, considering factors such as circuit stability, efficiency, distortion rate, and cost, we choose Solution Three.

Since the input voltage is only 0~5mV, we place the electronic switch and volume control module after the small signal amplification. Recognizing the potential interference between power sources, each module has an independent power supply. The OCL circuit uses ±15V power supply, the volume control module uses ±12V power supply, and the single-chip microcomputer subsystem uses a 5V power supply. As the speaker is directly connected to the power amplifier circuit output, a protection circuit is needed to prevent speaker damage.

3. Theoretical Analysis and Calculations

3.1. Selection and analysis of power amplification circuits

3.1.1. OTL Circuit

When the DC operating voltage is relatively high, the OTL circuit can achieve a large output power with high efficiency. However, since only one power supply is used, a coupling capacitor needs to be connected at the output end of the circuit. This limits the bandwidth, making it unsuitable for our requirements. Therefore, this approach is abandoned.

3.1.2. Full Symmetrical OCL Circuit

The OCL circuit, without output capacitors, utilizes direct coupling between the amplifier and the speaker, resulting in excellent low-frequency characteristics. This form of power amplifier circuit connects the differential input amplification, excitation, and power amplification in a complementary symmetrical manner, fully utilizing the advantages of complementary transistor operation. It improves the open-loop performance, has good transient response, makes the circuit more stable, better reduces harmonic distortion, and can also prevent high-frequency self-oscillation. However, due to the difficulty in sourcing dedicated field-effect transistors for amplification and the challenging debugging process, we have also decided to abandon this approach.

3.1.3. TDA2030 Integrated Circuit OCL Power Amplifier Circuit

The TDA2030 is an excellent power amplification integrated circuit known for its high slew rate and low transient intermodulation distortion. Its peripheral circuitry is simple, making it easy to use and affordably priced. It offers high output power with relatively comprehensive protection features. However, due to the high voltage and output current requirements of high-power integrated blocks, careless handling during use can often lead to damage. Nevertheless, the TDA2030 integrated circuit incorporates a relatively sophisticated protection circuit. It automatically reduces current or shuts down in case of excessive output current or overheating, ensuring its own protection.

The power supply voltage ranges from±6~±18V. It delivers high output current with minimal harmonic distortion and crossover distortion(±14V/4Ω，THD=0.5%).

It can be connected in either single-supply or dual-supply configurations. Here, we have chosen the dual-supply configuration. The circuit schematic is illustrated in Figure 1 as shown.

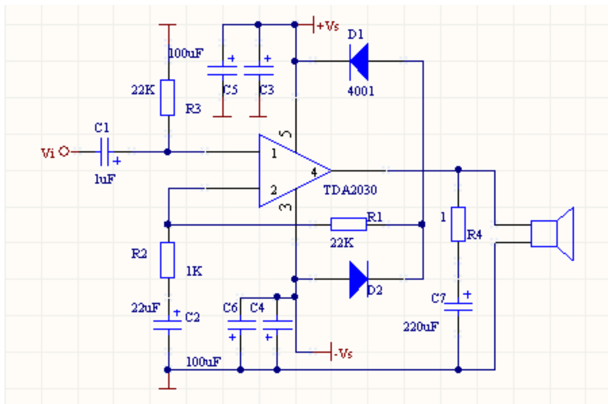

Figure 1. OCL circuit

Pin 1 of the TDA2030 is the inverting input. Capacitor C1 serves as the coupling capacitor. If the audio signal input from the previous stage has no DC component, C1 can be omitted to improve low-frequency response. R3 is the biasing resistor, typically chosen to be several tens of ohms. If the resistance value is too small, it will reduce the input resistance value of the amplifier.

Pins 5 and 3 are respectively the positive and negative power supply terminals. If the power supply leads to the amplifier are relatively long, filter capacitors must be connected near pins 5 and 3 to eliminate self-excitation caused by coupling due to the resistance of the power supply leads. This is known as decoupling. Capacitors C3, C4, C5, and C6 in the diagram serve this purpose.

Large-capacity capacitors are used for low-frequency decoupling. However, due to the presence of inherent inductance in large-capacity capacitors, they are unable to effectively decouple high frequencies. Therefore, it is common practice to parallel small-capacity capacitors with large-capacity ones to decouple high frequencies. Small-capacity capacitors have minimal inherent inductance, making them more effective at decoupling high frequencies.

Pin 2 serves as the inverting input terminal, forming a mixed AC-DC negative feedback circuit here. This configuration stabilizes the static operating point, ensuring that the DC voltage at the output remains stable around 0V. The output signal is derived from a voltage divider circuit consisting of R2 and R1. A small portion of the output signal is fed back to pin 2, forming a closed-loop amplifier. The gain of the amplifier is determined by the ratio of the resistance values of R2 and R1.

Pin 4 serves as the signal output after audio amplification, which can be directly connected to the speaker for sound reproduction. D1 and D2 are output protection diodes used to eliminate excessive peak voltages. Since the speaker is an inductive component, it generates momentary peak pulses when the current changes abruptly.

R4 and C7 form a high-frequency RC absorption circuit. Due to the high gain of the integrated power amplifier and potential coupling interference resulting from improper circuit design, it is easy to generate high-frequency self-excitation. When high-frequency self-excitation occurs, it can lead to electromagnetic radiation interference, exacerbating the self-excitation and causing the power amplifier IC to overheat and burn out. The high-frequency self-excitation generates ultrasonic waves, which are imperceptible to the human ear. C7 is used for high-frequency bypassing, while R4 serves as a damping resistor to dissipate the energy of the high-frequency self-excitation (ideally, capacitors do not dissipate energy in theory), preventing vicious cycles of self-excitation.

If the capacitance of C7 is chosen to be too large, it will cause the frequency response of the amplifier's high-frequency end to decrease. The capacitance value shown in the diagram is the recommended value.

For the differential input operational amplifier circuit, end current measurement we can find: Suppose u_{11} acts alone while $u_{12}=0$. In this case, the circuit is equivalent to an inverting proportional operation circuit. We can obtain the output voltage u_{o1} produced by u_{11} as:

$$u_{o1} = \frac{-R_f \cdot u_{11}}{R_1} \tag{1}$$

Let's suppose when acting alone, $u_{11}=0$, Then, the circuit becomes an in-phase proportional operation circuit, and we can find the resulting output voltage u_{o2} for:

$$u_{o2} = (1+\frac{R_f}{R_1})u_p = \frac{(1+\frac{R_f}{R_1})\cdot R_f \cdot u_{12}}{R_1' + R_f'} \tag{2}$$

The total voltage is:

$$u_0 = u_{01} + u_{02} = \frac{-R_f \cdot u_{11}}{R_1} + \frac{(1+\frac{R_f}{R_1})\cdot R_f' \cdot u_{12}}{R_1' + R_f'} \tag{3}$$

When $R_1 = R_1', R_f = R_f'$,then:

$$u_0 = \frac{(u_{12} - u_{11})R_f}{R_1} \tag{4}$$

Therefore, we have:

$$i_o = \frac{u_o}{R_l} \tag{5}$$

3.2. Bandstop filter module

An active bandstop filter circuit is composed of a low-pass filter, a high-pass filter, and an in-phase proportional circuit. This circuit is slightly more complex than the passive bandstop filter circuit, but it is easy to debug and accurate.

Therefore, we choose the active bandstop filter circuit. The specific circuit schematic is shown in Figure 2.

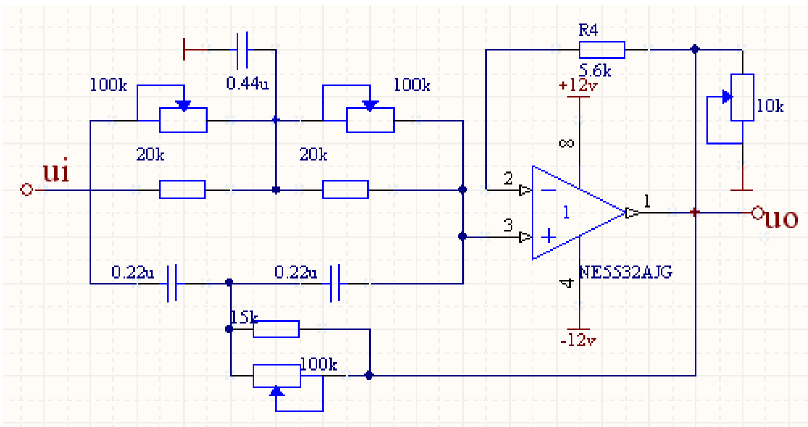

Figure 2. Active bandstop filter circuit

3.3. Protection Circuit Module

Since the speaker is directly connected to the output terminal of the power amplifier circuit, the output transistors can be damaged due to high current flow in case of strong signal input or output load short-circuit. Moreover, the speaker may not withstand the impact of high currents during strong signal input, power on, or power off. Therefore, a short-circuit protection circuit must be set up. The circuit consists of two parts: overload detection and amplification circuit, as well as relay. When the amplifier output is overloaded or the midpoint potential deviates significantly from zero, the overload detection circuit outputs an overload signal. After amplification, the relay is activated to disconnect the speaker circuit.

3.4. Preamplifier Circuit

Since the output signal amplitude of general signal sources is small, typically with gains of only a few tens of dB, it is difficult to achieve significant power output from power amplifiers. Therefore, it is necessary to add one or even multiple stages of voltage amplification circuits before the power amplifier to amplify the small signal from the signal source to the level required by the power amplifier. These types of amplifiers are called preamplifiers.

3.4.1. Using OP07 Operational Amplifier

This circuit has a simple structure, high input impedance, and adjustable gain. However, its common-mode rejection ratio is relatively small, reaching only 104 in actual testing. Therefore, we are abandoning this option.

3.4.2. Using OPA227 Operational Amplifier

The circuit schematic is shown in Figure 3. The advantages of this circuit are high input impedance, large common-mode rejection ratio, low total harmonic distortion (THD), very fast slew rate, and low noise. It can meet the requirements of the task, hence this option is chosen.

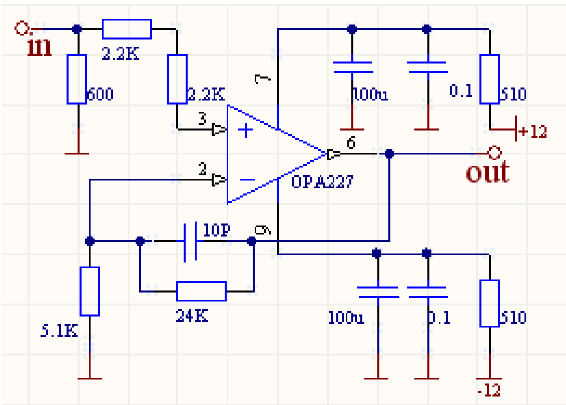

Figure 3. Preamplifier circuit

3.5. Volume Control Circuit

Volume control refers to controlling the size of the signal transmitted from the pre-amplifier circuit to the power amplifier circuit, thereby controlling the volume of the reproduced sound. Generally, there are two main types of volume control circuits: attenuator volume control circuits and electronic volume control circuits. Since we have created an electronic switch module, we choose the electronic volume control circuit here. The characteristics of this circuit are low power consumption, low distortion, volume control via buttons, overcoming the shortcomings of traditional potentiometers such as short lifespan and high noise.

3.6. A/D conversion module

3.6.1. Using AD0809 as the A/D Converter

The ADC0809 is a CMOS integrated circuit and belongs to the successive approximation type of converter with an 8-bit resolution. It has tri-state output latches at the data output terminal, enabling direct connection to the single-chip microcomputer's data bus. With 8 channels, it directly connects to 8 analog signals for conversion. However, the wiring between the ADC0809 and the single-chip microcomputer is relatively complex, and its precision in collecting various signals is not sufficiently high.

3.6.2. Using TLC2543 as the A/D Converter

The TLC2543 has the advantages of the ADC0809, and it has 11 channels, allowing conversion of 11 analog signals. It has a higher precision in collecting various signals and its connection to the single-chip microcomputer is relatively simple. Therefore, we choose the TLC2543 as the A/D conversion module.

3.7. Microcontroller Small System Modules

Using AT89S51 as the main control chip, with 5 independent buttons and a liquid crystal display module connected externally, the main function is to control electronic switches and volume, and to perform parameter testing and various displays. Compared with similar graphic dot matrix LCD display modules, the LCD display composed of ST7920 is much simpler in both hardware circuit structure and display program. Moreover, the price of this module is slightly lower than that of the same dot matrix graphic LCD module. Therefore, we choose to use ST7920 as the display module.

4. System Testing and Analysis

4.1. Testing Equipment and Models

Dual Channel AC Millivoltmeter (Model: AS2294B)
Digital Multimeter (Model: DT9205M)
Oscilloscope (Model: GPS-620)
Signal Generator (Model: ZQ1630)

4.2. Testing Method and Test Data

Testing was conducted using a 1 kHz sine wave signal and a sampling resistor of 0.44Ω. The measurement results are shown in Table 1.

Table 1. Parameter test display table

Number	Sampling voltage/V	Total voltage/V	Power supply power/W
1	0.05	20.5	2.10
2	0.08	19.8	3.65
3	0.10	20.1	4.7
4	0.15	29.8	11.2

4.3. System Debugging

The hardware debugging mainly focuses on the pre-amplification module, volume control module, and OCL power amplifier circuit module, while the power supply circuit is relatively simple. The power supply can be divided into three parts: providing 5V power supply for the single-chip microcomputer, providing ±15V power supply for the OCL power amplifier circuit, and providing ±12V power supply for other modules. We can directly judge the status of the power supply circuit by the LED lights. If the LED lights are emitting light normally, it indicates that the power supply circuit is working properly.

The OCL power amplifier circuit is the core part of this system. After soldering the circuit board according to the schematic diagram and fixing the heat sink on the TDA2030 integrated chip, connect it to the ±15V power supply. Firstly, test the output terminal voltage. If it is 0V, the circuit is normal; otherwise, the circuit needs to be checked again. Then, we input a sine wave signal into the circuit and measure the waveform at the output terminal with an oscilloscope. If it is an amplified sine wave, the circuit is normal. Next, we use an MP3 player to input a signal and listen to its sound quality to further refine the circuit.

Connect the pre-amplification circuit and perform the above operation again. If the same result is obtained, it indicates that the circuit is also normal. During this operation, we accidentally connected the signal line to the ground wire, resulting in no signal output in the end. The reason was found after careful inspection.

After the above circuit tests are completed without any abnormalities, we connect the volume control circuit and the electronic switch module to the circuit for testing. We also use the waveform changes to judge whether the circuit is normal. After inputting a sine wave, connect the oscilloscope to the output terminal, and then control the buttons to observe whether there is a change in the waveform amplitude. If there is a normal change, then this circuit is also normal.

5. Conclusion

This system has successfully completed all the contents included in the project, with a relatively simple design and good overall performance. In this design, we have also gained a lot, such as the importance of combining software and hardware. Simple theoretical knowledge is far from enough for us. We must connect it with practice in order to better consolidate our professional knowledge and cultivate independence in solving various problems encountered in practice. This design has a lot of content and is somewhat difficult, but my gains have also been richer. For example, it has made me more familiar with the working environment of AD16 and more proficient in wiring and layout operations; There has been a certain improvement in the theoretical analysis and design of circuits; I have gained a deeper understanding of the application of instruments such as oscilloscopes and signal generators as I design them, and I am proficient in operating them. Of course, we also encountered many difficulties in this process, such as the circuit not achieving the expected results during debugging, and sometimes even not working properly. However, after team discussions and our own continuous thinking and inspection, we can always find a solution and the design will not be interrupted.

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