International Journal for Housing Science and Its Applications
Publish August 4, 2025. Volume 46, Issue 3 Pages 2222-2230

https://doi.org/10.70517/ijhsa463186

Application of Debussy's matrix-based analysis of tonal color and acoustic properties in modern music composition

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Abstract Musical feature matrices were developed with computer technology and have been most fully developed in modern music composition in recent years. This paper extracts and analyzes the timbral feature matrix and tone intensity change envelope curve of Debussy's piano piece using the spectral analysis method of short-time Fourier transform. Also through the MIDI file in the extraction of each note length, pitch, pitch velocity, channel and other information, the establishment of note feature matrix, for digital music creation to lay the foundation. The experimental results and analysis show that the method in this paper can accurately recognize the time value of the piano music score, and its error values are all within the range of (-5%, 5%). The BER of the proposed algorithm in this paper is lower, the signal-to-noise ratio of the recognition algorithm is smaller, and there is a clear delineation of audio signal features. The tonal color and acoustic characterization of three piano pieces, Prelude, Moonlight, and Minuet, are completed. Finally, the sound color of “Moonlight” is improved with sine wave musical notes, which lays the foundation for modern music creation.

Index Terms feature matrix, short-time Fourier transform, Debussy, audio signal

I. Introduction

Laude Debussy, a French composer and music critic, is one of the pioneers and representatives of French Impressionist music [1]. Piano composition continued throughout Debussy's life, and his early "Arabesque" and "Bergamo Suite" were close to the Romantic style. "Prints", "Happy Island", two episodes of "Imagery" and "24 Preludes" are masterpieces of impressionism [2]-[4]. His music not only enriched the art of piano music in the 20th century, but also created a unique style in the history of piano art, which made him a representative figure of impressionist music [5], [6]. The 24 Preludes is called “the essence of impressionist music”, in the middle and late stage of Debussy's music creation, focusing on the full display of his mature and rich compositional techniques, which break through the boundaries of the traditional tonal language, Debussy's most mature, representative and creative works [7]-[10]. Before that, he still inherited the musical style of the Romantic period.

Debussy's musical works are full of love for nature and life, as well as innovations in musical form and structure [11]. His compositions often show exquisite treatment of melody and harmony, as well as exploration of color and timbre [12], [13]. Debussy focused on the mood and emotional expression of music, pursuing to transcend traditional musical forms and create musical works with unique flavor [14], [15]. In his creative career, Debussy constantly explored and challenged the boundaries of traditional music, making his works occupy an important position in music history [16]. His music has had a profound impact, not only inspiring many subsequent composers, but also on the entire music world and even modern music creation [17], [18].

In order to ensure that the recognition of the essence of the piano spectrum is clear, this paper is based on Matlab software, short-time Fourier transformations as the theoretical basis, the preparation of the function program, to obtain a single-tone spectrum. Through the short-time Fourier transform spectrum analysis method to extract the timbre feature matrix, to obtain the sound intensity change envelope curve. Taking the two as the carrier, analyzing the relationship between tone intensity and time change of piano music, the piano music sound can be clearly identified. By parsing the MIDI file format and establishing the cope feature matrix, the pitch, pitch velocity, timbre and other information of each note are obtained, and the start and end time of each note are calculated. Based on the derived sinusoidal signal, the matrix signal is converted to realize the musical sound simulation and modern music creation. In this paper, we take monophonic files as experimental samples to verify the effectiveness of this paper's method in music feature recognition.

II. Extraction method based on music feature matrix

With the rapid development of information technology, the number of digital music has increased, how to make good use of the recognition technology, which can be transformed into digital music can greatly improve the efficiency of music production, and at the same time to ensure the quality and effect of music. However, in the previous digital music creation process, the traditional music production methods were mostly followed, which could not well reflect the advantages and strengths of digital music, and could not improve the production efficiency, quality and popularity of music. Therefore, people are no longer satisfied with the general traditional mode of music creation, matrix operation recognition technology provides a more efficient method for digital music creation.

II. A. Short-time Fourier transforms

In digital signal processing technology, the Fourier transform and its inverse transform is a method of signal time-frequency analysis. This method links the time-domain description and frequency-domain description of the signal together, and the time-domain signal can be transformed into a frequency-domain signal by positive transformation, and the frequency-domain signal can be transformed into a time-domain signal by inverse transformation to be analyzed. However, the Fourier transform and its inverse transform is a signal of the overall transformation, either completely in the time domain for analysis and processing, or completely in the frequency domain for analysis and processing, can not get the signal spectral content of the law of change over time, that is, the traditional Fourier transform and its inverse transform is not applicable to non-semi-stable signals, there is no limitation in time and frequency domain segmentation of the defects. To overcome the above defects, it is necessary to find joint time-frequency methods [19].

The short-time Fourier transform is proposed, which can combine the time-domain and frequency-domain analysis of the signal, and the result of the transform reflects both the frequency content of the signal and the law of change of the frequency content with time.

The process of short-time Fourier transform is as follows: multiply the time signal with a window function whose time width is narrow enough, and the signal inside the window is approximately regarded as a smooth signal, and then carry out the Fourier transform inside the window to obtain the instantaneous spectrum of the signal. As the intercept window is moved on the time axis, the spectrum is obtained over the entire time domain. The expression for the continuous short-time Fourier transform is:

$$S(\omega,\tau)=\int_{-\infty}^{+\infty}x(t)g^{*}(t-\tau)e^{-j\omega t}dt \tag{1}$$

where $x(t)$ is the signal to be analyzed, * is the complex conjugate symbol, and $g(t)$ is the window function. In the short-time Fourier transform, $g(t)$ performs the time domain restriction and $e^{j\omega t}$ performs the frequency domain restriction. The result of the short-time Fourier transform of the signal, $S(\omega,\tau)$, reflects the relative containment of the signal components of the signal to be analyzed, $x(t)$, at time τ, frequency ω, and the signal can be expanded on the window function as a window $[\tau-\delta,\tau+\delta],[\omega-\varepsilon,\omega+\varepsilon]$ region of the state, δ for the time width of the window, ε for the frequency width of the window, the smaller the width of the window, and accordingly, the higher the resolution in the time-frequency analysis. In order to have a better time-frequency analysis, δ and ε should be small enough, but according to the principle of inaccuracy, δ and ε are constrained by each other, and both of them cannot be arbitrarily small at the same time. To prove the point, the following two extreme window functions are taken as examples.

(1) The smaller the time width, the higher the time resolution in the corresponding time-frequency analysis. The typical case is that the window function takes the unit impulse function $\delta(t)$, at this time the expression of the short-time Fourier transform is:

$$S(\omega,\tau)=\int_{-\infty}^{+\infty}x(t)\delta(t-\tau)e^{-j\omega t}dt=x(t)e^{-j\omega t} \tag{2}$$

From equation (2), the short-time Fourier transform has degenerated into the signal $x(t)e^{j\omega t}$, which retains the temporal transformation of the signal, but does not provide any frequency resolution, and is unable to realize the localization function in the frequency domain.

(2) When the window function is unchanged, i.e., the window function $g(t)=1$, the expression of the short-time Fourier transform at this time is:

$$S(\omega,\tau)=\int_{-\infty}^{+\infty}x(t)e^{-j\omega t}dt=X(j\omega) \tag{3}$$

From equation (3), the short-time Fourier transform has been degraded to a simple Fourier transform, which does not provide any time resolution and cannot realize the localization function in the time domain.

That is, when a window function is selected to carry out the short-time Fourier transform of the signal, the obtained time resolution and frequency resolution will be mutually exclusive, when the signal frequency resolution is high, its time resolution will be reduced, and vice versa.

In the short-time Fourier transform of the actual signal, due to the acquired signal are discrete signals, need to discretize the continuous short-time Fourier transform. If the acquired signal is $x(n)$, the expression for the discrete short-time Fourier transform of the signal is:

$$S(\omega, mN) = \sum_n x(n) g^*(n - mN) e^{-j\omega n} \tag{4}$$

where N is the step size of the window function moving on the time axis. However, the discrete-time short-time Fourier transform is only discrete in time and continuous in frequency, and the frequency ω needs to be discretized as well. By dividing ω into M points with a period of 2π, equation (4) is transformed into:

$$S(\frac{2\pi}{M}k, mN) = \sum_n x(n) g^*(n - mN) e^{-j\frac{2\pi}{M}kn} \tag{5}$$

From Eq. (5), the result of this discrete Fourier transform is a two-dimensional complex matrix with the sampling time points as the rows and their corresponding frequency values as the columns, and the modes of this matrix form the corresponding spectral magnitude matrix.

II. B.Piano Spectrum Analysis

The piano's entire articulation frequency band is maintained in the Fourier transform analysis region. The tuner can only assess the error of more than 0.5Hz frequency, in order to ensure the spectrum is precise and clear, using professional electronic frequency instruments, testing the piano's articulation fundamental frequency through the proofreading equipment, in order to accurately regulate the sound height of the strings, based on the Matlab software, according to the principle of the Fourier transform, the preparation of the FFT () function program, to obtain a single audio spectral analysis graph, indicating that the piano with multiple octave overtones as well as octave vibration In this way, it is clear that the piano timbre is uniquely rich in characteristics [20].

II. C.Tone feature matrix extraction

Since the weighted Cauchy function waveform is close to the piano waveform and can describe the fundamental frequency and octave analysis, the simulation effect is similar to the actual tone, so the five octave method is chosen to simulate the extraction of piano tone. Assume that the discrete Fourier transform of the discrete event signal x(m), i.e., $X(i\lambda)$, can be expressed as:

$$X_j(i\lambda) = T_j(i\lambda) \cdot P_j(i\lambda) \tag{6}$$

$$T_j(i\lambda) = \begin{cases} B_j \dfrac{b_j}{b_j^2 + (\lambda - \lambda_j)^2}, \lambda_j - b_j \le \lambda \le \lambda_j + b_j \\ B_j \dfrac{2}{|\lambda - \lambda_j|}, \text{other} \end{cases} \tag{7}$$

where λ_j represents the j tone fundamental or octave frequency and B_j represents the amplitude; b represents adjusting the waveform width around λ_j, P_j represents the sine-cosine function, based on the weighted Cauchy function, constructing the frequency domain waveform, and performing the piano playing simulation with the FFT transform to accurately simulate the playing timbre, intensity, and pitch for computerized performance [21].

Function simulation has limitations, namely, lack of expressiveness and timbre, complicated function calculations, excessive load, detailed record of amplitude ratio correlation with timbre matrix, and configure the corresponding vibration total energy at octave points to electronically synthesize timbre. Through the Fourier analysis method oriented to the international standard sound octave point amplitude analysis, to obtain the vibration energy allocation ratio, based on Matlab to develop the spectral feature extraction function, run the code, then you can get the piano music frequency point specific amplitude, and then obtain the octave point two-dimensional matrix.

II. D.Creating a note characterization matrix

After extracting the note length, pitch, pitch velocity, and channel information of each note, a note feature matrix of the MIDI file can be constructed from them, and the process of extracting the note feature matrix is shown in Figure 1. First, read the MIDI file in binary format. As mentioned earlier, the header of the MIDI file is fixed, so it is determined that the file has been opened correctly. Then it is necessary to determine the number of tracks, which are included in the header block. For MIDI files that do not have a track number of 1, i.e. if there are multiple tracks, the first track generally records some global information in the form of Meta events, such as track name, instrument name, note velocity, beat information, etc., so we need to record this information before proceeding to the next step of extracting the basic notes. For a track number of 1, the track directly records the information of the fundamental note, so we can go directly to the extraction of the fundamental note.

In the extraction of basic notes, the track header is first judged to determine the starting position of the pointer traversal. The delta-time is calculated via the note feature matrix building process to determine the start time and end time of each note. After the delta-time is discarded, the rest of the content consists of MIDI events, which have a very fixed format, consisting of a status byte plus a number of data bytes. The status byte represents the actual operation, which is determined by a lookup table operation. The data bytes represent some basic information about the note, such as pitch, timbre, velocity, etc. After extracting this information, the data bytes will be used as the basis for the MIDI event. After extracting this information, the useful parts are written into the note feature matrix.

The format of the extracted note feature matrix mainly uses the information of the note's onset time, duration, channel number, pitch, velocity, etc. The onset time and duration of the note are both used in the matrix. Among them, the onset time and pitch length are in seconds.

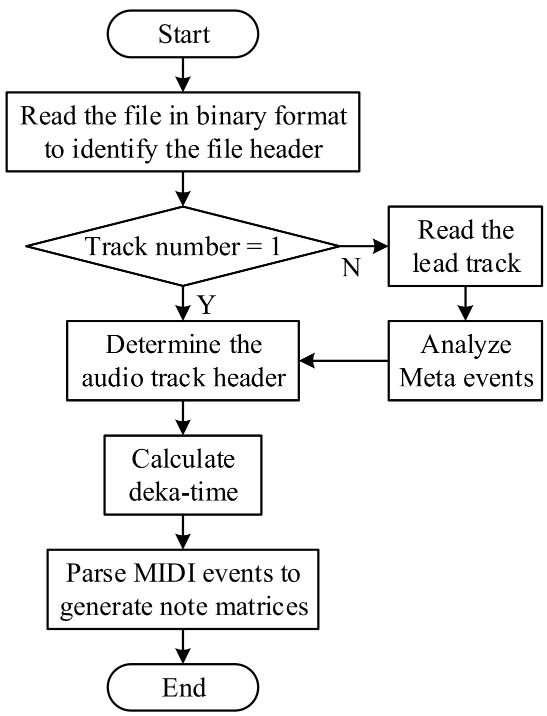


Figure 1: The establishment of the characteristic matrix of the notes

II. E.Tone Intensity Envelope Function

In order to electronically synthesize piano music, the envelope function needs to be extracted. The envelope function can present the temporal variation characteristics of the sound intensity, in order to extract the application of the envelope function, can ensure that the electronic synthesis of music conforms to the function change situation [22]. Through the international standard tone piano music amplitude characteristics waveform analysis, for the electronic synthesis of piano music, need to ensure that the time dimension of the vibration energy of the precise configuration to simulate the analysis, that is:

$$F(x) = \frac{x^2}{e^x}(x \geq 0) \tag{8}$$

This envelope function pianoforte vibrational curve matching directly contributes to the electronic synthesis of musical notes.

II. F. Musical sound simulation analysis

Based on the twelve mean rate, the base frequency calculation and analysis of musical notes are carried out, and the mathematical software is used to derive sinusoidal signals to illustrate the music, and the matrix signals are converted by the software function to realize the musical sound simulation. Because the music is very easy to appear fine noises when playing, it is necessary to innovate the envelope function to realize efficient and unadulterated articulation by its softness and vibration total speed effect, so as to eliminate the noises and guarantee the smoothness of the note transition. The tone description matrix can promote the piano music electronic synthesis, if the synthesized tone requirement is low, 18 octave point resonance articulation can be selected to equalize the synthesis speed and tone, and to meet the needs of piano electronic synthesis and music recognition.

III. Experimental results and analysis

III. A. Algorithm Validation

In order to verify the comprehensive performance of the feature extraction technique based on matrix operations such as timbre and notes proposed in this paper, this paper uses monophonic files as the experimental data, and all the experimental samples are selected from the McGill University Instrumental Timbre Sampling Library in the United States. It includes a lot of timbre information recorded in wav format, and 45.2 kHz is used as the sampling frequency of the experiment. The simulation environment for the experiment was Windows 8, 4 GB of RAM, and a 2.4 GHz Intel processor, and the specific information about the timbral characteristics of the piano arrangement included in the experiment is shown in Table 1.

Table 1: Piano tone

Feature number	Feature name	Characteristic dimension/dimension
1	Setting time	1
2	Time domain envelope segmentation ratio	2
3	Spectral mass	2
4	Spectral flux	1
5	Spectral roll drop	1

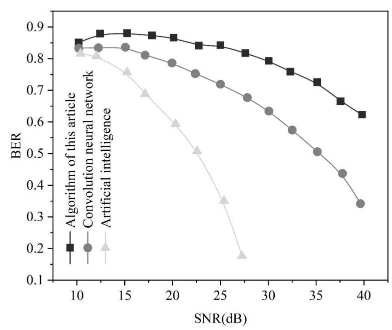
Table 2: Identification result

Serial number	Frequency/HZ	Detection time value/s	Exact time/s	Error / %
1	255.42	0.19	0.40	-1.23
2	288.24	0.23	0.40	-5.26
3	331.23	0.18	0.40	-0.73
4	346.81	0.24	0.40	-1.25
5	392.21	0.33	0.40	1.34
6	390.18	0.14	0.32	-1.12
7	385.23	0.18	0.32	0.25
8	291.47	0.22	0.32	-1.12
9	289.56	0.11	0.32	2.22
10	288.32	0.10	0.32	1.68
11	287.15	0.13	0.38	1.18
12	279.36	0.08	0.38	-0.48
13	274.21	0.05	0.38	1.45
14	285.36	0.13	0.38	0.58
15	286.73	0.16	0.38	2.22

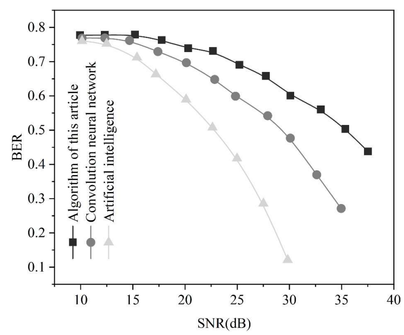
The frequencies, timbres and errors of the piano music obtained through timbre recognition are shown in Table 2. The time value is essentially the name of all notes in the piano music, which is the best way to separate the whole note, half note and quarter note.

Analyzing the table, it can be seen that the method of this paper can accurately identify the time value of all the notes in the piano music, and the time value of the whole note is 0.40 s, the time value of the half note and quarter note is 0.32 s and 0.38 s. By analyzing the time interval between the time value of each note, it can be seen that the biggest difference is between the time value of the whole note and half note, and the smaller difference is the time value between the whole note and quarter note. The largest difference is between the whole note and the quarter note, and the smaller difference is between the whole note and the quarter note. Analyzing the errors, it is clear that all the timings can be adjusted by controlling the timing errors within the range of (-5%, 5%).

BER is the bit error probability of a signal, and BER is used as an experimental index to compare the recognition performance of different recognition algorithms, and two different modulation modes are set up in the experiments, namely 16-QAM and 64-QAM. Under the two experimental conditions, the third-party software is used to output the BER results of different recognition algorithms. The experimental results are shown in Fig. 2. Comparing the experimental results in the figure, it can be seen that under different experimental conditions, the BER of the recognition algorithm proposed in this paper is lower, and when the three recognition algorithms are at the same BER level, the signal-to-noise ratio of the recognition algorithm proposed in this paper is smaller.



(a)16QAM



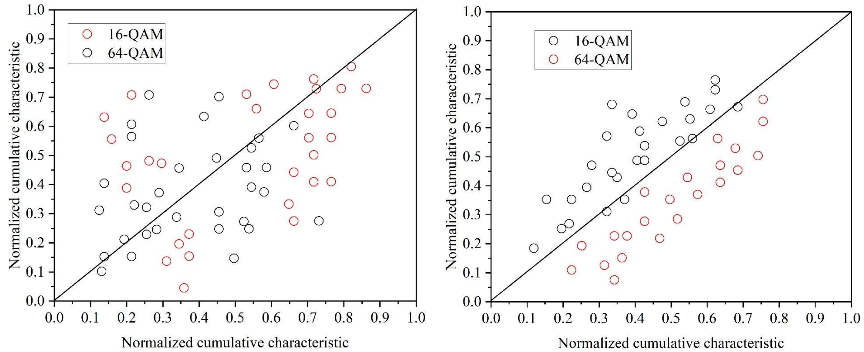
(b)64QAM

Figure 2: Experimental results of the audio feature dimension experiment

Automatic recognition of music is a newly emerging research project in recent years, which has an important research value in the field of music retrieval, and music recognition technology can help music beginners to learn music theory. For different recognition algorithms to carry out feature dimension compression level experiments, comparing the amount of features to be calculated for each recognition algorithm. Use the third-party software to output the experimental results, the specific results are shown in Figure 3. The signal sampling length and signal-to-noise ratio of each recognition algorithm in the experiment are uniform, and the experimental results show that the algorithm designed in this paper has a clear classification of feature volume, a more concentrated distribution, and good aggregation. The other two recognition algorithms have more scattered distribution of feature volume, larger degree of hybridization between feature volumes, no clear division between feature volumes, and more complex audio signal features, which require more computational processing time. Combining the above results with the results of BER experiments and note fundamental frequency calculation results, it can be seen that the recognition algorithm in this paper has a small computational cost, and the performance of the recognition algorithm is greatly improved with a very small amount of computation in exchange.

III. B. Analysis of the harmonic color and acoustic properties of the Prelude

Using the matrix feature extraction method to analyze the tonal color and acoustic characteristics of Debussy's piano area, it can be concluded that the prelude is a musical form that emerged from the 15th to 16th centuries, and was mostly improvised at first, with a length of no more than twenty bars. In the middle of the 17th century, when the suite was prevalent, the prelude became a “warm-up” before the opening of the suite, but it was not a part of the suite and was sometimes canceled. The Prelude is a single three-part work with a recapitulation, and the analysis of the form is shown in Table 3. And frequent off-key is also one of the characteristics of Debussy's compositions, in the “Prelude” of the A part of the use of dual-tonal compositional techniques, the high part of the melodic part of the mediaeval mode, while the low part of the F major, and then to the 9th measure of the low part briefly into the key of bB major, three measures later the tonality, however, shifted to the far-relative key of d minor, to the B part of the a Lydian mode and the e Ionian mode of the frequent conversion with an unexpected This shows that Debussy's use of double tonality and frequent modulation in the Prelude creates a dreamy effect. Through the special harmonic language and timbre processing, he creates a kind of hazy and ethereal sound effect, blurring the boundary between reality and fantasy.



(a) Convolution neural network (b) Recognition algorithm for this article

Figure 3: The audio signal characteristics of different recognition algorithms

Table 3: The harmonics color analysis of the prelude

	A	B	A'	Coda
Subsection	1-19	20-65	66-76	76-89
Modularity	F	a	F	F

The Minuet is a more complex pirouette, which is analyzed in Table 4. The term minuet, or menute, is derived from the French language and refers to a form of dance in which the movements and steps are delicate and short. The minuet flourished in the Baroque period, and the minuet is generally referred to as the minuet accompanied by instrumental music. In the 18th century, the minuet became a full-fledged piece of music, rather than just an accompaniment to the dance. The A part of the piece is 1-21 bars long and is set in the key of A minor. In the first three measures of the piece, the harmony is established on the fourth, first, and fifth steps. Usually, the beginning of a piece will use the main chord or the dominant chord to clarify the tonality, but the minuet uses the subordinate chord at the beginning, which is a refreshingly new chordal connection of IV-I-V, and the melodic voices appear in the inner voices, and the use of skipping notes and ornamental notes add to the wit of the melody.

Table 4: The harmony and color analysis of "the little step dance"

	A	B	A`	C	A`	D	Coda
Subsection	1-21	22-41	42-49	50-72	73-81	82-96	97-104
Modularity	a	bB-d-a	a	a	bE	#f	a

Moonlight is the only title piece in the suite and is one of the most widely known of Debussy's piano compositions. It is the most widely known of Debussy's piano compositions. This little piece is Debussy's piano work of the same name, composed under the influence of Weiland's poems. It has been said that “where language fails, music begins”. If we want to select the piano piece that makes people feel the most sense of picture, I believe that “Moonlight” must be on the list, because it uses simple musical language and delicate feelings to draw a picture of a mesmerizing and gentle moonlit night. The analysis of the composition is shown in Table 5, with bars 1-26 as the first part, and bars 1-14 can be divided into two phrases of 8+6 asymmetry, with the second phrase being a variation development of the previous phrase. At the beginning of the first bar, Debussy uses the inversion of the first chord of bD major, the superimposition of the second and third degree to show the melody, and the second degree is not resolved, the use of a large number of sustained tones makes the melody even more long, not only enriches the harmony but also weakens the chord's functional role. The long bass and the melodic voice form seventh chords. These seventh chord progressions are not treated according to the functional harmonic relationship, but are left unresolved, emphasizing the fluidity between the voices, as can be seen from the score, the chords are arranged in an open way, and this combination of arrangements makes the harmonic effect even more ethereal, ending in the 8th measure in an open genitive chord.

Table 5: The harmonic color analysis of moonlight

	A	B	A'	Coda
Subsection	1-26	27-50	51-65	66-72
Modularity	bD	bD-E	bD	bD

IV. The application of piano music simulation in digital music creation

The MATLAB language is used to build a rapid prototype to try to simulate the piano music. Firstly, we start from the simulation of the simplest sinusoidal musical note, then gradually increase the difficulty, and in the interaction, improve the timbre, and finally realize the goal of simulating the piano musical note.

Take the first stanza of "Moonlight" as an example, first calculate the fundamental frequency of each musical note in the stanza by using the twelve equal temperament law, then generate the sine signal with the relative value of vibration amplitude of 1 and sampling frequency of 8kHz in MATLAB to express these musical notes, and then use MATLAB's "sound" function to transform the music signal matrix into the vibration signal of the speaker, so as to realize the goal of simulating the piano music. MATLAB "sound" function can be used to transform the music signal matrix speaker vibration signal, thus realizing the simulation of the music.

By analogy, the frequency and duration of each musical note in the first measure are calculated as shown in Figure 4.

Then, according to the beat identification, it can be seen that this song has two beats per measure, from which the duration of each musical note is determined. The duration of one beat is 0.5 s. In MATLAB, the sampling frequency used to simulate the piece is set to fs = 8000 Hz, which means that 8000 data points are simulated in 1 s. The number of points occupied by a musical note indicates the duration of the note. With a matrix to store the electronic synthesis of music signals, in the MATLAB "sound" function can be played. The program plays the first bar of "Moonlight" with a perfect sine wave and generates a "dfhtys.wav" music file. The sine wave performance sounds very pure and soft, close to the sound of a blowjob, but without any overtones, the human ear can perceive the sound of the music as very different from the sound of a piano.

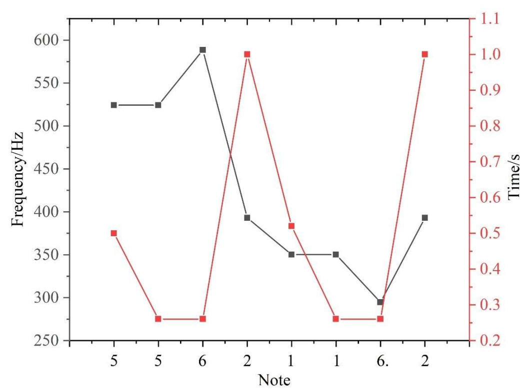


Figure 4: The first small frequency and beat of the eastern square red

V. Conclusion

This paper is based on the short-time Fourier variation spectrum analysis method, the establishment of timbre, note characteristic matrix and tone intensity change envelope curve, can clearly identify the piano music sound. It realizes the analysis of Debussy's tonal color and acoustic properties, and its application research in modern music creation. It is shown that the method in this paper can accurately recognize the time values of all notes in piano music with an error value of no more than 5%. Under the two experimental conditions of 16-QAM and 64-QAM, the recognition algorithm proposed in this paper has a lower BER, a smaller signal-to-noise ratio, a clear classification of the feature volume, a more concentrated distribution, and better aggregation. In the application of modern music creation, the Debussy piano music "Moonlight" short score is taken as an example, and it is simulated with perfect sine wave, which improves the timbre and is close to the sound of xiao, without any overtone component.

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