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Computer simulation analysis on piano timbre

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ABSTRACT

Computer Simulation technique could achieve a positive effect on the analysis process of piano timbre, which help achieve the ultimate goal of making the piano timbre judgment more scientific and rational. However, according to previous studies, researchers do not attach much attention to the relationship between waveform around five peaks in the frequency spectrum and specific amplitude ratios, which will have a relatively direct impact on the timbre of the piano and makes piano timbre analysis process without strong comprehensive theoretical support and computer simulation process less scientific and rational. However, this research intends primarily to discuss the piano tone model, makes the two types of parameter interface established, and provides effective access to parameters based on the premise of tone. This essay also discusses the Fourier analysis of combination tone and the reconstruction of timbre of the piano, making the tone spectrum analysis of the piano playing and the reconstruction of the piano tone effectively guaranteed. It also provides a solid theory and data support to computer simulation analysis of piano tone. This is the main idea of this research process, which can also fully reflect the specific objectives of the study and the specific methods, with the aim to provide a solid foundation of theory and data to further researches.

KEYWORDS

Piano timbre; Computer; Simulation analysis; Experimental study.

INTRODUCTION

The analysis of piano timbre can mainly be realized by means of computer simulation. So this process needs to build its model by computer, making the two different parameter interfaces to be established. This study effectively analyses the modeling of piano timbre, timbre of the parameter acquisition, Fourier analysis of the tone and the reconstruction of timbre of the piano, to investigate the specific factors of the analysis of the timber of piano and to further improve the computer simulation analysis

MODELING OF PIANO TIMBRE

In the field of piano music, spectrum and amplitude envelope have a decisive role on each of the timbres, so the model of timbre can be expressed with the following formula.

$$T(p,t) = A(p,t) \bullet E(p,t) \quad \textbf{(1)}$$

In the formula above, the meaning of p refers to the absolute pitch, while t represents the time. T(p， t) is the concrete embodiment of the sound function, and this is the form of basic embodiment of the characteristics of piano timbre. However, $A(p,t)$ represents concrete number variations of the amplitude, from which we can see the spectral characteristic. $E(p,t)$ referred to amplitude envelope function, as well as its main forms of the effective reflection of amplitude contours. The following discussion focuses on the related research and exploration of the modeling process of these two features.

Establishment of spectral model

The spectral model refers to the combination of characteristics is frequency characteristics in the field of piano music as is well-known. In the piano music, the sound is not in a state of harmony, and the relationship between audio and pitch fails to show integer relationships, mainly because there are some differences among stiffness[1]. For this, during the establishment of the timbre model, inharmonious factors which are not relevant are not taken into account, thereby enabling this model to maintain a high degree of simplicity, which gives the piano sound the core of feeling. However, in the course of piano music playing, people often do not feel the music phase, so it is not necessary to consider this factor in the establishment of the spectral model. The piano audio model can be reflected by the following formula.

$$A(p,t) = \sum_{i=1}^{n} \{c_0(p,i) A \sin[2\pi i f(p)t]\} \quad \textbf{(2)}$$

In the formula above, i represents is the basic variables of the number of overtone, while co (p,i) is an overtone peak amplitude‘s specific coefficient. However the basic parameter position of the pitch during the process is positioned as 1, and the harmonics amplitude coefficient can be obtained by the piano music sample analysis. In the formula above, A represents the peak amplitude of pitch size. The f (p) represents the pitch frequency of p. The calculation can be obtained by combining the first international piano with pitch frequency ratio between the two adjacent keys, and p is an absolute high encoding. The formula is shown as followed:

$$p = O + K + S \quad \textbf{(3)}$$

The TABLE 1 shows the relevant encoding process of 12 key signatures. However, it is not difficult to see from the results that there are 15 key signatures, which means that 3 of them are repeating tones.

TABLE 1 : Key signature coding table

Key signature(1=)	C	*C	D		*	*	*	G	*	A	*	B
Coding K	0	1	2	3	4	5	6	7	8	9	-2	-1

In the formula above, S represent is relatively high encoding. At the same time, the encoding of the bass is set to 1, and the encoding of the tenor is set to 13, and the encoding of the treble is set to 25, which allows relatively high notes to rise to the half treble with the bass and the code units to gradually rise. Its rising degree of encoding is 1, which can be seen in TABLE 2.

TABLE 2 : Relative pitch coding table

Scales	1	2	3	4	5	6	7
Encoding of the bass	1	3	5	6	8	10	12
Encoding of the tenor	13	15	17	18	20	22	24
Encoding of the treble	25	27	29	30	32	34	26

Establishment of the amplitude envelope

In the course of playing one note by the piano, the basic process of sound production is made up of three stages: the first is sound, followed by attenuation, and the last is the process of passing. The establishment of the amplitude envelope can be reflected byby following formula. However, the curve changes within the cycle of amplitude envelope can be fully seen in Figure 1.

$$E(p,t)=\begin{cases} k_1 t, & \mathrm{t}\in(0,\mathrm{t}_1) \\ k_2 t-k_2 t_2+E_2, & \mathrm{t}\in(\mathrm{t}_1,\mathrm{t}_2) \\ E_2\exp[-\alpha(t-\mathrm{t}_2)], & \mathrm{t}\in(\mathrm{t}_2,\mathrm{T}_e) \end{cases} \tag{4}$$

From the formula above you can see, E (p,t) is a basic function of the amplitude envelope, and T_e represents the basic cycle of the envelope. In this research study, by the corresponding experimental procedure of the establishment of amplitude envelope model, in a beat, E1, E2 represent the amplitude envelope coefficients of the t1,t2 respectively, while the k1,k2 represents the slope of the two preceding paragraphs. In this $k_1=E_1/t_1, k_2=\left(E_2-E_1\right)/\left(t_2-t_1\right)$. A represents the recession factor in the third paragraph by an easier description. The time coefficient can by defined. c_t1,c_t2 respectively represents the time coefficient. Therefore, the two equations conditions of $c_{\_t1}={}_{t1}/_{Te,}c_{\_t2}=\left({}_{t2}-{}_{t1}\right)/_{Te}$ can be set up. In this study, its amplitude envelope parameter can be set as c_t_1, c_t_2, E_1, E_2, and A. However, in the case of the process of playing the piano, pitch difference among pitches can cause different amplitude envelope parameters.

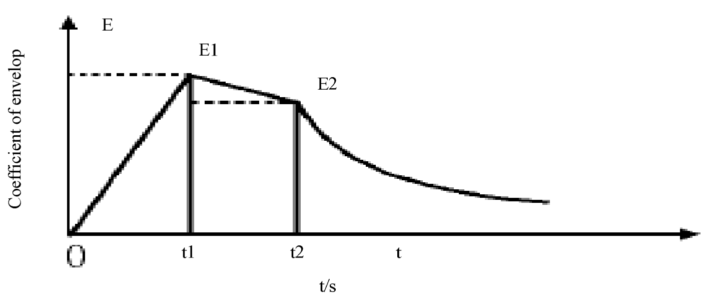

Figure 1 : Amplitude envelope curve of the piano

TIMBRE OF THE PARAMETER ACQUISITION

This study above has effectively established the timbre model and from the establishment, two different types of parameters in interface can be obtained. The first type is the spectral parameters and the other type is envelope parameters. Two different types of parameters can be represented as C_o(p，i)and c_t_1、c_t_2、E_1、E_2. However, after these two parameters have been obtained, the tone of the piano can be obtained too. Howeve in the process of acquiring envelope parameters, the specific method can be achieved by the waveform images. Its main principle is based on sound samples to give a valid receipt, enabling the waveform effects to be expressed through visual form and envelope parameters to be appropriately estimated. For access to spectrum parameters, method is to effectively collect piano music samples, and to efficiently analyze based on Discrete Cosine Transform method, which enables to give a sequence of spectrum parameters a clear embodiment.

Collection of piano music samples

In general process of collecting music, the method of collecting samples is the recording, but in the music recording, the inevitable noise will appear so there will be errors in the following analysis, which will exert corresponding negative impacts. However in music recording, if ultra-low noise recordings are used, the investment of human, material and financial resources will increase. The piano music collection method is concrete way of curing analysis, allowing music samples to get professional treatment, which allows the piano music of the sample collection process to reflect the low noise music samples[3].

In the piano keyboard, there are 88 buttons with 52 white key and 36 black keys. But for the proportion between two adjacent keys in the piano keyboard, the piano music collection largely focuses on the sample of the white keys. And the sample of the black keys is collected by alternatively collecting the sample of the white keys. Sound samples collected are C1-B1、C-B、c-b、c1-b1、c2-b2、c3-b3 and c4-b4 respectively. And these sounds do not include the sound variations. So the music can be validly edited, as is shown in Figure 2.

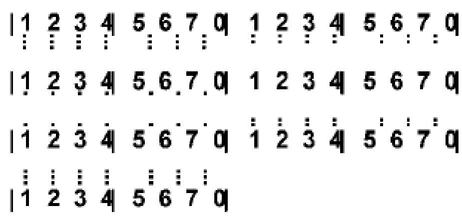

Figure 2 : Syllable music

After the valid edit of the scale scores, the original MIDI files and WAV format files can be effectively conversed and stored.

Analysis of piano music samples

In analysis of the music samples, its main purpose is to effectively obtain the spectrum parameters, and the method that is selected is DCT method.

After the effective arrangement of the musical spectrum, the spectrum will be processed by DCT calculation and analysis, which enables the nature of the musical spectrum to be fully reflected. The algorithm mainly contains four steps: First one is the effective projection of the width of gene frequency in the spectrum, whose equation is $S(1) = f(p)/\Delta f$, $\Delta f = F/(2N)$, and it is a sample spectrum represented by its own frequency[4]. Second it is to effectively obtain the samples numbers of the overtone i, with $S(i) = i * S(1)$ to be reflected. And d in the S(i) will be determined, which can be obtained by the formula of $[S(i) - S(1)/2, S(i) + S(1)/2]$. Next step is to find out the maximum value and minimum value of Ac(u) in the field of D, to ensure that the DCT coefficients of i times of overtone are small by the specific formula $A'(i) = A'_{\max} - A'_{\min}$. The last is the calculation of the amplitude peak of i times of overtones, with the basic formula of $C_0(p,i) = A'(i)/A'(1)$.

THE SOUND OF FOURIER ANALYSIS AND RECONSTRUCTION OF PIANO TIMBRE

Spectral analysis of piano music playing

In the computer simulation analysis of piano timbre, in order to ensure the analysis to be more scientific and representative, the single factors among different ranges of multiple pianos have been effectively analyzed in this experiment, with a large amount of materials and diverse settings in the aspect of playing, which leads to differences between playing dynamics and playing types. However, the sound of the piano that has been collected has been transformed effectively, so that the time domain waveform and frequency domain waveform can be fully reflected. The analysis method is the method of Fourier analysis. The specific discussion can be conducted and its specific process is shown as followed.

First of all, by the corresponding calculation, the piano frequency of a^1 that can be obtained is 430 Hz, which results in a slight gap with the theory of value of f_{a1}. The main reason for this gap is because a certain gap between the accordatura and standard sound among each string in the piano. It also is the fundamental reason for the frequency differences[5].

The second is to effectively obtain the fundamental frequency or frequency domain range around 100 points within the frequency range. The phase information can be fully reflected at the time, and the other value is set as 0. In this case, the time domain waveform can be effectively recovered, which in turn makes relevant basic characteristics, such as the pitch in the original playing of the piano and values to be effectively obtained and to provide a solid foundation for the analysis of spectrum in the playing of the piano.

The last is to effectively obtain the waveform around the headmost five peaks within 100 points of the frequency domain waveform and then convert it into time domain waveform. The specific comparison between two kinds of waveforms confirms that there is no specific change between the two waveforms which in turn makes relevant basic characteristics, such as the pitch in the original playing of the piano and values to be effectively obtained. After that, the specific ratio of the five peaks in the frequency domain will be changed, which makes echo or reverberation conditions created, and proves the change of tone from this occurs. However, the mean value around the harmonic frequency ranges from one to five does not make any changes, but only the waveforms around harmonic frequency ranges from one to five can be effectively changed. Therefore, the changes of sound waveform amplitude are greatly obvious.

From the experiments above you can see, in previous studies and discussion, the waveform around the five peaks in the spectrum and relationship among specific amplitude ratios do not receive enough attention, which has direct impact on the timbre of the piano[6].

Reconstruction of the piano timbre

Human ear judges the tone height by the fundamental frequency. The analysis of the actual performance data and the waveforms, can tell that the sound energy proportion from the fundamental wave in the frequency domain to five times harmonic wave and frequency waveform in the small range of harmonic frequency reflect that the playing pitch, timbre and the information of sound energy. In the experiment, the function simulation is designed as the waveforms from the fundamental frequency to five times harmonic frequency ranges. On the basis of the fundamental frequency of the known accordatura and the proportion of the harmonic frequency peak values from one to five, all the music of the piano timber can be reconstructed successfully.

Four similar functions are selected in the experiment (Cauchy function, Gaussian function Sinc functions cosine and sine - weighted Cauchy function) to get the frequency domain waveform of the actual playing of the piano. The fourth function is used to describe the fundamental frequency and frequency doubling information to more appropriately simulate the piano timbre[7]. When the appropriate parameter is selected, Fourier Transform domain is turned into time domain, and there is little difference with the piano timbre which is actually acquired, but its sound effects and MIDI the timbre of Grand Piano in MIDI are quite similar. In the experiment which uses the Weighted Cauchy function to get to the frequency domain waveform, $Y(j,\omega)$ is used to represent for the discrete of Fourier Transform with the discrete time signal $Y(n)$. $Y(j\omega)$ can be obtained by the following formula:

$$Y(j\omega)=\sum_{i=1}^{n}Y_i(j\omega),Y_i(j\omega)=S_i(j\omega)*F_i(j\omega), \tag{5}$$

$$S_i(j\omega)=\begin{cases} A_i\dfrac{a_i}{a_i^2+(\omega-\omega_i)^2} & \omega_i-\alpha_i\le\omega\le\omega_i+\alpha_i, \\ A_i\dfrac{2}{|(\omega-\omega_i)|} & \text{other} \end{cases} \tag{6}$$

In the formula above, ω_i refers to the fundamental frequency of the sound or the frequency multiplier, A_i is the base frequency point or amplitude of the multiplier points, a_i is used to adjust the fundamental frequency or the width of the frequency doubling waveforms around. F_i is a function of sine and cosine.

CONCLUSION

Above all, it is the relevant research and exploration of the computer simulation analysis of the timbre of piano. It mainly focuses on the establishment of the model of the piano timbre, the Fourier analysis of sound and the two parts of the reconstruction of the piano timbre. These timbre parameters help to establish the process effectively and to achieve the ultimate goal of improving the computer simulation analysis.

REFERENCE

[1] Fu Quanwei; The collision and fusion of traditional and modern-musical works in terms of digital music, Musical works, **(7)**, 126-127 **(2013)**.

[2] Li Min; Some thoughts on the establishment of vocal music accompaniment music library, Arts Criticism, **(6)**, 130-132 **(2012)**.

[3] Li Yujia; The aesthetic paradox of computer music in the digital age, Journal of Northwest University, Philosophy and Social Sciences Edition, **41(3)**, 112-116 **(2011)**.

[4] Xu Dengfeng; Computer supporting system of music technique and its digitization, Hundred Schools in Arts, **(6)**, 230-233 **(2010)**.

[5] Cao Xihe, Cao Ying, Sun Lin; Piano timbre simulation by computer oriented to preschool education and teaching, Journal of Henan Normal University, Natural Science, **38(1)**, 47-50 **(2010)**.

[6] Wang Feng, Zhang Xueying, Li Bingnan; Research and discuss of chord recognition based on PCP, Mathematics in Practice and Theory, **(5)**, 97-101 **(2010)**.

[7] Li Juan; Study on computer aided music production courses for young children, Journal of the Chinese Society of Education, **(S1)**, 38-39 **(2010)**.