

Portable Class-D Type Audio System Design with Fast Walsh Transform and Adaptive Noise Cancellation Method

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Abstract

The electrical and electronic systems can be divided into two wide categories generally referred to as analog and digital systems. Since 1970's engineering system and industry have been rapidly digitizing with the development of electronics. Digital techniques are useful because it is easier to get an electronic device to switch into one of a number of known states than to accurately reproduce a continuous range of values. In this paper, a design technology is introduced to implement mobile and wireless Class-D type audio system with adaptive noise cancellation algorithm based on the fast Walsh transform. The proposed adaptive cancellation algorithm is very efficient and electrical performance characteristics such as signal distortion and efficiency have shown good results.

Keywords: Class-D Type, Mobile and Wireless Transmission, Fast Walsh Transform, Adaptive Noise Cancellation

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1 Introduction

In 1909, Lee De Forest invented an audio sound system with triode vacuum tube. The triode was a three terminal device with a control grid that can modulate the flow of electrons from the filament to the plate. The triode vacuum amplifier was used to make the first AM radio. Then, the audio system also rapidly converted to digital system according to the development of digital technology. Accordingly digital audio

technology during the 1970s, rapidly replaced analog audio technology in most areas of sound reproduction, amplification and communication. Digital audio system may comprise a compression, storage, and transmission components for the audio signal processing. Forecasting is a scientific process of using historic data to determine the direction of future trends. Every forecast is based on a selected model and by different approaches of dynamic variables it will have different strengths and weaknesses. In general, audio system can be classified into following four types, Class-A, B, AB and D type according to system characteristics such as electrical output efficiency, distortion factor, linearity and total harmonic distortion. In a Class-A type, bias current flowing in the output appear so its efficiency is about 20%. The Class-B type operates in the opposite way of the Class-A type to improve system efficiency. So an output device only conducts for half the sinusoidal cycle to remove current flow in an output device such as the Class-A type. And the Class-AB type is a combination of the Class-A and B type, and is one of the most common types of existing power audio systems. The efficiency of Class-AB type is about 50%. In a Class-D type, a switching or PWM (Pulse Width Modulation) technology was applied because switches are either fully on or fully off and power losses are significantly reduced in the output devices. So efficiency of Class-D type is about 95% and other operational characteristics are sufficient to satisfy [1]. However, an unwanted noise may be generated in the signal processing and the noise has a negative impact on the sound quality. Thus adaptive noise cancellation algorithm using fast Walsh transform is proposed to solve these problems and improve sound quality.

2 Design of Class-D Type Audio System for Portable Application

2.1 Class-D Type Audio System

The Class-D type audio system was invented to overcome the lower efficiency of linear system. Really Class-D type audio system efficiency is very high, so it is increasingly being used in mobile phones, PDA and other small portable applications on behalf of Class-AB type. And Class-D type has advantages of design in the temperature problem, battery operating time and lightweight. For basic circuit operation, sinusoidal signal v_s is wave with a frequency ranging from 20Hz to 20KHz typically. This signal is compared with a high frequency triangle waveform v_t to create the PWM signal v_p . This PWM signal is then used to drive the power stage, creating the amplified digital signal, and finally a low pass filter is applied to the signal to filter out the PWM carrier frequency and retrieve the sinusoidal audio signal v_o . The sound source is amplified and reconstructed through the system. The effective gain k can be determined by applying a dc voltage at the input as shown in equation (1). If v_s is increased, the $\tilde{v}_p$ low-frequency time average of v_p increases linearly until it reaches the level V_{OP} , which corresponds to the positive clipping voltage at the output [2]. The basic Class-D type circuit and its voltage waveform are shown in figure 1.

$$k = \frac{\tilde{v}_p}{v_s} = \frac{V_{OP}}{V_{TP}} \quad (1)$$

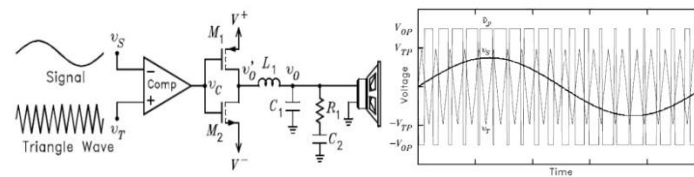


Fig 1. Basic Class-D type circuit and voltage waveform

2.2 Portable Class-D Type Audio System Design

In this paper, the 15W/stereo Class-D audio amplifier TPA3110D2 of TI(Texas Instruments) is used to design and implement a digital audio sound system[3].The TPA3110D2 can drive stereo speakers as low as 4Ω . The high efficiency of the TPA3110D2, 90%, eliminates the need for external heat sinks when playing music and the outputs are also fully protected against shorts to GND, VCC and output to output. Figure 2 shows functional block diagram and simple application of TPA3110D2.

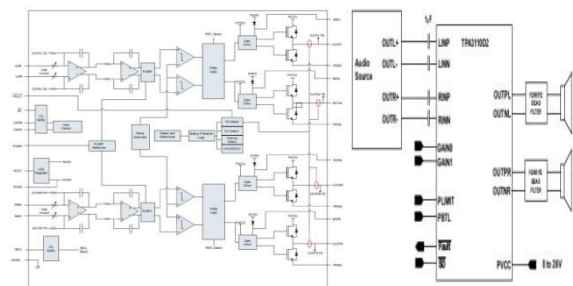


Fig 2. Functional block diagram and simple application of TPA3110D2

Bluetooth technology is applied to add wireless sound transmission and mobile features. Bluetooth is a wireless technology standard for exchanging data over short distances (using short-wavelength UHF radio waves in the ISM band from 2.4 to 2.485 GHz) from fixed and mobile devices, and building personal area networks (PANs) It was invented by telecom vendor Ericsson in 1994. Bluetooth is a standard wire-replacement communications protocol primarily designed for low-power consumption, with a short range based on low-cost transceiver microchips in each device[4]. For the design, Bluetooth v4.1 is used that is announced by the Bluetooth SIG in December 2013. In figure 3, block diagram of proposed portable Class-D type audio system is shown. And designed system shows 25dB signal to noise ratio, 8 ohm impedance, 70dB channel separation and 20~20KHz frequency response. Its frequency response result is shown in figure 4.

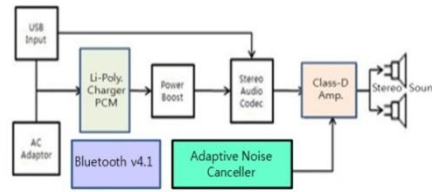


Fig 3. Block diagram of portable Class-D type audio system

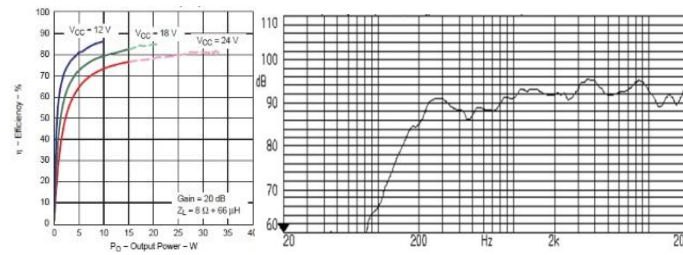


Fig 4. Efficiency and frequency response of designed system

For optimal utilization of Bluetooth v4.1, Bluetooth wireless data transmission experiment is carried out according to various environmental conditions. One of the results obtained by this experiment, recognition time between Bluetooth transmitter and receiver is changed from 77m sections by each baud rate change. In this case, data transmission is very stable and fast. But the results depend on a change of surrounding environments [5]. For example, wireless data transmission result is received bad influence by a microwave or noise.

3 Adaptive Noise Algorithm Using Fast Walsh Transform

Typically, wireless data transmission is influenced by various surrounding environments and this problem may affect the quality of wireless sound reproduction. Therefore adaptive noise cancellation algorithm using the fast Walsh transform is proposed for solving the problem in this paper. A function $f(t) = t$ can be transformed by the fast Walsh transform and its coefficients F can be obtained also. In this case, i th coefficient F_i is defined as follow [6][7]:

$$f(t) = \sum_{i=1}^N F_i W_N(t) \quad (2)$$

In figure 5, block diagram of adaptive noise cancellation system is shown. The $u(n)$ is signal source and $r(n)$ is noise source. $U(s)$ is transfer function of first acoustic path from signal source and $P(s)$ is transfer function of adaptive filter from output $p(n)$. And $e(n)$ is a residual error signal from error sensor and desired output signal $y(t)$, $\tilde{u}(n)$ is a gained source signal by $U(s)$ and $\tilde{p}(n)$ is gained and filtered signal by $P(s)$. Noise cancellation is a change in the optimal filtering includes

generating noise estimate by filtering the reference input and then subtracting this noise estimate from the primary input containing both signal and noise [8].

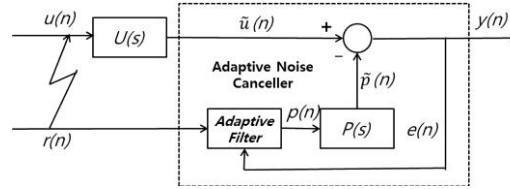


Fig 5. Block diagram of adaptive noise cancellation system

Now we can express the adaptive noise cancellation system using fast Walshtransform as:

$$\tilde{u}(n) = \sum_{i=0}^m \tilde{U}_i P_W W_N(t) \quad (3)$$

$$\tilde{p}(n) = \sum_{i=0}^m \tilde{P}_i P_W W_N(t) \quad (4)$$

$$e(n) = \sum_{i=0}^m [U_i - Y_i] P_W^T W_N(t) \quad (5)$$

where $W_N(t)$ is fastWalshfunction and P_W is operational matrix with $t = m$.

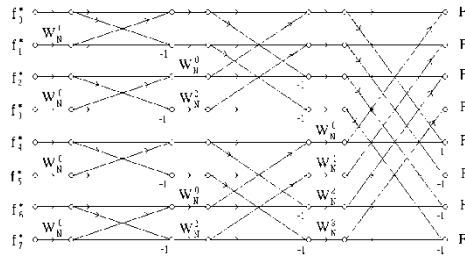


Fig 6. Signal processing graph for the adaptive noise cancellation with WT

Figure 6 shows signal processing follow graph for proposed adaptive noise cancellation algorithm. Inparticular,calculation and processingtimebycomputerismainlydependent on thenumber of times ofmultiplication, so the expanding term value of N is grater,using the adaptive noise algorithm by the fast Walsh transform would bemore effective [9].Then desired output $y(t)$ can be obtained from equation (5) and we can determine parameters for adaptive noise cancellation system [10].

$$\begin{aligned} Y^T(a_p I + a_{p-1} P_W + \dots + a_1 P_W^{p-1} + a_0 P_W^p) W_N(t) \\ \cong U^T(b_q I + b_{q-1} P_W + \dots + b_1 P_W^{q-1} + b_0 P_W^q) W_N(t) \end{aligned} \quad (6)$$

4Conclusion

In this paper, a design and implementation method of portable Class-D type audio system with adaptive noise cancellation algorithm based on fast Walsh transform is proposed. The proposed Class-D type amplifier showed satisfactory electrical performance characteristics, efficiency 90% on condition impedance 8 ohm and gain 20dB. And algorithm for adaptive noise cancellation system is simple and useful to build system hardware and firmware.

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