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A Technical Study on Sound Effects Processing and Expressive Enhancement in Contemporary Music

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SUMMARY: *In order to enhance the sound expression and artistic effect in modern music production and performance, this paper designs and implements a real-time digital sound processing system based on DSP, and completes the design of the core sound algorithm module for delay, reverb and equalization. The delay effector uses a digital delay to access the memory in order to obtain a precise delay. Excellent filters such as FIR filter with good filtering effect and IIR type filter with infinite impulse response are used to construct the overall reverb model. Considering the number of audio segments and their complexity, a 10-band graphic equalizer is used to segment the full band audio. The comprehensive sound processing effect of the system is realized on the matlab platform. The results show that the three designed effects perform well in the sound processing presentation, and the test indexes of the system and its chip meet the design requirements. The sound effectors control the noise level of the left and right channels at -98.6 dB and -100.1 dB, which provides a reliable data reference for the development of audio processing equipment.*

KEYWORDS: *DSP; digital sound processing; FIR filter; Gardner reverb model; matlab platform*

1 Introduction

Audio processing involves pre-processing sound signals before sending them to speakers or headphones for playback [1]. Audio technologies are categorized into compensatory and decorative effects. Decorative effects artificially create effects absent in the original audio or enhance existing ones, primarily including artificial reverb, pitch shifting, dynamic range compression, intelligent volume control, and stereo enhancement [2, 3]. Compensatory audio effects refer to signal processing techniques that compensate for deficiencies in electroacoustic devices or playback environments, aiming to achieve playback quality closer to the intended recording standards. Key technologies include speaker, headphone, and room equalization; virtual bass; and crosstalk cancellation [4-6]. Digitalization represents the prevailing trend in technological advancement, and the digitization of audio is an inevitable development [7]. With the continuous refinement of intelligent algorithms and advancements in artificial intelligence technology, digital effects processing for audio signals has experienced rapid development. The realization of intelligent processing in sound effect systems has become a significant research direction [8, 9].

Current research on intelligent algorithms in audio processing primarily focuses on music emotion recognition. By controlling lighting changes based on emotional elements, these

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approaches enhance musical expressiveness [10]. For instance, Reference [11] employs a hybrid approach combining support vector machines and artificial neural networks to intelligently recognize and classify human emotions in musical performances. This fusion algorithm achieved satisfactory results in terms of recognition recall and F1 scores. Reference [12] constructs a comprehensive model integrating audio feature extraction and emotion classification based on deep learning algorithms. While its emotional recognition performance varies across different musical styles, it achieves high accuracy and F1 scores for joy, sadness, and tranquility. Reference [13] optimizes the backpropagation neural network algorithm for music data analysis, improving both recognition accuracy and speed on public music datasets. Reference [14] designed an intelligent algorithm based on recurrent neural networks to extract musical melody features, accurately identifying the emotions conveyed within music databases. This algorithm enables music discovery using emotional tags. Reference [15] combined convolutional neural networks, recurrent neural networks, and feedforward neural networks in music emotion research. By training this hybrid model with emotionally tagged music fragments, it converts Mel-frequency spectrograms into specific emotions for recognition and classification. Reference [16] proposes a music classification method compatible with both emotion and intelligent algorithms. This approach uses musical emotional features as a key indicator, significantly reducing the computational complexity of music emotion classification while improving the accuracy of automatic music classification. Reference [17] employs an intelligent algorithm for identifying musical emotion. It analyzes acoustic, melodic, and audio features of music and combines them rationally. Test results show the algorithm's Kappa coefficient exceeds 0.75, demonstrating good accuracy in musical emotion recognition and classification. Reference [18] employs convolutional neural networks and long short-term memory models to integrate music with emotional image visualization. Simulation experiments reveal that the model achieves peak matching accuracy with musical emotion when the emotional classification loss function weight is set to 0.2.

In recent years, automatic music generation has remained a prominent research topic in AI-driven music, yielding substantial achievements [19]. Reference [20] developed an interactive evolutionary optimization algorithm-based automatic music arrangement method. This approach rapidly generates melodies matching target styles or fragments while achieving high satisfaction scores. Reference [21] designed an intelligent pop music arrangement method based on an adaptive multimodal particle swarm optimization algorithm. By solving the arrangement model through optimization, the proposed method achieved scores above 90 in rhythm, completeness, harmony, and overall effect, enhancing the expressiveness of musical works. Reference [22] integrated evolutionary techniques and genetic algorithms to perform controlled selection of musical rhythmic features while replacing new harmonic features with subsequent ones. This approach generated high-quality new compositions that passed artificial intelligence evaluation tests. Reference [23] established a Hidden Markov Model (HMM) for chord recognition in music, subsequently applying a multi-style chord-based music generation network to produce chordal compositions. The HMM achieved an 81.8% chord recognition accuracy and received high evaluations for generating music across diverse styles.

Music audio signal processing technology must be grounded in a deep and comprehensive understanding of music itself. Despite significant advances in AI for audio signal processing, substantial challenges remain [24, 25]. The open-source audio processing and music information retrieval library madmom, developed in [26], incorporates multiple intelligent algorithms including Hidden Markov Models. It supports beat detection, accent and beat marking, rhythm estimation, and chord recognition, providing reliable technical support for sound effect processing. Reference [27] utilizes AI algorithms to construct an intelligent speech system integrating audio data decompression algorithms and rapid reconstruction techniques. This system achieves rapid audio compression while elevating subjective sound quality scores from 87 to 92 points. Reference [28] improved traditional spectral subtraction methods and validated them through simulation experiments. The enhanced method achieved higher signal-to-noise ratios in de-reverberated music signals and superior average opinion scores compared to alternative approaches. Similarly, Reference [29] compared the noise reduction effects of Wiener filtering, wavelet transform, spectral subtraction, and improved spectral subtraction in electronic music production. The improved spectral subtraction achieved a signal-to-noise ratio of 20.36 dB and a signal distortion ratio of 7.94 dB, making it applicable to practical audio processing. In summary, these studies demonstrate that precise identification, classification, and enhancement of audio signals are becoming feasible, poised to revolutionize fields such as speech recognition and music composition [30]. Through automated music arrangement, melody generation, and audio processing, the perceptual and expressive capabilities of modern music are being significantly enhanced.

Based on the digital signal processor, this study carried out the algorithm design and specific implementation of three types of core sound effects, namely delay, reverb and equalization. The overall architecture of the system is first designed, and the ARM+DSP dual-core architecture is introduced as the hardware substrate of the system. Then the delayer is used to realize the delay processing of the input digital signal, and several kinds of filters covering the FIR filter are proposed to construct the overall reverb model. A 10-band graphic equalizer is used to gain different frequency components in the audio, and the integration of the system effect algorithms is completed on the matlab platform, and finally the hardware and software design of the system is improved. Simulation experiments are carried out on matlab software to verify the practical application of the effect algorithms in this paper, and the system and its chip are tested in several index dimensions.

2 DSP-based digital sound processing system design

2.1 Overall system design

DSP-based digital sound processing system, the entire hardware system selection of ARM + DSP dual-core architecture, the hardware system is a digital audio system system control core, the selection of TI's TMS320C6713 type of digital signal processor, to complete the system algorithms and related computational processing, the audio processor's main system mainly consists of: the microcomputer processor, the keyboard, the LCD display and reset circuit. The use of 32-bit high-performance low-power chip as the control chip of the system, the system CPU using three-stage pipeline technology, can simultaneously complete the finger, decoding, finger execution, the inheritance of these functions to enable it to perform the control function of the human-machine interface module.

From the aspect of the whole sound algorithm of the system hardware carrier, it is a 32-bit tellurian floating-point DSP with independent units and floating-point fixed points, in addition this system hardware also has a RISC-like instruction set, which has a better network performance.

There is a 32-bit external storage interface within this system, and each excuse is interconnected to form a network. This interface allows seamless interfacing of the system's synchronous and asynchronous storage. There are also two multi-channel buffered serial ports, each capable of full duplex communication with multiple channels for sending and receiving. In addition, this system has more direct memories that enable timer management for multiple channel controllers, and can also be utilized for efficient audio transmission using its powersaving mode principle, which is a real-time audio transmission over a transport protocol, thus achieving high-definition audio.

2.2 Delay, Reverb and Equalization Processing

There are many peripheral devices in the audio system, the previous sound effectors are implemented using analog technology, with the development of digital signal processing technology, modern music can be implemented in the digital domain through digital signal algorithms.

2.2.1 Delay effects

Delayer, is a device that delays the sound signal for a certain period of time by some means and then outputs it. Using it can solve the problem of sound and image synchronization, and at the same time it can also be used to improve the clarity of the sound reinforcement.

The digital delay method can be used to obtain a precise delay by accessing the memory. Its implementation process is as follows: firstly, calculate the amount of delay: suppose the time to be delayed is t, and the sampling frequency of the signal is fs, then the number of delayed sampling points is $N = t.fs$. Then, open a buffer of length $N+1$, initialize the buffer to 0; finally, define two pointers, one for the input pointer, and the other one for the output pointer, so that the output pointer is pointing at the last memory cell of the buffer, and the input pointer is pointing at the first memory cell of the buffer. When inputting a sample point, write it to the storage cell pointed by the current input pointer and add 1 to the input pointer to make it point to the next storage cell; at the same time, read the content of the storage cell pointed by the current output pointer and use it as the output, and add 1 to the output pointer. whenever the input pointer and the output pointer is added 1, you need to judge whether the pointer is beyond the buffer boundary. If so, it will point to the first memory cell. This realizes the delayed processing of the input digital signal.

2.2.2 Reverb Effects

A reverb effector is a device that adds a reverb effect to the original audio signal by manual means. Digital reverb can be divided into sampling reverb and algorithmic reverb two kinds. Sampling reverb is to sample the reverb pulse sequence of a certain space and store it, and then use it with the original audio signal to be processed for the convolution operation, the output is the audio signal with the characteristics of the space reverb.

Algorithmic reverberation is based on the generation mechanism of reverberation, through digital signal processing algorithms to construct a certain space reverberation pulse sequence, as well as the reverberation time, the density of reflected sound and other parameters to simulate the reverberation effect in the real environment. Algorithmic reverb is widely used in current reverb effectors. Some of the world's top reverb effector manufacturers have very good reverb algorithms, but they are generally not open to the public, the following will discuss some classic algorithmic reverb models.

(1) FIR reverb modeling is easily thought of by the reverberation impulse response characteristic diagram of the room, as the reflected sound is a delay to the direct sound, Figure 1 shows a single delay unit of the FIR filter [31], which can be used to obtain a single reflected sound. Naturally, multiple reflections can be obtained using multiple delay units. Instead of using this method to obtain the full reverberant sound, a finite order FIR filter is generally used to obtain the pre-reflected sound.

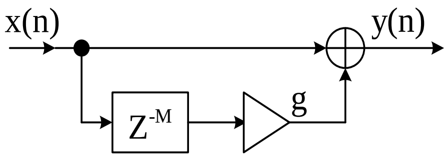


Figure 1: Primary reflection sound model

(2) The reverberation model of the comb filter is a type IIR filter, and its unit impulse response is infinite.

(3) The all-pass filter reverberation model has an amplitude-frequency response of $|H(e^{j\omega})|=1$ and the same gain at all frequencies, which avoids acoustic coloration, but the density of its impulse response is too low for satisfactory reverberation. If multiple all-pass filters are cascaded can improve its impulse response density, but there is still a gap relative to the real situation.

(4) Nested all-pass filter model A nested all-pass filter model is formed by replacing the delay unit in the all-pass filter with another all-pass filter.

The z-transform of this system is:

$$H(z)=\frac{Y}{(z)}X(z)=\frac{G(z)-g}{1-gG(z)} \tag{1}$$

Its range is:

$$|H(z)|=\sqrt{\frac{|G(z)|^2-g(G(z)+G^*(z))+g^2}{1-g(G(z)+G^*(z))+g^2|G(z)|^2}} \tag{2}$$

Therefore, if $|G(z)|$ is 1, then $|H(z)|$ is also 1, i.e., it is all-pass. Since the impulse response of the inner all-pass filter G(z) can be fed back to the input by the feedback branch of the outer layer and used as input again, the overall impulse response density is greatly increased. A more natural reverberation effect can be obtained by using this model.

(5) The Schroeder reverberation model takes the form of a comb filter bank cascaded with an all-pass filter bank. The parameters in this model include the delay time and attenuation coefficient of the four comb filters and the delay time and feedback gain of the two all-pass filters.

(6) The Moorer reverberation model is an improvement of the Schroeder model. Due to the consideration of the air and other high-frequency acoustic signal absorption is greater than the low-frequency signal, then the reverberation time of the high-frequency signal is shorter than that of the low-frequency signal, so it adopts a low-pass comb filter instead of the general comb filter in Schroeder's model, and at the same time, it also increases the number of comb filters.

(7) Gardner reverberation model Gardner proposed three reverberation models on the basis of previous research, namely, small room reverberation model, medium room reverberation model and large room reverberation model. The model according to the different room sizes were used in different cut-off frequency of the low-pass filter to simulate the attenuation characteristics of the air on the high-frequency signal. The model can obtain a more realistic post-reverberation effect.

According to the above analysis, the overall reverberation model used in this paper is shown in Figure 2.

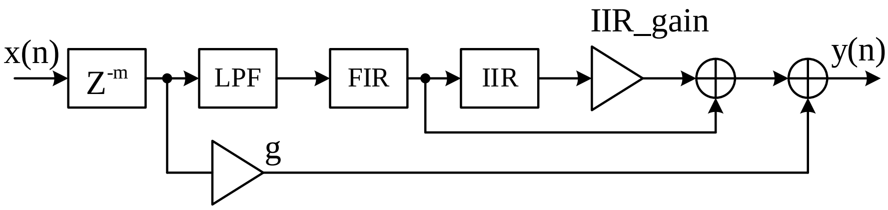


Figure 2: Overall reverberation model

2.2.3 Equalizer

An equalizer (EQ) is a device that can independently adjust the gain of different frequency components of an audio signal. For speakers with uneven frequency response curves, an equalizer can be used to compensate for the frequencies in order to reproduce the sound realistically.

There are many types of equalizers, among which the graphic equalizer is a widely used equalizer, and this section will mainly discuss the graphic equalizer.

Graphic equalizer [32] is through a set of center frequency according to the regular distribution of the band-pass filter will be divided into a number of different frequency bands of full-band audio signals, in the center frequency of each band were placed in the center of a push-pull key to independently regulate the gain of each band, can be directly based on the distribution of the position of the push-pull key to know the shape of the equalization curve; regardless of the band to enhance or attenuate the band, band-pass filter's center frequency and its bandwidth always remain the same. Whether a band is boosted or attenuated, the center frequency of the bandpass filter and its bandwidth always remain the same, i.e., its quality factor is fixed. The graphic equalizer used in the audio field divides the full frequency band into different numbers of frequency bands according to different application requirements; the more bands are divided, the more detailed the frequency division is, the finer the adjustment will be, but the more difficult it is to adjust, and in this paper, we use a 10-band graphic equalizer.

2.2.4 Integrated sound processing GUI design

The design of delay effector, reverb effector and equalizer effector are discussed above, and the following comprehensive sound processing GUI is designed in matlab, and the test of various sound processing effects is carried out on this platform.The Matlab GUI design includes the front panel interface design and the writing of M files, which consists of different types of controls, each of which has its own properties. Each control has its own attributes and its parameters can be set by the user.

2.3 DSP Implementation of Audio Processing Algorithms

2.3.1 Basic DSP hardware design

A complete DSP system is usually composed of a DSP chip and other corresponding peripheral devices.

In the DSP application system design, a perfect reset design will greatly improve the stability and reliability of system operation. When the power supply is just added, the DSP chip is in the reset state, and the chip is reset when RS is low. In order to make the chip initialization is correct, generally should ensure that RS is low for at least 3 CLKOUT cycles. For the actual DSP application system, its reliability is a problem that can not be ignored. Due to the high clock frequency of the DSP system, it is very likely to occur during the operation of the phenomenon of interference and interference, when the system may be dead, in order to overcome this situation, in addition to the software to make some protection measures, the hardware must also make the corresponding processing. The most effective hardware protection measures is the use of automatic reset circuit with monitoring functions. Automatic reset circuit in addition to the power-on reset function, but also has the ability to monitor the system operation and reset again in the event of system failure or crash. The basic principle is to provide a monitoring line for monitoring system operation through the circuit, when the system is running normally, should be in the specified time to monitor the line to provide a high and low level changes in the signal, if the signal does not change in the specified time.

The automatic reset chip selected for this system is MAX706R from MAXIM:

In design, the clock is often not sufficiently emphasized. In fact, the clock is a very important part of the circuit design. There are generally two ways to provide a clock to a DSP chip. One is to use the clock signal from an external clock source. The external clock signal is added directly to the X2/CLKIN pin of the DSP chip, while the X1 pin is left dangling. The external clock source can be a crystal oscillator with a stable frequency.

Another method is to utilize the internal oscillator of the DSP chip to form a clock circuit. A crystal is connected between the X1 and X2/CLKIN pins of the chip to start the internal oscillator.

In this system, the internal oscillator is used, and the crystal used is 20 MHz. through an external phase-locked loop control circuit, the appropriate multiplier is selected to provide the system clock inside the CPU.

2.3.2 DSP software development process

The software of this system is mainly composed of two parts. The first part belongs to the VC5402 side, mainly including initialization CPU; CY7C1041, FLASH in-system programming; McBSP to receive and send data; DSP side of the communication program with the master control program; the second part belongs to the PC side, mainly composed of communication program part.

A typical room impulse response, the first received is a direct signal, and then a series of sparse impulses through the walls and other objects of a single or a few times the reflection, with the extension of time by a number of reflections formed by the rapid increase in the density of the impulse sequence, the amplitude of the amplitude decreases and superimposed on each other or canceled out, so that a single impulse indistinguishable from the amplitude of the generation of an exponential decay with time.

The design of the reverberation effect consists of two parts: the design of the comb filter and the all-pass filter.

From the principle and realization model of the reverb algorithm, it can be known that the reverb is realized by a large number of all-pass filter modules and delay line modules, in which the delay line also exists in the all-pass filter, so the delay line is the most basic unit in the reverb algorithm. The delay line can be realized through the circular buffer of the ring, that is, according to the actual needs of the system, set a suitable size of the storage unit to realize the delay. For example, when the delay time t is 1ms and the sampling rate f_s of the system is 44.KHz, the number of storage units N required by the system is:

$$N = t \times f_s = 1ms^* 44.1KHz \approx 44 \tag{3}$$

For each circular buffer, the read/write operation includes the following steps: read the data of the storage unit pointed to by the current address; save the latest data to the storage unit pointed to by the current address; add 1 to the address of the current storage unit, determine whether the address after adding 1 reaches the end address of the buffer, and if it does, take the first address of the circular buffer as the current address. In order to ensure the modularity of the whole algorithm software and facilitate the debugging and upgrading of the software, a data structure is designed for each delay line in the software implementation as follows:

The delay line is realized by means of circular buffer's. According to the required delay time and the signal sampling rate of the system, a continuous storage space is opened in the off-chip extended RAM to save the current data for a period of time, and then the data is operated on. The mathematical relationship between the delay time t, the sampling frequency f_s of the system and the number N of storage units required is shown in the following equation:

$$N = t \times f_s \tag{4}$$

According to the model of the reverberation algorithm, the number of storage units N required for the delay time t can be calculated so that a continuous section of storage units of length N is opened in the off-chip RAM, and when the system is initialized, the first address of this section of storage units is set to be the current address, the start address of the saved storage units, and the end address of the lower 16 bits, and the initial current content of the storage units is zero.

Digital equalizer, that is, to adjust the frequency response of the frequency bands, so as to produce different effects: such as heavy bass, radio, treble and so on. Can compensate for the lack of frequency components in a variety of program signals, but also can suppress the overweight frequency components. For audio equalization shelving filter, by amplification or attenuation, bandwidth, center frequency of the three parameters is the most intuitive indicators. It is convenient to derive the transfer function of the filter from these three parameters. The implementation method is time domain filtering and frequency domain filtering. Frequency domain filter utilizes the FFT algorithm, the effect is quite good; time domain filter processing fast but less effective. Frequency domain filtering specific operation: first of all, the time domain to the frequency domain of the transformation, and then in the frequency domain wave, and then transformed from the frequency domain back to the time domain. Time-domain filtering does not need to transform the time domain and frequency domain, and can be filtered directly in the time domain. The equalizer used in this thesis uses the method of frequency domain filtering [33].

3 Delay, reverb and equalization algorithm simulation

3.1 Simulation of delay algorithm

Simulation code on the matlab software not only can intuitively get the algorithm to process the sound waveforms, but also be able to use headphones to hear the algorithm to process the sound, so that for a variety of effects have a deeper understanding of the experience of writing programs to achieve the effect of the delay on the matlab software, Figure 3, Figure 4, respectively, is the input sound signal waveforms and the output waveform after the delayed effect of the processing, the delay of the waveform is delayed backward by about $4.5*10^4$ data points. The delayed output signal waveform obviously accomplishes the delay effect.

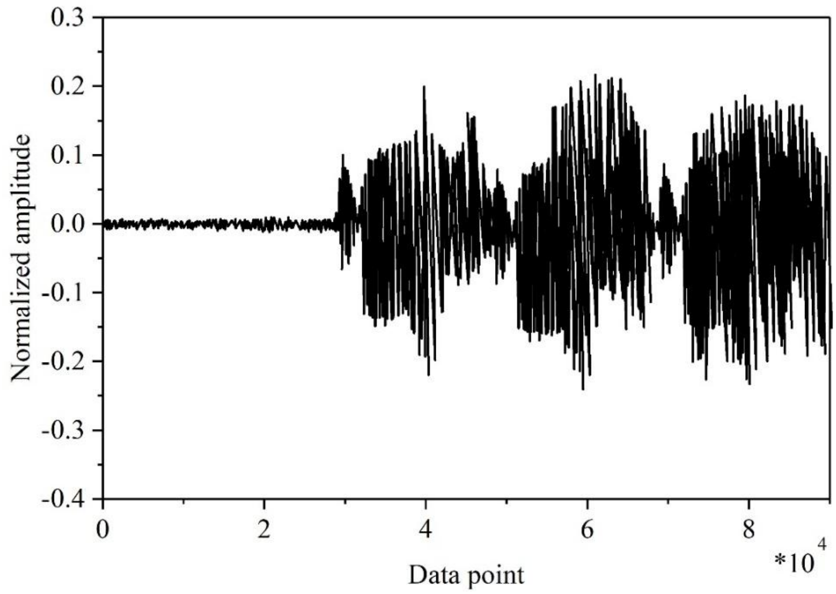


Figure 3: Enter the waveform of the sound signal

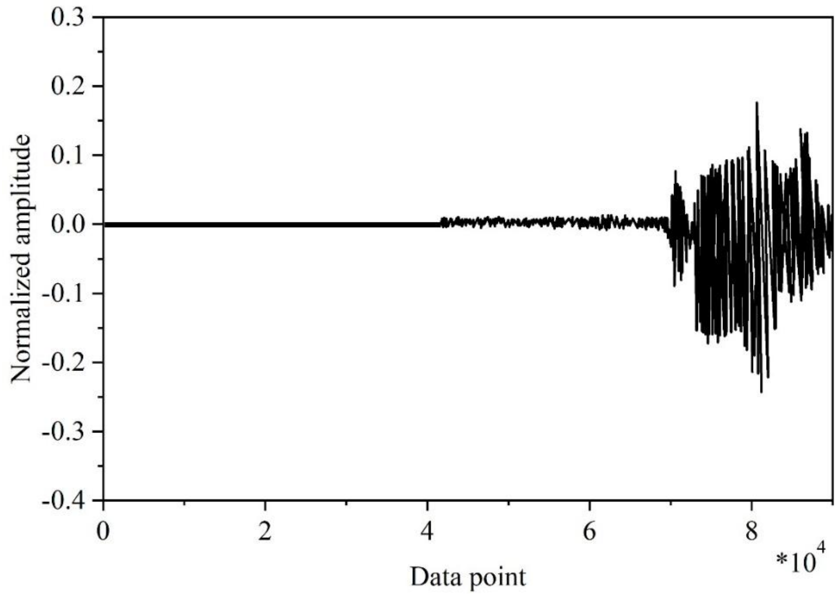


Figure 4: The output waveform of the delayed effect

3.2 Reverb Algorithm Simulation

The following shows the simulated reverb waveforms of the overall reverb model on matlab software. Fig. 5 and Fig. 6 show the input sound waveform and output sound waveform of the reverb effect, respectively. It is confirmed by actual listening that the overall reverb effect realized by the model in this paper is good, and the range of waveform fluctuation is expanded from -0.27~0.64 to -0.8~0.7.

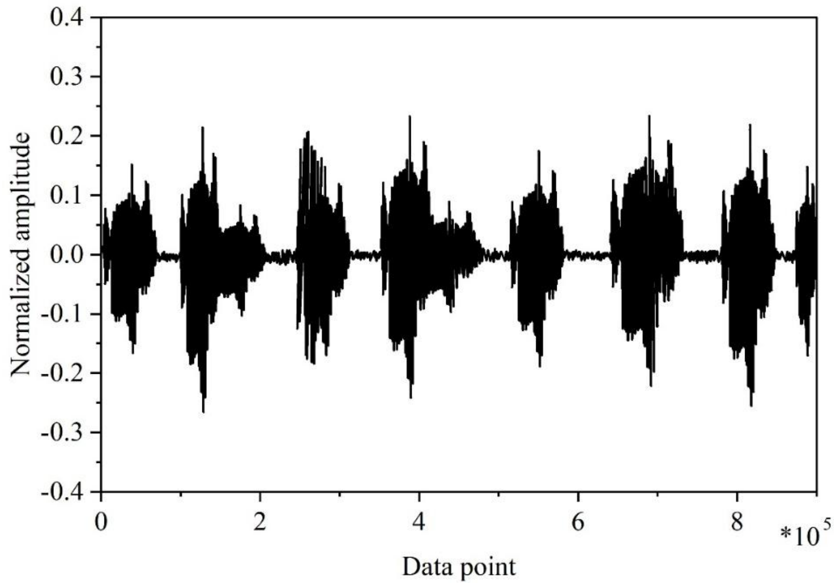


Figure 5: The input sound waveform of the reverberation effect

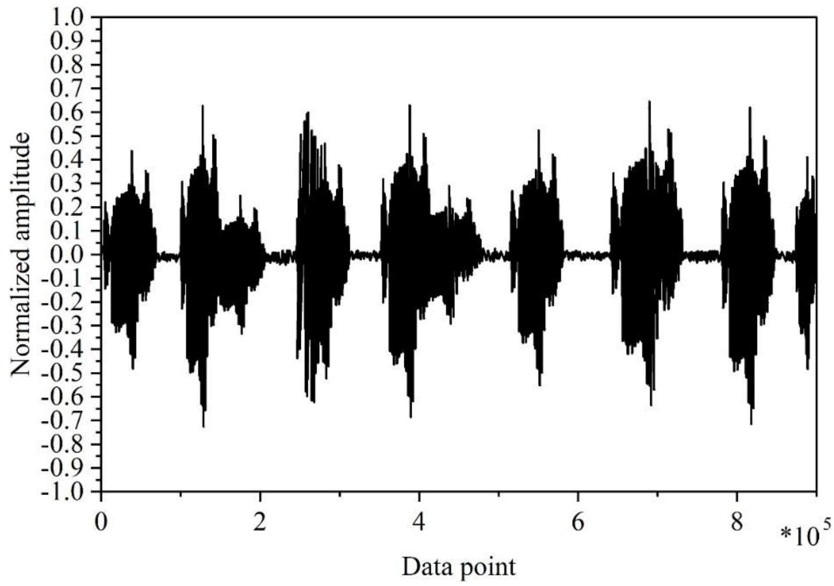


Figure 6: The output sound waveform of the reverberation effect

3.3 Equalization Algorithm Simulation

The equalization algorithm was tested in a music player equipped with Android 5.1, and after completing the corresponding development work, the player was used to play 32 bit audio data, and the equalizer was turned on to adjust the equalizer gain. The quantization bit depth of the audio data can be viewed and confirmed by Adobe Audition. Figure 7 shows the input audio information used for testing in Adobe Audition. From the log output, it can be seen that the system uses native equalizer processing when playing music. Secondly, it is clear from the frequency analysis graph and the frequency analysis data that the equalizer settings have clearly produced the appropriate effect, indicating that the native equalizer is capable of processing 32 bit audio data.

Obviously, it can be seen from Figure 7 that the frequency analysis graph before setting the equalizer gain and the frequency analysis graph after setting the equalizer gain have been different, due to the whole section of the frequency analysis graph is larger, this paper only intercepts part of it. Table 1 for the processing before and after part of the frequency analysis data, from the frequency analysis data in Table 1 can also be seen, the audio data has been according to the user's operation to occur corresponding changes, indicating that the system has been able to make the corresponding action of the user's processing requirements. From the subjective listening experience, when the input audio is 32 bit, turn on the audio equalizer function, set the gain of each band, the system can react to the user's operation in real time, whether the band gain is set individually for a certain band or set multiple bands at the same time, the system can respond correctly, and the positive and negative gain settings as well as the listening experience of the different band settings is the same as that of the native equalizer, which proves that this paper's scheme is feasible. This proves the feasibility of the program in this paper.

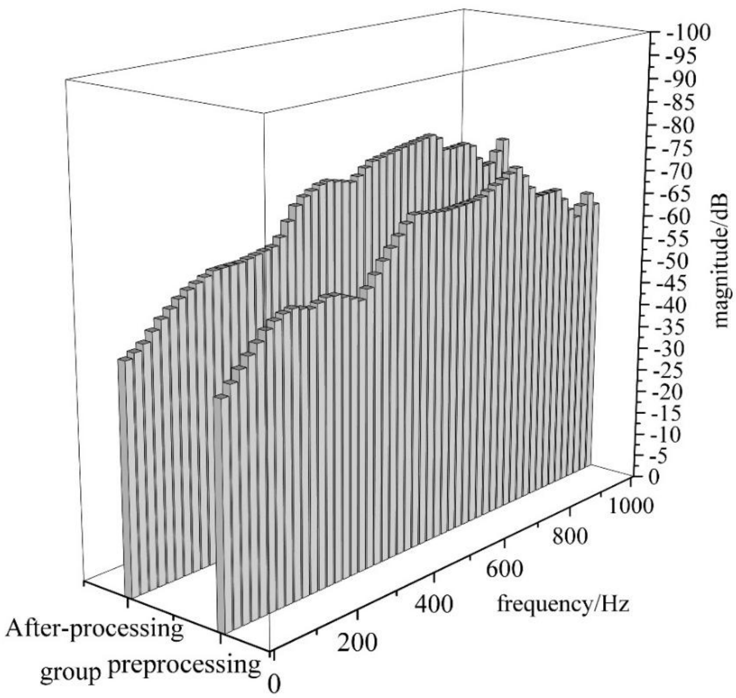


Figure 7: Processing and frequency analysis

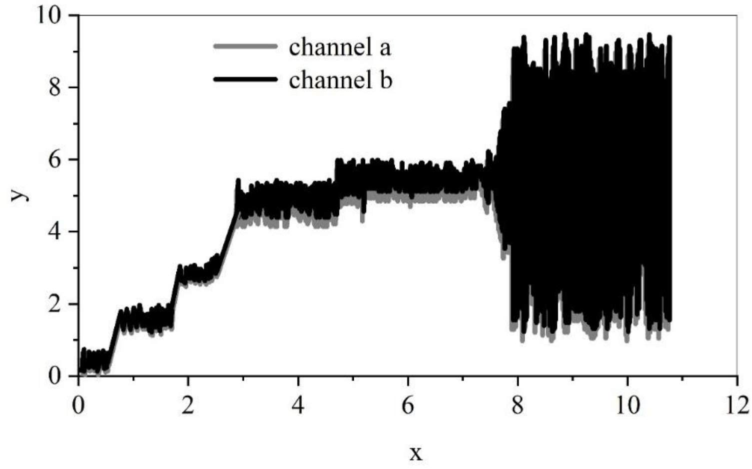
Table 1: The analysis of the frequency analysis of the posterior

Frequency (hz)	Before processing(dB)	After processing(dB)
86.07	-57.38	-57.18
129.16	-61.66	-62.69
344.63	-68.98	-72.58
645.95	-71.54	-80.25
990.56	-59.15	-68.36
2024.16	-60.05	-69.84
4005.05	-59.56	-69.28
6287.88	-52.13	-58.97
9991.4	-83.3	-89.15
16020.64	-102.57	-97.88

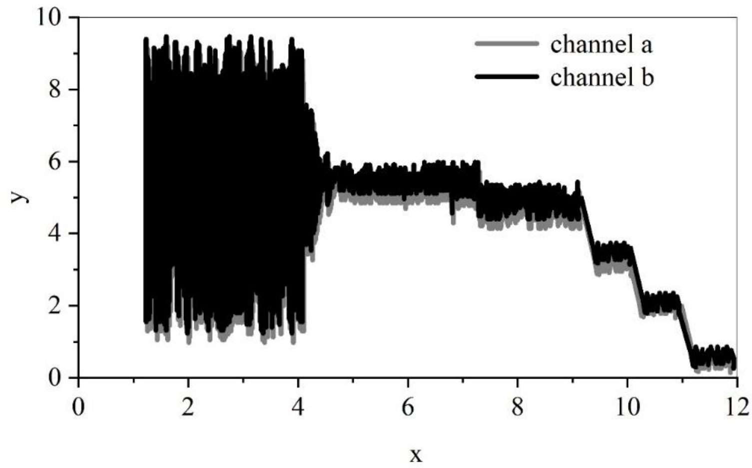
4 System and chip performance testing

4.1 Chip Performance Test

Fig. 8 shows the results of the chip power-up and power-down tests, Fig. (a) shows the power-up waveform of the chip, and Fig. (b) shows the power-down waveform of the chip. In the figure, channel a and channel b of the oscilloscope are connected to the two ends of the chip differential output, and channel M is the final output. From Fig. 8, it can be seen that when the chip is powered up, the output waveform increases slowly from 0 to the normal value. When powering down, the output drops to 0. Therefore, the power-up and power-down of the chip satisfy the design requirements.



(a)The radio form of the chip



(b)The lower wave of the chip

Figure 8: Electrical and lower electrical test results on the chip

4.2 System testing

4.2.1 Objective testing of sound quality

Frequency response refers to the reduction characteristics of audio equipment for waveform signals of different frequency components. Ideally, audio equipment should be able to treat any frequency components equally, that is to say, regardless of the frequency of the input signal, the output volume should be equal, and its frequency response curve should be a horizontal straight line.

In this test, we can see that the sound effects perform quite well and professionally. From the low frequency range of 20Hz all the way to the high frequency range of 20kHz, the line transition is very smooth, the test result is almost a straight line, and there is no obvious sign of signal attenuation. System frequency response test results as shown in Figure 9, the frequency response of the sound effect device in the 20Hz and 20K Hz frequency response to meet the design requirements of the indicators.

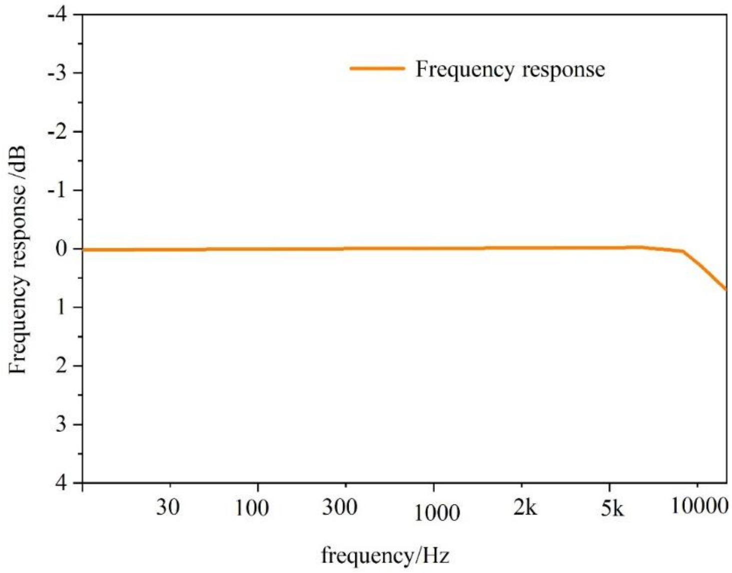


Figure 9: Test results of system frequency response

The test of noise floor and sound effect device hardware environment has a greater relationship, the main test of the sound effect device wiring structure, anti-interference ability, analog line design, as well as front and rear stage isolation in this regard have a direct relationship. If the noise floor of the sound effect device is too large, even in the absence of audio signal output, the speaker will appear annoying communication sound. Figure 10 shows the results of the noise level test, from the test results in Figure 10 we can see that the noise level of the sound effect device control is quite excellent, through the phase of the left and right channels, volume and other calculations to get the results of the left and right channels were -98.6 dB and -100.1 dB, to meet the design requirements of the indicators, thanks to the sound effect device itself is very high quality of DACs, ADCs, and the board alignment design and so on. This is due to the high quality of the DAC and ADC, as well as the board alignment design.

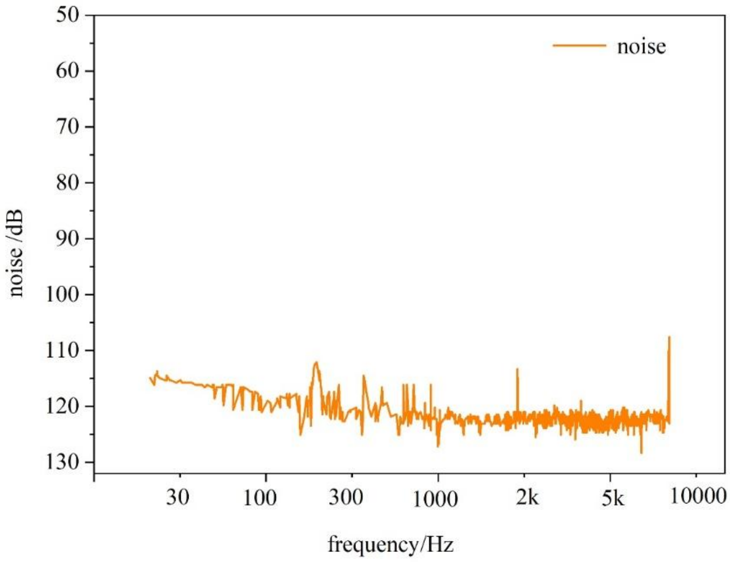


Figure 10: Test of background noise

The dynamic range test is a measure of the variation in volume of a sound effector, that is, a comparison of the strongest volume the sound card can produce with the weakest volume. If the dynamic range of the sound effector is good, it will perform well with sounds that vary greatly in range (thunder, percussion, etc.). The results of the dynamic range test are shown in Figure 11, which indicates that the final score of the sound effector is 102.8dB, meeting the design specifications.

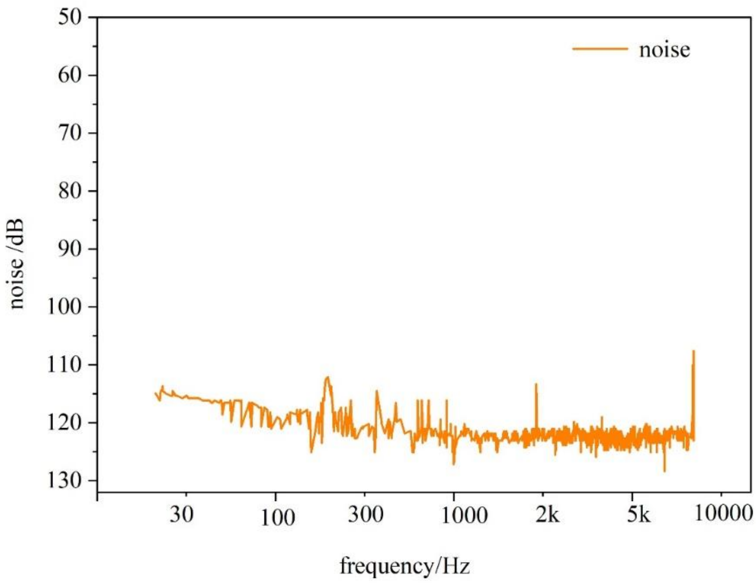


Figure 11: Dynamic range testing

This is the Total Harmonic Distortion plus Noise (usually written THD+N) test. THD+N is the sum of the distorted harmonic frequencies generated by the device itself, which characterizes the degree of coincidence between the input signal and the output signal, and the lower the value, the smaller the distortion. For the THD+N parameter, a THD+N of more than 0.1% can be detected by the human ear. The results of this test are shown in Figure 12. In this test, the total harmonic distortion of the sound effector is 0.0019%, which meets the performance specification. The odd harmonic content of the sound effector is not very pronounced over the entire frequency band, and there are basically no unwanted sharp pulses near 2kHz, 5kHz, etc. The performance is relatively flat.

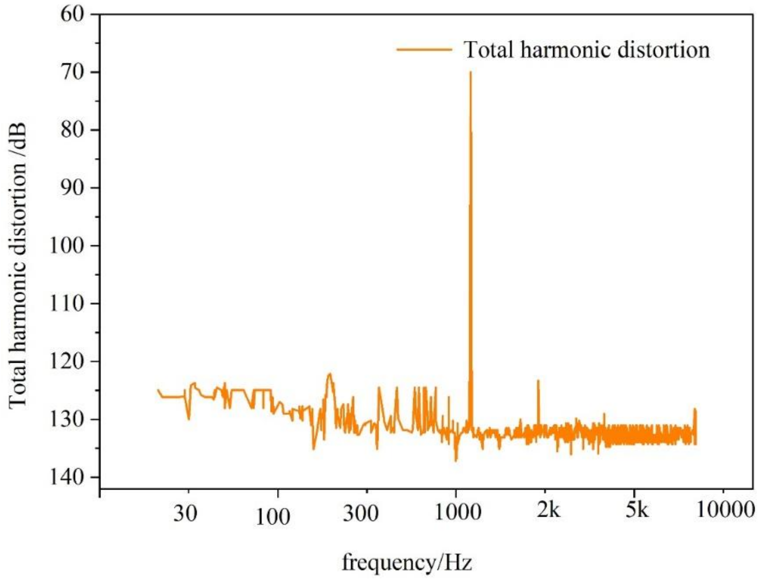


Figure 12: Test results of total harmonic distortion

4.2.2 Subjective testing of sound quality

In order to make the test more accurate, we in the whole testing process, the sound system first does not access the digital sound processing system, listen to once, and then the digital sound processing system access, listen to the second time. Through such a comparison, we found that the sound effect of digital sound processing system is obvious, in the corresponding story link, the effect of the sound effect can only be described by the word "shock", the sound of the whistling bullets across the sound, the cacophonous shouting, and the characters of the rapid breathing sound are all fully manifested, every subtle change has not been spared, as if the grand battle of the war. Changes have not let go, as if the magnificent war scenes are really reproduced in front of the eyes, the overall feeling is very real, thick, the effect of the scene is very shocking and compelling. Finally, after a careful audition, the sound engineer expressed his affirmation of the expressive power of these sound effects, believing that the effects realized by the effector were close to the performance of similar excellent products at the top of the world.

5 Conclusion

This paper carries out a research on the design and realization of a DSP-based digital sound effect processing system.

(1) Through simulation experiments, the usability of various sound processing modules of this system is confirmed, and the delay effect based on the digital delay method, the reverb effect based on the overall reverb model, and the music equalization effect based on the 10-band graphic equalizer are realized. The above modules enhance the practicality and interactivity of the system, and can meet the requirements of sound processing in actual performance and recording environments.

(2) In the system test and chip performance test, not only the chip power-up and power-

down tests meet the design requirements, but also the frequency response, noise level, dynamic range and total harmonic distortion of the system meet the system design requirements. The objective and subjective test results of the system perform well, and can effectively improve the spatial sense and expressiveness of the audio signal. The design of the DSP-based digital sound processing system is successful, and it can provide more excellent sound processing for music, and it provides feasible hardware and software design solutions for the practical application of sound processing technology.

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