Retraction

Retracted: Tone Recognition Database of Electronic Pipe Organ Based on Artificial Intelligence

Mathematical Problems in Engineering

This article has been retracted by Hindawi following an investigation undertaken by the publisher [1]. This investigation has uncovered evidence of one or more of the following indicators of systematic manipulation of the publication process:

(1) Discrepancies in scope
(2) Discrepancies in the description of the research reported
(3) Discrepancies between the availability of data and the research described
(4) Inappropriate citations
(5) Incoherent, meaningless and/or irrelevant content included in the article
(6) Peer-review manipulation

The presence of these indicators undermines our confidence in the integrity of the article's content and we cannot, therefore, vouch for its reliability. Please note that this notice is intended solely to alert readers that the content of this article is unreliable. We have not investigated whether authors were aware of or involved in the systematic manipulation of the publication process.

Wiley and Hindawi regrets that the usual quality checks did not identify these issues before publication and have since put additional measures in place to safeguard research integrity.

We wish to credit our own Research Integrity and Research Publishing teams and anonymous and named external researchers and research integrity experts for contributing to this investigation.

The corresponding author, as the representative of all authors, has been given the opportunity to register their agreement or disagreement to this retraction. We have kept a record of any response received.

References

[1] S. Zhao, "Tone Recognition Database of Electronic Pipe Organ Based on Artificial Intelligence," *Mathematical Problems in Engineering*, vol. 2021, Article ID 5526517, 12 pages, 2021.

Research Article

Tone Recognition Database of Electronic Pipe Organ Based on Artificial Intelligence

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In the past few decades, artificial intelligence technology has experienced rapid development, and its application in modern industrial systems has grown rapidly. This research mainly discusses the construction of a database of electronic pipe organ tone recognition based on artificial intelligence. The timbre synthesis module realizes the timbre synthesis of the electronic pipe organ according to the current timbre parameters. The audio time domain information (that is, the audio data obtained by file analysis) is framed and windowed, and fast Fourier transform (FFT) is performed on each frame to obtain the frequency domain information of each frame. The harmonic peak method based on improved confidence is used to identify the pitch, obtain the fundamental tone of the tone, and calculate its multiplier. Based on the timbre parameters obtained in the timbre parameter editing interface, calculate the frequency domain information of the synthesized timbre of each frame, and then perform the inverse Fourier transform to obtain the time domain waveform of each frame; connect the time domain waveforms of different frames by the cross-average method to obtain the time-domain waveform of the synthesized tone (that is, the audio data of the synthesized tone). After collecting the sound of the electronic pipe organ, the audio needs to be denoised, and the imported audio file needs to be parsed to obtain the audio data information. Then, the audio data are frequency-converted and the timbre characteristic information is analyzed; the timbre parameters are obtained through the human-computer interaction interface based on artificial intelligence, and the timbre of the electronic pipe organ is generated. If the timbre effect is not satisfactory, you can re-edit the timbre parameters through the human-computer interaction interface to generate timbre. During the experiment, the overall recognition rate of 3762 notes and 286 beats was 88.6%. The model designed in this study can flexibly generate electronic pipe organ sound libraries of different qualities to meet the requirements of sound authenticity.

1. Introduction

With the rapid popularization of the Internet in today's society, the accelerated merger of computers, communications and consumer electronics, the development trend of electronic product specialization and miniaturization [1], with the technological development of software and hardware equipment, the capabilities and superiority of artificial intelligence have quickly penetrated into life. Of course, various fields in the music technology field have become the hottest words in the field of music technology.

We know that the promotion of music is actually based on technological changes. Without the invention of tools, there would be no musical instruments; without the emergence of electronic technology, there would be no electronic music. Therefore, how much change the major technology of artificial intelligence can bring to music is also one of the motivations for me to write this article. The generation of artificial intelligence arrangement is an interdisciplinary field. It requires researchers to master a lot of interdisciplinary knowledge, including music production, music technology, artificial intelligence, and automatic accompaniment. Because it is an emerging field, domestic research is still relatively poor.

Artificial intelligence (AI) has brought the frontier of development to the fields of power electronics and power engineering. Bose believes that these technologies provide powerful tools for modern smart grid (SG) control. His applications include the automation design of modern wind power generation systems and their operating status monitoring under operating conditions, the failure mode recognition of SG subsystems, and the SG control based on real-time simulators. Although the concept of these application examples he

proposed can be extended to develop many other applications, there is still a lack of specific data [2]. Raedt et al. studied uncertainty in probability theory and graphical models and studied relations logically. He focused on two detailed representations: Markov logic network (an expansion of the relationship between undirected graphical models and weighted first-order predicate calculus formulas) and Problog (probability expansion of logic programs), which can also be regarded as Turing complete expansion of the relational Bayesian network. His research discusses Markov logic networks but still lacks theory [3, 4]. Din et al. believes that due to the existence of various pollutants produced by human, agricultural, and industrial activities, the quality of surface water has decreased. Traditionally, the concentration of SWQP is measured through intensive field work. His research attempts to develop an artificial intelligence modeling method to plot the concentration of optical and nonoptical SWQP. Although his research can retrieve different SWQP concentrations from Landsat8 images, it still lacks logic [5]. Hashemi et al. believes that process-based models have been widely used for storm surge prediction. He used the high-resolution wave and surge modeling system on the East Coast of the United States to numerically simulate 1,050 tropical tropical storms. His research provides an unprecedented data set. Although his research can be used to train artificial intelligence models for surge prediction in those areas, the accuracy is not high [6].

The timbre synthesis module realizes the timbre synthesis of the electronic pipe organ according to the current timbre parameters. The audio time domain information (that is, the audio data obtained by file analysis) is framed and windowed, and fast Fourier transform (FFT) is performed on each frame to obtain the frequency domain information of each frame. The harmonic peak method based on improved confidence is used to identify the pitch, obtain the fundamental tone of the tone, and calculate its multiplier. Based on the timbre parameters obtained in the timbre parameter editing interface, calculate the frequency domain information of the synthesized timbre of each frame, and then perform the inverse Fourier transform to obtain the time domain waveform of each frame; connect the time domain waveforms of different frames by the cross-average method to obtain the time-domain waveform of the synthesized tone (that is, the audio data of the synthesized tone). After collecting the sound of the electronic pipe organ, the audio needs to be denoised, and the imported audio file needs to be parsed to obtain the audio data information [7]. Then, the audio data are frequency-converted and the timbre characteristic information is analyzed; the timbre parameters are obtained through the human-computer interaction interface based on artificial intelligence, and the timbre of the electronic pipe organ is generated.

2. Analysis of Experimental Results of Construction of Electronic Tube Wind and Sound Recognition Database

2.1. Artificial Intelligence. Artificial intelligence replaces human physical and mental power [8], to a certain extent, reduces the number of repetitive jobs required for the

development of cultural industries, in the planning, creation, and production of cultural products. At the same time, higher requirements are put forward for the talent structure, and the input of creative jobs and high-end scientific and technological talents has increased, so that the cultural industry has great development potential and vitality [9, 10]. Artificial intelligence is the most powerful new engine to promote the efficiency of cultural industry production and operation and will help the cultural industry enter a new height [11]. It can be seen that various voice technologies of artificial intelligence make it possible to customize sound and improve the level of audio production. It has broad application prospects in the field of cultural industry and is expected to become indispensable in the production of video and song dubbing and soundtrack production technology [12, 13].

2.2. Electronic Pipe Organ Playing Game Music. Like game design and other popular cultures, video game music has also had its place in the mainstream industry [14]. Nowadays, an internationally renowned symphony orchestra will perform a soundtrack for a game [15, 16]. With the continuous improvement of game development technology, the field of game music is also gradually growing. The electronic pipe organ is also gradually designing the field of game music, its timbre is rich and changeable, it can simulate the performance of a variety of musical instruments, and it can also be played by overlapping multiple timbres [17]. Now is the age of electronic music, the performance of popular music in the electronic organ is very important, and the simulation requirements for various popular emerging electronic musical instruments are also quite high [18, 19]. For example, the electronic pipe organ has the timbre of the jazz organ, and through the setting of the timbre parameters, it can create the ever-changing jazz organ timbre and play colorful jazz music [20]. Therefore, performers must have high performance skills and master the performance characteristics of modern electronic musical instruments [21, 22]. The electronic pipe organ has the function of simulating other musical instruments. To find how to simulate the timbre of other musical instruments more vividly, we must first master the playing skills of the electronic pipe organ [23]. The diversification of musical instruments and the variability of music have created diversified requirements for the touch method of the electronic pipe organ [24, 25].

2.3. Tone. From the perspective of the filter bank, the STFT of the signal is

$$X^{\text{STFT}}_{(k,n)} = \sum_{j=n}^{n+N-1} x(\text{jw})(j-k)\exp\left(-i2\pi\frac{j}{N}\right). \quad (1)$$

Among them, jw is a continuous window function [26]. The kth component of the CQT spectrum of the nth frame signal is

$$X_{(k,n)}^{\text{CQT}} = \sum_{j=n-[N_k/2]}^{n+[N_k/2]} x(j) a_k \left(j - n + \frac{N_k}{2} \right). \tag{2}$$

Among them, k represents the frequency subscript of CQT [27, 28]:

$$K = \left\lceil B \cdot \log_2 \left(\frac{f_{\max}}{f_{\min}} \right) \right\rceil, \tag{3}$$

where B represents the number of subscripts per octave [29]:

$$a_k^n = \frac{1}{N_k} w \left(\frac{n}{N_k} \right) \exp \left[-i 2 \pi n \frac{f_k}{f_s} \right], \tag{4}$$

where f_s represents the sampling frequency. The center frequency f_k corresponding to the subscript k is

$$f_k = f_{\min} 2^{(k-1)/B}, \tag{5}$$

$$Q = \left(2^{1/B} - 1 \right)^{-1}. \tag{6}$$

Among them, Q represents the quality factor. The window length N_k corresponding to the frequency subscript k is

$$N_k = \left\lceil Q \frac{f_s}{f_k} \right\rceil, \tag{7}$$

$$0 < H_k \le \frac{1}{2} N_k, \tag{8}$$

where H_k represents the frame jump range that can be selected.

2.4. Tone Synthesis.

The sound library generation software needs to complete audio file reading and writing, audio collection, audio file analysis, tone analysis, tone parameter acquisition, tone synthesis, and other functions. Considering the applicability of the software and the convenience of operation, the C# language is used to develop Windows in Visual Studio Express 2013. Use the Form application to facilitate user interface design and software system development.

The audio information analysis module is used to analyze audio information, complete audio denoising preprocessing, note extraction, FFTAFFT pitch recognition, octave extraction, audio information display, and other functions; the timbre parameter acquisition module edits and acquires timbre parameters through the human-computer interaction interface. The timbre synthesis module is used to synthesize timbre files, supports the batch synthesis of multiple timbres, and displays the synthesis progress in real time; the analog performance module is used to simulate the timbre or music [30, 31]. The timbre synthesis model is shown in Figure 1.

The additive synthesis method emphasizes the role of the fundamental and overtone series in the sound. Based on the Fourier principle, the signals of different frequencies are superimposed to form a new sound, as follows [32, 33]:

$$x_{f1}(t) = \sum_f A_f \sin \left(2 \pi f t + \phi_f \right). \tag{9}$$

According to the principle of additive synthesis, this paper synthesizes electronic pipe organ timbre, adopts Fourier transform algorithm, and uses the directly measured original signal to calculate the frequency, amplitude, and phase of different sine wave signals in the signal by accumulation. After a lot of analysis and experiments, the immutable features in the model are determined. The specific model is expressed as follows:

$$x_{f_1}(t) = \sum_{k=1}^{K} \sum_{f \in D_k} s_f v_k A_f \sin \left(2 \pi f t + \phi_f \right) + \sum_{f \in D_k} A_f \sin \left(2 \pi f t + \phi_f \right). \tag{10}$$

In the formula, $x_{f_1}(t)$ represents the sound of the electronic pipe organ when the pitch frequency is f_1. Variable frequency range D meets

$$D = \bigcup_k^K D_k = D_1 \cup D_2 \cup \ldots \cup D_k \cup \ldots \cup D_K, \quad k = 1, 2, \ldots K, \frac{x - \mu}{\sigma}. \tag{11}$$

Variable frequency range of section k is

$$f_k - \partial f_1 \le D_k \le f_k + \partial f_1. \tag{12}$$

Among them, f_1 is the pitch frequency. The multiplier of the sound is

$$f_k = k f_1 (k = 1, 2, 3, \ldots, K). \tag{13}$$

The timbre synthesis uses the timbre synthesis model proposed in this article to synthesize timbre to verify the effect of synthetic timbre. The steps for synthesizing tones are as follows:

(1) Framing and windowing the actual electronic pipe organ timbre, and fast Fourier transform (FFT) for each frame to obtain the frequency domain information of each frame.

(2) Use the harmonic peak method based on improved confidence to recognize the pitch, obtain the fundamental tone f, and calculate the multiplier $f = k$.

(3) According to the audio synthesis model of the editable tone, select the appropriate tone parameters (maximum multiplier K, amplitude modification coefficient Dk, synthesis parameter 2, linear s) to form the frequency domain information of each frame of synthesized tone.

(4) Perform inverse Fourier transform of the frequency domain information of each frame of the synthesized tone to obtain the time domain waveform of each frame and then use the cross-average method to connect the time domain waveforms of different frames to obtain the time domain waveform diagram of the synthesized tone.
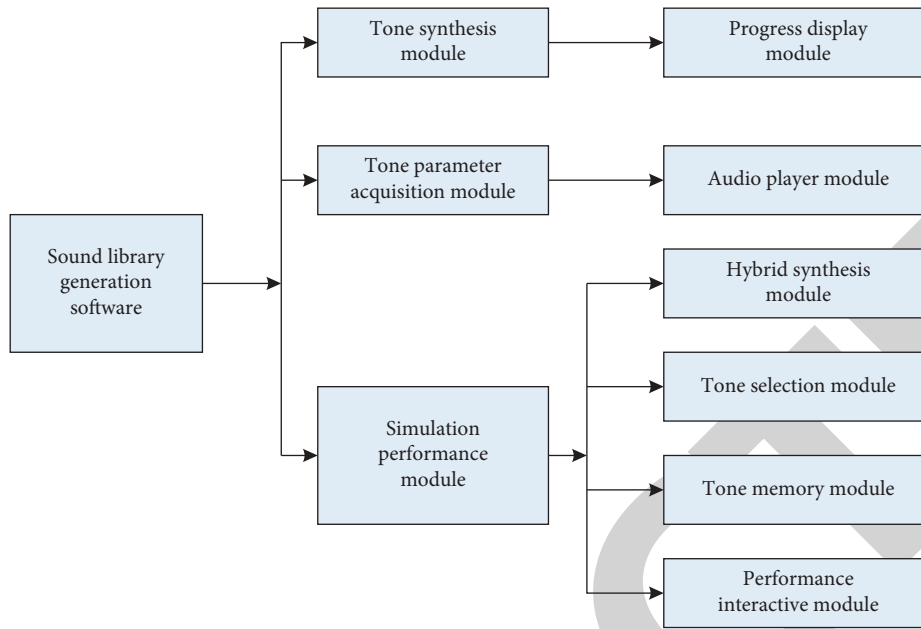
FIGURE 1: Tone synthesis model.

3. Experiment of Constructing Electronic Pipe Organ Tone Recognition Database

3.1. Overall Architecture of Sound Library Generation Software. The electronic pipe organ sound library first needs to input the first frequency, and the software supports 2 first input methods: electronic pipe organ sound collection and the import of existing audio files. After collecting the electronic pipe organ sound, the audio needs to be denoised, and the imported audio file needs to be audio. File analysis was done to obtain audio data information. Then, the audio data are frequency-converted and the timbre characteristic information is analyzed; the timbre parameters are obtained through the human-computer interaction interface based on artificial intelligence, and the timbre of the electronic pipe organ is generated. The timbre generated by analog playback can be saved as an audio file if the timbre effect is satisfactory or directly generate and save the electronic pipe organ sound library in batches; if the timbre effect is not satisfactory, the timbre parameters can be re-edited through the human-computer interaction interface to generate timbre. The overall structure of the sound library generation software is shown in Figure 2.

3.2. Tone Library Generation Process

3.2.1. Audio File Reading and Writing Module. The audio file reading and writing module of the software is used to complete audio file input, audio file analysis, audio file writing, and other functions.

(1) Audio File Import. Click the audio import in the menu file, the open file window will pop up, select an audio file in the ".wav" format to open, and the system will analyze the
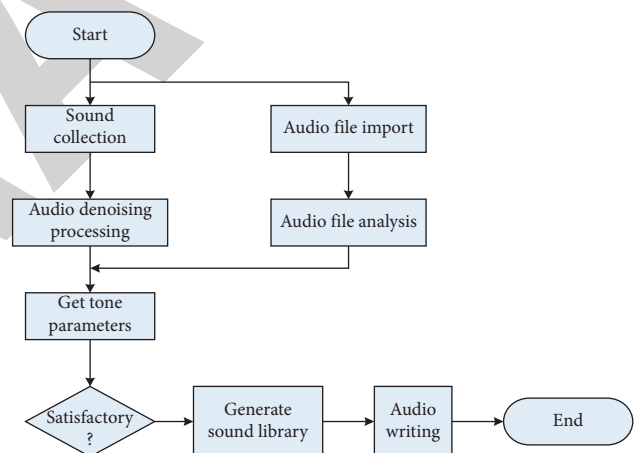

FIGURE 2: Overall architecture of sound library generation software.

audio file. After the analysis is completed, the audio information will be analyzed and displayed.

The structure of a WAV file includes a "RIFF" block, a "fmt" sub-block and a "data" sub-block. The "fmt" sub-block contains the number of channels, sampling rate, number of bits, etc. Perform audio file analysis according to the data format of WAV file, define class WavReader, and complete the analysis and writing function of WAV file. Class WavReader defines read function ReadWAVFile() and write function buildwav() and various variable packaging functions getNumChannels(), getdsize0, getwavdata(), and getwavdata20, variable type conversion, and other auxiliary functions.

(2) Sound Signal Collection. The collection of sound signals is completed through an interactive interface. The upper part of the interface is the audio waveform display

area. When the acquisition is completed, the system analyzes the buffered recording and automatically displays the time-domain waveform of the recording; there are five buttons at the bottom of the interface, and the "collection start" and "collection end" buttons are used to control the sound collection. The module starts or stops sound collection; the "Play" button realizes the playback of the buffered recording; the "Clear" button is used to clear the buffered recording and at the same time clear the audio waveform in the upper area of the interface; click the "Save" button to pop up the save file dialog box to realize the buffer recording preservation.

Under the file menu list, there are three items of "audio import," "audio capture," and "save as." Audio import realizes the audio file import function and supports commonly used WAV format audio files; audio capture through the microphone and sound card of the sound capture module realizes the collection and preservation of sound; save as it is used to realize the writing of audio files.

3.2.2. Audio Information Analysis Module. Audio information analysis module is used for audio information processing and analysis, including denoising preprocessing module, note extraction module, FFTIFFT module, pitch recognition module, octave extraction module, and audio information display module. The pitch extraction module uses the harmonic analysis method with improved confidence proposed in this paper to extract the pitch.

3.3. Tone Parameter Acquisition Module. The timbre parameter module is used to obtain timbre parameters through the interactive page of timbre parameter editing. The top of the tone parameter editing interface is the menu bar, the upper left is the currently open file list, the upper right is the audio waveform display area, the lower part is the tone parameter editing area, and the bottom is the six control buttons. The tone color parameter editing area displays the pitch frequency of the audio. It supports the modification of generation parameters, three types of correction line types, and fourteen spectral coefficients. The generation parameters default to 0.3, the correction line type defaults to linear, and the fourteen spectral coefficients default to 1. The "Reset" button is used to reset the parameters with one key. The "Play" and "Stop" buttons control the playback and stop of the audio. The "Generate Tone Source" button synthesizes the tone based on the current tone parameters and refreshes the audio waveform display. In the area, the "Save" button is used to save the current sound. Click the "Generate Sound Library" button to enter the sound library generation interface and perform batch synthesis of sounds.

3.4. Tone Synthesis Module. The timbre synthesis module realizes the timbre synthesis of the electronic pipe organ according to the current timbre parameters. In order to facilitate the generation of electronic pipe organ sound library, the software supports batch synthesis of multiple tones. The specific steps are as follows:

(1) Obtain the file number n and file name in the audio source path.

(2) Open the first electronic organ tone file under the sound source path, and perform file analysis to obtain audio data.

(3) The audio time domain information (that is, the audio data obtained by file analysis) is divided into frames and windows, and fast Fourier transform (FFT) is performed on each frame to obtain the frequency domain information of each frame.

(4) Use the harmonic peak method based on improved confidence to recognize the pitch, obtain the fundamental tone of the tone, and calculate its multiplier.

(5) Based on the timbre parameters obtained in the timbre parameter editing interface, calculate the frequency domain information of the synthesized timbre of each frame, and then perform inverse Fourier transform to obtain the time domain waveform of each frame; the time domain waveforms of different frames processed by the cross-average method. Connect to obtain the time-domain waveform of the synthesized tone (i.e., the audio data of the synthesized tone).

(6) Write the audio data of the synthesized tone into an audio file with the same name and save it in the save path. The current progress displays +1.

(7) Open the next tube organ tone file in the sound source path for analysis, and repeat steps 3–7 until all n files are completed.

3.5. Simulation Performance Module. The simulation performance module is used to perform electronic pipe organ performance. It can perform real-time timbre synthesis and switching according to performance needs. It supports mixed performance of multiple timbres, which is equivalent to an upgraded version of intelligent electronic pipe organ. The simulation performance module includes a tone storage module, a performance interaction module, a tone selection module, a hybrid synthesis module, and an audio playback module.

The tone storage module is used to store the tone library of different levels for each instrument and is provided to the tone selection module and the hybrid synthesis module; the performance interaction module is used for performance interaction, including setting the tone performance mode, performing the simulation performance of the instrument, and according to the automatic performance. The score takes the score information; the timbre selection module judges according to the feedback information and the score information provided by the performance interaction module, determines whether timbre synthesis is needed, and gives synthesis instructions; the hybrid synthesis module uses the timbre storage module after receiving the synthesis instructions The feedback information and score information provided by the timbre library and interactive module synthesize mixed timbre; the audio playback module performs audio playback.

The simulation method includes the following steps:

(1) Use the interactive unit to perform performance interaction and obtain feedback information, and obtain score information according to the score.

(2) The timbre generation unit judges based on the feedback information and score information provided by the interactive unit and selects the timbre in the timbre storage unit for playback or gives synthesis instructions.

(3) After receiving the synthesis instruction from the tone generating unit, the mixing synthesis unit uses the tone library in the tone storage unit and the feedback information and score information provided by the interactive unit to synthesize and play the mixed tone.

3.6. Audio Pitch Shift and Real-Time Harmony. In sound recording and playback, real-time recording of performance music segments can be stored in the cache, and subsequent changes can be played. However, if the "pf~" technology of the "gizmo" object can be used to realize real-time audio processing, it will be possible to form a harmony group with the audio real-time pitch shift and the original sound. FFT is an efficient DFT algorithm, namely, fast Fourier transform (FastFourierTransform), and the "pft~" object can perform fast Fourier transform on the input waveform to obtain each harmonic component and perform various analysis and processing before performing inverse fast Fourier transform (IFFT), thus getting the output result. The "gizmo~" object that works with the "pfft~" technology can be adjusted in the frequency domain. Based on the FFT algorithm, the speed can meet the real-time requirements.

4. Electronic Pipe Organ Tone Recognition Database

4.1. Tone Effect Analysis. The timbre test effect is shown in Table 1. From the auditory timbre effect, judge whether it is a tube organ audio. In Table 1, the electronic pipe organ timbre discrimination refers to the number of five actual electronic pipe organs judged by the testee as the number of electronic pipe organs, and the synthetic timbre discrimination refers to the number of five synthetic timbres judged by the testee as the number of electronic pipe organs. The data "10 × 5" mean that each of 10 people judged 5 tones as a tube organ, and the data 9 × 5 + 1 × 4 mean that 9 of 10 people judged 5 tones as a tube organ, and one judged 4 tones as a tube organ. In the results of the auditory discrimination test, among the 30 testees' judgments on the 5 synthesized tones, 10 testees who did not receive formal music education (hereinafter referred to as Type A test subjects) judged that all 5 synthesized tones were electronic pipe organ tone, 10 testees who have learned instrument performance for more than 2 years (hereinafter referred to as Type B test subjects) judged that all 5 synthesized tones are electronic pipe organ sounds, and 10 test subjects who have learned electronic pipe organ performance for more than 2 years (hereinafter referred to as among the category C test subjects), 9 people judged that all 5 synthesized tones are electronic pipe organ sounds, one person thinks that 4 are electronic pipe organ

sounds, and one is not electronic pipe organ sounds. In the A and B test subjects, the matching degree of the synthetic tone is 100%, and in the C test subjects, the synthetic matching degree is 98%. In summary, the test match is close to 100%, and it can be concluded that the synthetic tone and the actual electronic pipe organ tone cannot be distinguished by human hearing. From the theoretical comparison experiments and auditory discrimination experiments, the following conclusions can be drawn: the synthetic timbre belongs to the electronic pipe organ timbre, which meets the authenticity and high quality requirements of the electronic pipe organ sound library generation.

The time-domain waveform of the synthesized timbre is shown in Figure 3. In the figure, the abscissa represents the sampling point, and the abscissa divided by the sampling frequency 44100 is the time (in seconds); the ordinate represents the amplitude of each sampling point. The specific tone synthesis model and experimental size are shown in Figure 4. The frequency domain waveform of the synthesized tone is shown in Figure 4. In Figure 4, the abscissa represents the frequency, the unit is 1 Hz, and the range is from 1 Hz to 220S0 Hz; the ordinate represents the amplitude of each frequency.

Comparing the synthesized timbre with the actual electronic pipe organ timbre, the time-domain waveform comparison of the two is shown in Figure 5. In Figure 5, the top is the synthetic tone, and the bottom is the actual tone. The abscissa represents the sampling point. The abscissa divided by the sampling frequency of 44100 is the time (in seconds); the ordinate represents the amplitude of each sampling point. The frequency domain waveform comparison between the two is shown in Figure 6. In Figure 6, the upper is the synthetic tone, and the lower is the actual tone. The abscissa represents the frequency, the interval is 1 Hz, and the range is from 1 Hz to 22050 Hz; the ordinate represents the amplitude of each frequency. From the comparison of the time-domain waveforms of the synthesized timbre and the actual electronic pipe organ timbre in Figure 5, it can be seen that the time-domain waveforms of the two are roughly the same, with only very subtle details different. Therefore, it can be judged that the two are very different from the volume and waveform changes. Similarly, the real-time domain envelope is approximately the same; from the frequency domain waveform comparison of the synthetic tone and the actual electronic pipe organ tone, it can be seen that the frequency domain waveform characteristics of the two are roughly the same, the frequency distribution is clean and clear, and both are at the frequency of the multiplier. The amplitude is larger, and there are more frequency components with smaller amplitude in the low frequency range. This part of the frequency forms the tap sound of the electronic pipe organ at the moment of pressing; that is, the frequency domain distribution of the two is approximately the same. Therefore, according to the timbre is jointly determined by the frequency domain (frequency composition and amount) and time domain, it can be basically judged that the timbre of the two is similar, and both belong to the timbre of the electronic pipe organ. The frequency of the electronic pipe organ is shown in Table 2. The beat of the electronic pipe organ is shown in Table 3.

TABLE 1: Tone test results.

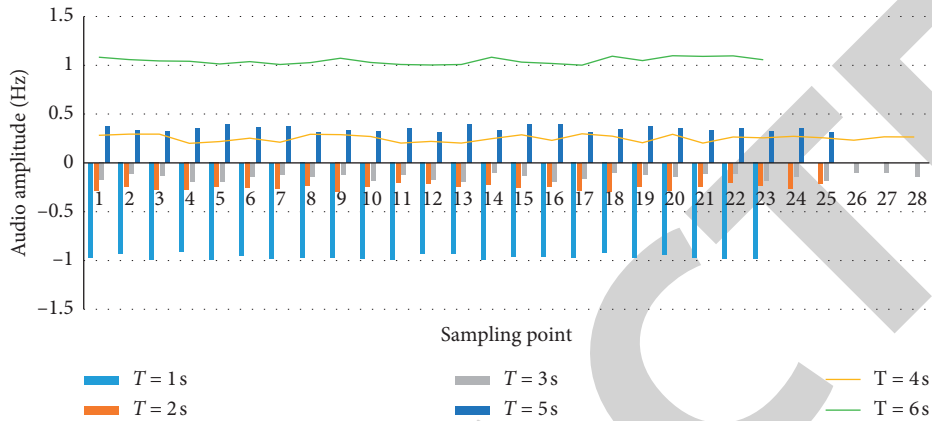| Subject	Piano tone judgment	Synthetic tone discrimination	Suitability (%)
No formal music education (10 people)	10×5	10×5	100
Play musical instruments more than 2 years (10 people)	10×5	10×5	100
Learn piano for more than 2 years (10 people)	10×5	$9 \times 5 + 1 \times 4$	98

FIGURE 3: Time-domain waveform of the synthesized voice.
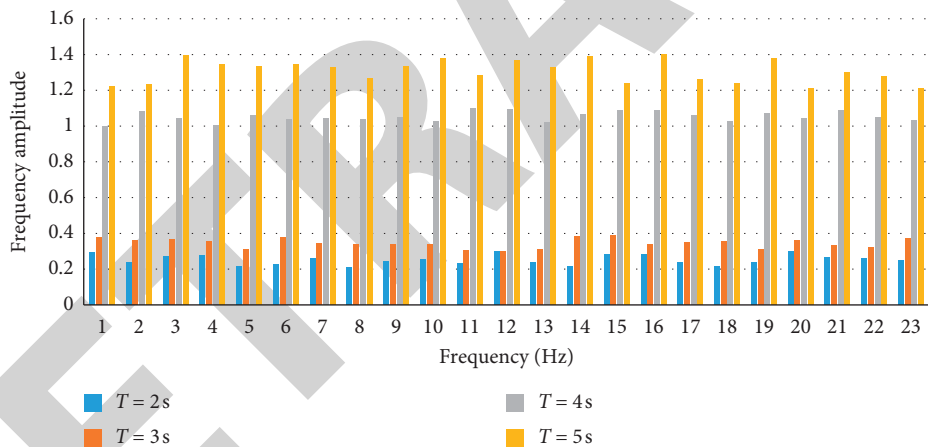

FIGURE 4: The frequency domain waveform of the synthesized tone.

4.2. Recognition Effect of Chords. Choose three different styles of songs. The chord recognition results are shown in Table 4. Through the analysis of the recognition results, it is found that most of the errors occur in empty chords, especially in some fast-paced time periods. Since the analysis window is definite, if the input is a fast-paced segment, it means that the window will span more notes, and a frame of data is likely to contain more than one type of chords, which confuses the system. The beat synchronization analysis algorithm will effectively avoid this problem, because not only is usually the speed of chord changes slower than the speed of beat changes, but also empty chords rarely appear in a beat. In addition to some sporadic errors, some errors are concentrated on the minor a seventh chord and the major C triad. The minor A seventh chord is composed of four pure tones A, C, E, and G, of which C, E, and G are major C triads. The three pure tones included. As mentioned earlier,

because this article regards the minor A triad and minor A seventh chord as the same type, in the presence of pure tone G, the system is likely to mistake the minor A seventh chord as a major C triad. Therefore, we hope that the system can effectively avoid such confusion when increasing the type set of chords to include seventh chords in the system chord classification.

The relative frequencies used when extracting the base collar are shown in Table 5. There are 86 wav files used to test the recognition rate of the system, including typical music of foreign nationalities, classical music, and popular classics. Under this definition, the overall recognition rate of 3762 notes and 286 beats is 88.6%. The accuracy of recognition is still quite high. It must be noted that, in order to facilitate the following comparison, in the abovementioned method of extracting the pitch time value, the difference between two
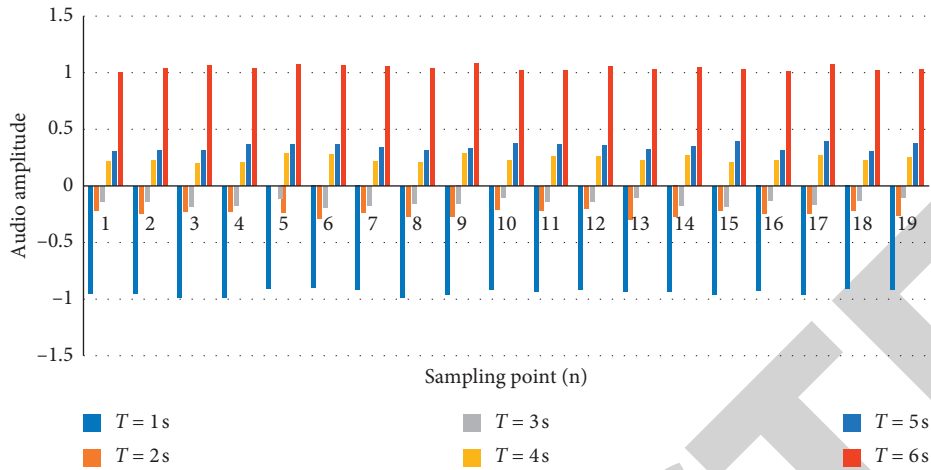
FIGURE 5: Comparison of synthesized sound and actual electronic pipe organ sound.
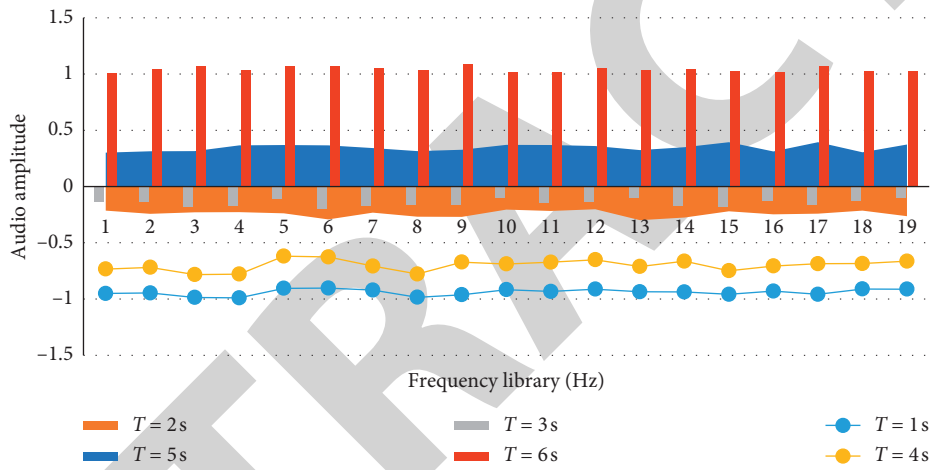

FIGURE 6: Frequency domain waveform comparison between the two.

TABLE 2: Frequency of electronic pipe organ.

Note	5	5	6	2	1	1	6	2
Frequency (Hz)	523.25	523.25	587.33	392	349.23	349.23	293.66	39.2

TABLE 3: Tube organ beat.

Note	5	5	6	2	1	1	6	2
Time (s)	0.5	0.25	0.25	1	0.5	0.25	0.25	39.2

consecutively repeated sounds and a continuous sound of equal duration is not distinguished, that is, in the extraction and normalization. In the process of transformation, for example, two types of sounds are considered equal. It can be seen that the experimental results and theoretical values have a good approximation. Here, the normalized way is to use the smallest time unit in the music (such as quarter beat, etc.) as the metric for the duration of each note played and take its proportional value.

The research result of the dual-tone pitch time value extraction module is shown in Figure 7. The experimental data are still derived from the performance of learners of electronic pipe organ performance. However, the data used in the module test are the basic two-tone chords in music and will not change after triggering the vibration; that is, it is stable on two notes. To test the performance of this module, 100 two-tone chords with different ranges, different spans, and different degrees of harmony were selected. The dual-tone pitch time value extraction module accurately reflects

TABLE 4: Recognition results of chords.

Test data	Length	Recognition rate (%)
Fragments of piano music	8.26 seconds/44 frames	81.1 8
Song please, please me	125 seconds/662 frames	78.70
Song yesterday	121 seconds/644 frames	74.53

TABLE 5: Relevant frequencies used when extracting base collar.

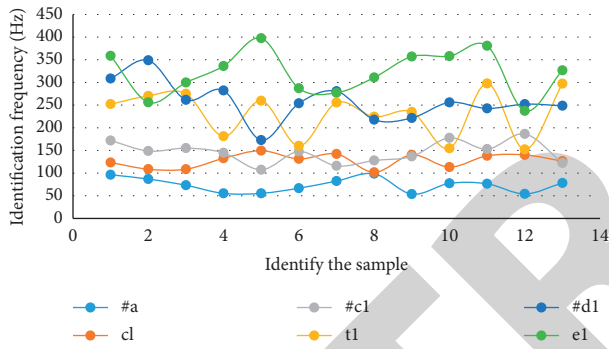| Musical alphabet	Calculated frequency (Hz)	Normalized starting frequency (Hz)	Normalized stop frequency (Hz)	Program use pitch (Hz)
#a	220	213	235	220
cl	233.08	226	239	233
#c1	246. 94	240	253	217
t1	261.63	254	268	262
#d1	277.18	269	281	277
e1	293. 66	285	301	294
f1	311.13	302	319	311
#f1	329. 63	320	338	330
g1	349.23	339	358	349

FIGURE 7: Research results of the dual-tone pitch time value extraction module.

the chord time value and pitch information. The data input to the module in.wav format is also output as a time-event graph. In the figure, two straight horizontal lines distinguish the time distribution of the central *c* and central *g*. Through this module, the corresponding height and vibration time of the dual-tone chord can be analyzed. We tested on 100 experimental samples, 82 samples with recognition rate above 95%, 1 sample with recognition rate lower than 50%, and 6 samples lower than 70%. Observing the experimental results, it is found that such experimental samples are samples of absolute concord chords in music, and the two-tone interval is more than 13 degrees apart; that is to say, this module has a low resolution of the absolute consonant interval of the polytone, which is due to its frequency spectrum caused by aliasing. This module has a low resolution effect on dual-tone experimental data with frequent changes in pitch and a relatively high computational

complexity. This is a place that can be further improved in future work.

4.3. Basic Function Analysis. The basic function test result is shown in Figure 8. The electronic keyboard product will enter the main interface of the auxiliary system about 30 seconds after powering on. The main interface defaults to the first big button in the upper left corner. Clicking the big button will switch the bottom panel between tone selection and tone attribute compilation. When the bottom panel has multiple pages, you can turn pages, and you can also turn pages. Each control can be adjusted normally. The default value of the slide bar is generally 63, which can be adjusted between 0 and 127 with the button or the scroll wheel, and there is no border crossing. Press any one of a group of rhythm keys on the keyboard, the bottom panel of the main interface will become the rhythm selection panel, and press it again to become the rhythm attribute editing panel of the corresponding group. Pressing the exit key when any interface is in an idle state will return to the main interface. If you do not change the main interface parameters through the keyboard physical keys, the main interface will remain the same as when you left it the last time. The pop-up edit box shows the selected file, and the suffix name is hidden in the edit box. In the state of renaming, the cursor in the edit box can be moved and characters can be deleted or inserted in the corresponding position, which can realize the input of English and number punctuation. And the system has a sound processing method for a variety of naming accidents. The overall function test is shown in Table 6. The test data of the file management module are shown in Table 7. The electronic pipe organ is shown in Figure 9.
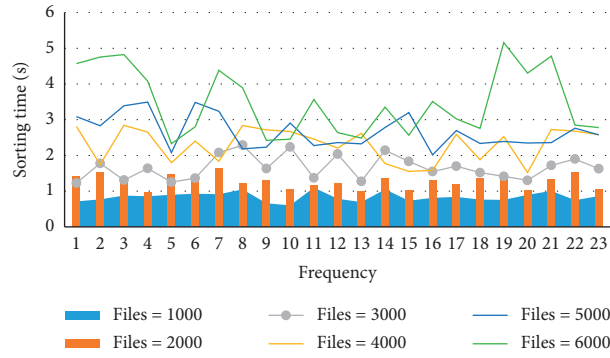
Figure 8: Basic function test results.

Table 6: Overall functional test.

Frequency	Start time (s)	Key response (ms)	File transfer	Record audio (M)
1	15.6	20	1.7 M/s	6
2	16.2	30	1.6 M/s	7
3	15.9	20	1.6 M/s	8
4	17.1	25	1.8 M/s	7
5	16.4	25	1.9 M/s	8

Table 7: File management module test data.

Frequency	Number of files	File layers	Positioning time (s)	Sort time	Copy rate	Average download speed
1	1000	2	1.2	2.0 s	1.8 M/s	504 K/s
2	2000	3	1.5	2. 4 s	1.8 M/s	485 K/s
3	3000	4	1.6	3.2 s	1.8 M/s	512 K/s
4	4000	5	1.9	3.6 s	1.8 M/s	492 K/s
5	5000	3	2.3	4.3 s	1.8 M/s	497 K/s
6	6000	2	2.0	3.5 s	1.9 M/s	502 K/s

Figure 9: The electronic pipe organ (from http://alturl.com/qsobx).

5. Conclusion

The electronic pipe organ sound library first needs to input the first frequency, and the software supports 2 first input methods: electronic pipe organ sound collection and the import of existing audio files. After collecting the electronic pipe organ sound, the audio needs to be denoised, and the imported audio file needs to be audio. File analysis was done to obtain audio data information. Then, the audio data are frequency-converted and the timbre characteristic

information is analyzed; the timbre parameters are obtained through the human-computer interaction interface based on artificial intelligence, and the timbre of the electronic pipe organ is generated. The timbre generated by analog playback can be saved as an audio file if the timbre effect is satisfactory or directly generate and save the electronic pipe organ sound library in batches; if the timbre effect is not satisfactory, the timbre parameters can be re-edited through the human-computer interaction interface to generate timbre.

The simulation performance module includes a tone storage module, a performance interaction module, a tone selection module, a hybrid synthesis module, and an audio playback module. The tone storage module is used to store the tone library of different levels for each instrument and is provided to the tone selection module and the hybrid synthesis module; the performance interaction module is used for performance interaction, including setting the tone performance mode, performing the simulation performance of the instrument, and according to the automatic performance. The score takes the score information; the timbre selection module judges according to the feedback information and the score information provided by the performance interaction module, determines whether timbre synthesis is needed, and gives synthesis instructions; the hybrid synthesis module uses the timbre storage module

after receiving the synthesis instructions, synthesizing mixed sounds from feedback information and score information provided by the sound library and interactive module.

The audio playback module performs audio playback. Use the interactive unit to perform performance interaction and obtain feedback information, and obtain score information according to the score; the timbre generation unit judges according to the feedback information and score information provided by the interaction unit, selects the timbre in the timbre storage unit for playback, or gives synthesis instructions; mix after the synthesis unit receives the synthesis instruction from the timbre generation unit, and it uses the timbre library in the timbre storage unit and the feedback information and score information provided by the interaction unit to synthesize and play the mixed timbre.

Data Availability

The data that support the findings of this study are available from the corresponding author upon reasonable request.

Conflicts of Interest

The authors declare that they have no conflicts of interest.

Acknowledgments

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