

DESIGN OF DIVERSIFIED VOCAL MUSIC TEACHING SYSTEM UNDER MULTI-OBJECTIVE MANAGEMENT

Wei Zhan

Department of Clinical Medicine, Jiangxi Medical College, Shangrao, 334000, China
zw18270156886@163.com

Abstract: Aiming at the problem that the vocal music high-frequency signal is masked by the low-frequency signal in the vocal music teaching system, a diversified vocal music teaching system under multi-objective management is designed. In the hardware part, synchronous buck conversion and synchronous dynamic storage are realized by designing power supply circuit, microprocessor and common functional modules. In the software part, considering the particularity of vocal music course, extract the characteristic parameters of vocal music signal, and identify the vocal music label to avoid the vocal music high-frequency signal being masked by low-frequency signal, so as to manage vocal music audio resources. On this basis, the software design is completed through the diversified design of five modules: basic information management, student music homework management, music practice management, online classroom management and information notification management. The function of the system is simple and safe in user operation. In terms of performance test, the average response time and throughput of the system are better than the vocal music teaching system based on Moodle platform and Web Service, and has strong application performance.

Key words: multi-objective management; vocal music class; diversified teaching; system design; teaching system; vocal label identification;

0 Introduction

In recent years, the first mock exam has changed people's work and life patterns. In teaching, people's knowledge acquisition is changing greatly. In particular, the development of Internet technology has enriched and diversified the original online education based on B/S, and people gradually accept the mode of completing various learning tasks through the network [1]. The diversified teaching system of vocal music course makes full use of the existing mobile Internet technology and network technology, combined with the characteristics of music major, and constructs a comprehensive teaching platform integrating basic information management, student music homework management, music practice management, online classroom management and information notification management on the computer platform [2]. Through the teaching platform, it provides professional support for the teaching and learning of music major, and provides great convenience for distance learners through mobile network. It is mainly reflected in that teaching and learning can provide the recipient with learning content, learning process and learning methods. This diversified learning mode not only meets the normal classroom learning, but also completes the practice or examination operation for learners through the platform, making the learning effect better [3]. The online teaching and learning platform under multi-objective management makes full use of the convenient and fast advantages of the current mobile Internet,

and embodies the advantages of completing learning tasks anytime and anywhere through the mobile terminal. It is easy to carry, can make full use of learners' spare time to complete learning and practice, and ensure the learning effect. The system fully moves the traditional education mode into the network platform, such as student homework, learning practice, online classroom and teaching notice, and provides a traditional and efficient learning and teaching mode [4]. Moreover, music teaching has strong professional characteristics, such as the need to identify sound through notes. The system makes full use of the multimedia technology of mobile platform, provides simple interface design, and completes the teaching system in line with music specialty [5]. This paper designs the diversified teaching system of vocal music course under multi-objective management, promotes music teaching, combines the traditional music teaching mode with modern technology, provides reference and promotes music education modernization.

1 Hardware design of diversified vocal music teaching system under multi-objective management

The hardware part of this system design mainly includes power circuit part, microprocessor part, common function module and peripheral interface part, as shown in Figure 1. The microprocessor part of vocal music teaching system also includes three parts: random access memory and read-only memory. The operation of program and data processing are completed by this part. The function module and peripheral interface are composed of wireless connection module, audio codec module, PCIe interface and HDMI interface. The following is a detailed description.

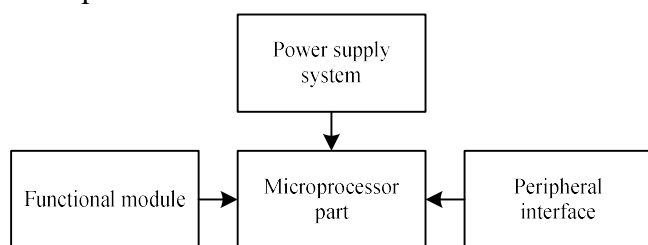


Figure 1 System hardware design block diagram

In embedded system, power supply reliability is an important foundation. The success or failure of the circuit system needs to be determined. Voltage fluctuation will lead to system failure and even damage hardware. At the same time, because each functional module and interface circuit often have different voltage requirements, the power module must also have multi voltage output capability. The power requirements of the whole system are shown in Table 1.

Table 1 Power requirements of the system

| Input voltage / current | Output voltage / current | Supply object |
|-------------------------|--------------------------|---------------|
| 5V/5A | 1.8V/2.5A | VDD_ARM_IN |
| | 1.8V/2.5A | VDD_SOC_IN |
| | 1.5V/1.6A | VDD_MEM |
| | 3.3V/150mA | VCC_EMMC |
| | 1.8V/100mA | VDD_COMBO |
| | 3.3V/100mA | VDD_AU |

Due to the high requirements for various indicators, a special power conversion control chip is often used as the core of the power module, which not only improves the performance of the module, but also increases the system integration and reduces the risk. According to the voltage requirements of processor, peripheral interface and module, WM8326 chip is selected as the power management subsystem. Because the system input voltage is +12V and the working voltage of WM8326 chip is +5V, MP1499 is selected as the synchronous step-down converter chip. The architecture of the power module is shown in Figure 2.

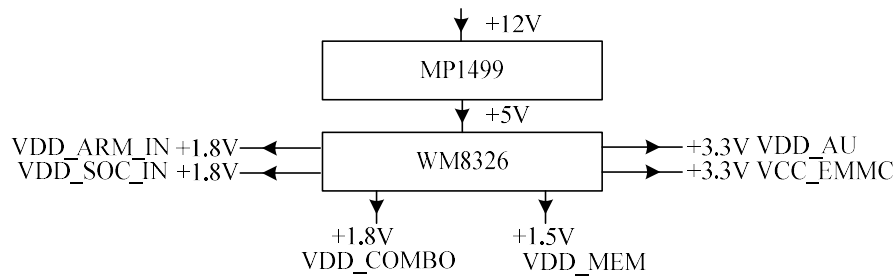


Figure 2 Power module architecture

MP1499 is a high frequency synchronous rectifier buck switching mode converter. After a wide range of input voltage reaches 5A peak output current, it can be adjusted linearly. It has higher efficiency in the output current load range. The current model can provide a fast transient response and cycle stability. At the same time, it also has the comprehensive characteristics of overcurrent protection and thermal shutdown protection. WM8326 single chip can solve the power management requirements of most mobile devices. Therefore, compared with multi chip solution, wm8326 has great cost advantages on the basis of ensuring performance. It is specially applied to a series of portable consumer products with low power consumption requirements. At the same time, it is also applicable to any multimedia application processor. As the brain of the whole hardware system, microprocessor module controls and dominates the operation of the whole hardware system. i.MX6 is selected as the microprocessor in this paper. In addition to its powerful ARM Cortex A9 kernel, i.MX6 also has rich on-chip resources. The i.MX6 has a dynamic random access memory controller, which can support the current mainstream memory chips, such as synchronous dynamic random access memory. Multiple linear regulated power converters are integrated in i.MX6 to provide on-chip resource voltage, which can reduce the design of power supply part of the whole hardware system. i.MX6 has many of the most important SDIO interfaces of embedded systems recently. It can not only support SD card, NAND flash, EMMC and other read-only memory, but also support many wireless connection devices, such as wireless network card, Bluetooth, camera module, GSM / GPRS module and so on, which greatly enriches the scalability of the hardware system. In the functional module, mt6620 device is selected to highly integrate WiFi, BT, FM and GPS into a single chip. Advanced hardware architecture and strong radio coexistence algorithm make MT6620 have great advantages in wireless connection applications. At the same time, due to the high integration of single chip, the package size and power consumption of this module are greatly reduced. Therefore, the area of printed circuit board

is reduced and the design resources are greatly optimized. The module block diagram is shown in Figure 3.

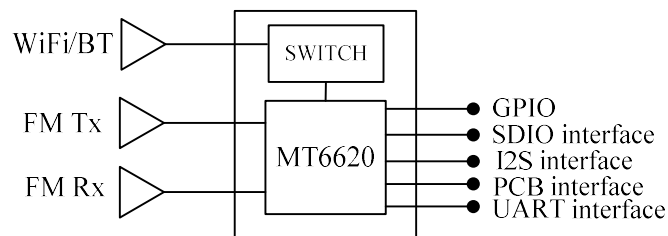


Figure 3 Block diagram of MT6620 module

The audio decoding chip used in this system design is WM8524, which is a Stereo DAC with internal charge and hardware control interface. The chip adopts unipolar 3.3V power supply to provide 2Vrms linear drive output. At the same time, it has a grounded reference output and uses DC servo, which eliminates the linear drive coupling capacitance and effectively saves the power supply. The chip can be controlled and configured through the hardware control interface, and can support the common audio sampling rates of 8KHz and 192KHz. Therefore, the hardware design of diversified teaching system for vocal music course is completed.

2 Software design of diversified vocal music teaching system under multi-objective management

2.1 Extracting characteristic parameters of vocal music signal

Before the research of system modeling and simulation, the system software is divided into two parts. The corresponding functions are vocal music target management, label recognition and diversified teaching management. The vocal label recognition subsystem mainly consists of three parts: feature extraction, template library and pattern matching. The system extracts the characteristic parameters of the test signal, and obtains the template with the highest matching degree with the test signal in the template library through the matching algorithm. The vocal label corresponding to the template is the recognition result of the system. In pattern recognition, the original data of vocal signal is not only huge, but also the characteristics are not obvious enough, so it is difficult to directly apply it to the system for pattern recognition [6]. The essence of feature extraction is to extract a small amount of representative data that can characterize the original signal from a large number of parameterized sample data. For the audio signal, the ideal feature parameters need to meet two conditions: one is to be able to well reflect the vocal tract characteristics of the speaker and the human auditory model. Second, there are great differences among the orders of characteristic parameters, that is to say, the characteristic data representing the signal has better data independence. Audio signal processing is usually carried out in time domain and frequency domain, but the noise interference in audio is usually an irresistible factor, and the time domain parameters are greatly affected by noise [7]. Therefore, when the system selects the characteristic parameters as the recognition basis, the frequency domain characteristic parameters with strong anti noise performance are selected. In this paper, MFCC is used to extract the characteristic parameters. MFCC is a linear cosine transform based on logarithmic power spectrum. Mel scale is closer to the response of human auditory system. This frequency distortion can better represent sound. When the music signal is transmitted by traveling wave on the basement

membrane of the inner ear cochlea, due to the low frequency and long wavelength of the low-frequency signal, its transmission distance is greater than that of the high-frequency signal, resulting in the phenomenon that the high-frequency signal is masked by the low-frequency signal [8]. And for different frequencies, the masking ability of higher frequency sound is also different. The higher the audio, the greater the masking ability. Therefore, the human auditory system is equivalent to a filtering system to filter the treble [9]. In the design and implementation, a group of band-pass filters can be designed, which are arranged from dense to sparse according to the masking ability of each frequency point based on the auditory characteristics of human ears. The conversion relationship between linear frequency and Mel frequency is as follows:

$$f_1 = \alpha \log \left(1 + \frac{f_2}{\beta} \right) \quad (1)$$

In formula (1), f_1 and f_2 represent Mel frequency and linear frequency; α and β represent conversion parameters, and the values in this paper are 2595 and 700. After nonlinear transformation, it can be considered that the auditory sensory ability of human ears to sound is linear in Mel frequency domain. The extraction process of MFCC parameters is as follows: FFT transform the vocal signal to obtain the linear spectrum, and the calculation formula is as follows:

$$z(a) = \sum_0^{s-1} z(x) e^{-2\pi x a / s} \quad (2)$$

In formula (2), z represents linear spectrum; a indicates frequency; s represents the number of points of Fourier transform; x represents the original signal; e represents the natural logarithm. The data is mapped to mel scale through triangular filter bank. Any type of independent filter in Mel frequency filter bank can be selected. Considering the full utilization of the boundary information of each frequency band, triangular filter is used to form Mel frequency filter bank. The transfer function of the filter is defined as follows:

$$g(a) = \frac{s}{h_0} \beta e^{(f_2 - a) / \alpha} \left[f_1 + \frac{u(f_1^{\max} - f_1^{\min})}{u + 1} \right] \quad (3)$$

In formula (3), $g(a)$ represents the transfer function; h_0 represents sampling frequency; u represents the number of filters; f_1^{\max} and f_1^{\min} represent the highest and lowest frequencies allowed to pass through the passband of the triangular bandpass filter. The logarithmic energy of each triangular filter bank is calculated, and the MFCC is obtained by discrete cosine transform.

$$\lambda(c) = \sum_0^{s-1} W(u) \cos \left[\frac{\pi c(u - 0.5)}{u} \right] \quad (4)$$

In formula (4), $\lambda(c)$ represents the characteristic parameter; c is the dimension of characteristic parameters; $W(u)$ represents the logarithmic energy output by the triangular filter bank. Because the Mel frequency cepstrum coefficient not only reflects the human auditory effect, but also does not have any assumptions and restrictions on the input signal, it has better robustness.

In this paper, the dimension of MFCC characteristic parameters is set to 12 orders, and 24 orders of characteristic parameters are generated after difference operation.

2.2 Design vocal label recognition algorithm

After feature extraction, the template information database is established and the feature parameters of the audio to be tested are extracted, the best matching data template needs to be obtained through the recognition algorithm, and the original signal represented by it is the recognition result. Based on the realization of single note recognition, this paper realizes multi note recognition. Firstly, continuous notes should be divided into independent single note sequences by note time value segmentation algorithm. Then the dynamic time warping algorithm is used to recognize single notes, and finally the recognition results are combined and output. According to the results of multi note recognition, a vocal label is established. The process is shown in Figure 4.

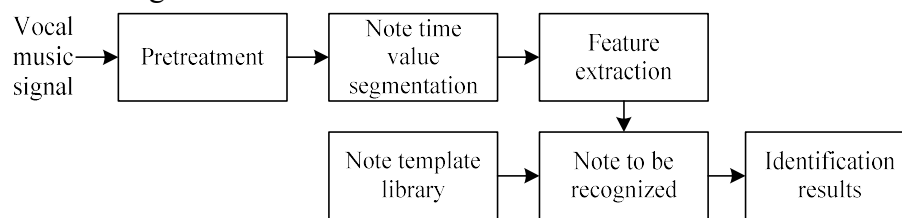


Figure 4 Vocal label identification process

Note segmentation is to segment continuous piano notes into single notes with different lengths in time domain according to the change of certain characteristics. Note segmentation is the basis of music score recognition. The effect of effective speech segmentation directly affects the result of music score recognition [10]. There are many note segmentation methods. Different segmentation methods need to extract different feature parameters and corresponding principles. The short-time average zero crossing rate and short-time energy can also be used to realize note segmentation, but the time-domain parameters of the signal are often affected by the signal-to-noise ratio, and the performance decreases sharply at low signal-to-noise ratio, so it is not suitable to use [11]. The note segmentation algorithm based on energy entropy ratio is proposed on the basis of spectral entropy. If the energy does not fluctuate greatly in the distribution of each frequency band, the signal corresponding to this frequency band contains a large amount of information and the entropy of the corresponding signal is also large. Therefore, the instability of the signal can be detected by information entropy to obtain the correct note segmentation point in continuous notes [12]. However, when using entropy to segment directly, there will be the problem of large audio energy but small spectral entropy. In order to solve this problem, the energy entropy ratio is introduced, that is, the ratio of short-time energy to entropy of each frame signal. The specific implementation steps of note segmentation method based on energy entropy ratio are as follows: calculate the spectral entropy of each frame, and it is as follows:

$$R(i) = \frac{Q(i)}{\sum_i^{M/2} Q(i)} \quad (5)$$

In formula (5), $R(i)$ represents the spectral entropy of each frame; i represents the number of frames; $Q(i)$ represents spectral density; M indicates the frame length of the audio to be tested. Then, the energy entropy ratio is calculated, and the formula can be expressed as:

$$E(i) = \sqrt{1 + \left| \frac{J(i)}{R(i)} \right|} \quad (6)$$

In formula (6), $E(i)$ represents the energy entropy ratio; $J(i)$ represents the energy spectrum. A predetermined value is set with each trough point in the energy entropy ratio as the reference point. After the trough point moves left and right on the time axis, the energy entropy ratio of the corresponding point is compared with the predetermined value, the start and end points of notes are defined, and the note segmentation is completed. Then the dynamic time warping DTW algorithm is used to realize note recognition. It is a more robust distance measure in time series. It allows similar shape matching even if the signals are in different phases on the time axis. Due to the great randomness of pronunciation body vibration, the length of pronunciation time can not be well controlled. If the linear uniform expansion method is adopted to align the frame length of the test file and the template file, the time length transformation of each small segment in the audio file under different conditions will be ignored, resulting in low recognition rate [13]. Therefore, when doing pattern matching, it is necessary to nonlinear bend the time axis of the audio signal, which solves the problem of unequal frame length between the signal to be tested and the template signal caused by different pronunciation length. Suppose that the template sequence contains A frame vocal signal and the audio sequence to be tested contains B frame vocal signal. The matching process of two vocal signals can be considered as finding the best path from point [1,1] to point [A, B]. The path can be expressed as:

$$\mathcal{G}(i) = \min \sum_1^A d[\omega(i), \nu(i)] \quad (7)$$

In formula (7), $\mathcal{G}(i)$ represents the matching relationship; d represents Euclidean distance; $\omega(i)$ and $\nu(i)$ represent the vocal audio parameters to be tested and the vocal audio parameters of the template. When solving the corresponding relationship between the audio to be identified and the template audio, using dynamic programming will divide the complex global problem into many local problems, obtain the optimal solution of the local problem, and finally linearly combine them to obtain the final solution. The optimal solution can be expressed as:

$$d[\omega(i), \nu(i)] = \sum_b^b (\kappa_b - \gamma_b)^2 \quad (8)$$

In formula (8), κ_b represents the data in the vocal audio parameters to be measured; γ_b represents the data in the audio parameters of the template; b represents the number of parameters per frame. On the basis of single note recognition, the recognition result of multiple notes is obtained, that is, the target management label of vocal music curriculum resources is obtained.

2.3 Set up diversified teaching management modules

According to the vocal music label under multi-objective management, a diversified teaching management module is established. The diversified teaching module can be further divided into basic information, student vocal music homework, vocal music practice, online classroom and information notification management module. User functions include student, teacher and other user management. Student functions include student registration, student information review, student information modification and student information deletion. When modifying student information, students modify it online, and teachers complete the review online. After students graduate, teachers delete the student information in the system ^[14]. Student information includes student number, name, gender, date of birth, nationality, political outlook, major, home address, health status, accommodation, contact information, parent name, contact information, etc. Teacher management: complete the input, modification and deletion of teacher information, and provide batch data import function ^[15]. Other user management: including other user information entry, information modification and deletion. The vocal music job management module includes music job query, vocal music job upload, vocal music job maintenance and other related operations ^[16]. This module designs resource class and resource file class, which are used to query, add, save and delete vocal music homework. Vocal music homework includes audio, video, documents and other related resources. Vocal music practice is used to realize the process that students complete music practice through mobile phones. The practice questions make sound through mobile phones to reflect the practice effect, including accompaniment practice, test question practice and famous works ^[17]. Therefore, it designs students' basic operation class, students' practice class and practice score management class, as well as extended score introduction class and so on. Online classroom includes online classroom, learning log, learning reflection, online questioning, resource browsing, etc. Online classroom is that learners can log in to the system for learning detection at any time after formulating relevant target performance. In online classroom, they can use the learning tools under this module to record notes, feelings and problems for future discussion and reference ^[18]. After completing the learning task, the system will automatically record the learning log for the calculation of later assessment results ^[19]. This module designs online class, learning log, learning reflection record and student information management. When the online class needs music homework, start self-learning by obtaining the latest music homework from the resources class, and record the learning process (start and end time) through study logs. In the vocal music teaching system, teachers can complete the dynamic release of teaching information and notices through this platform, and push them to the mobile terminal for each student by means of push ^[20]. Information notification includes educational information and daily notification, including the addition, modification and deletion of basic information, as well as the uploading and browsing of corresponding documents. These two operations are the same, so you need to design a parent class to complete the above operations in class design. For individual classes, such as information, including vocational education information, campus information and class information, they are identified by category attributes. The designed information table is shown in Table 2.

Table 2 Consultation information

| Physical name | Type | PK/FK | Chinese interpretation |
|---------------|--------------|-------|------------------------|
| BH | String (20) | PK | Information number |
| BT | String(50) | PK | Information title |
| LB | String(10) | PK | Information category |
| SJ | datetime | PK | Release time |
| FB | String(20) | FK | Publisher |
| NR | String(2000) | FK | Information content |
| ZL | integer | FK | Number of visitors |
| BJ | String(100) | FK | Browse class |

Through the diversified design of five modules: system basic information management, student music homework, music practice, online classroom and information notification, the design of system software is completed. Integrating hardware and software, the diversified teaching system of vocal music course under multi-objective management is designed.

3 System test

3.1 System function test

This paper designs a diversified vocal music teaching system under multi-objective management. In order to prove the application effect of the system, the system is tested below. Firstly, test the realized functions through test cases, click the function menu and enter relevant data and conditions to test. The data maintenance function tests whether it can be saved normally by inputting data. The data query and statistics function completes the statistics by selecting the setting conditions, and tests whether the statistical results meet the preset conditions. The function test mainly describes whether the test function operates normally through the test case. The test finds that there are situations that cannot be handled in the system function and system errors. The test case tests the design function, including function points, test results and conclusions, etc. The test of online classroom module shows that both students and teachers can complete online teaching and learning tasks through this corner, and the operation is stable. Through the function test, we can get the following conclusions: when the system data is added, there is a corresponding integrity verification operation on the client without passing through the server. This inspection is completed at the client, and the corresponding data type verification is given, which eliminates the unnecessary entry of invalid data and dirty data, ensures the data quality, and can be saved to the database after entering the complete data. When the system deletes, the system has corresponding confirmation operations to reduce the user's accidental deletion. After the test, the data query operation can perform fuzzy matching according to the keywords entered by the user and display them in the form of list. The test results meet the expected query objectives. Therefore, the system function has the characteristics of simplicity and safety in user operation. Based on the functional test results, further system performance test is carried out.

3.2 System performance test

The system performance test is implemented through LoadRunner 9.5. The system response time and system throughput are tested by executing query operations at 200 concurrent users at the same time. To prove the superiority of the system performance, this paper compares it with the vocal music teaching system based on Moodle platform and Web Service. The test results of system response time and throughput are shown in Table 3 and Table 4.

Table 3 Comparison of system response time (s)

| Number of tests | The system of this paper | System based on Moodle platform | System based on Web Service |
|-----------------|--------------------------|---------------------------------|-----------------------------|
| 1 | 2.14 | 4.21 | 5.14 |
| 2 | 1.46 | 5.65 | 4.76 |
| 3 | 2.52 | 4.54 | 5.52 |
| 4 | 1.86 | 6.42 | 6.25 |
| 5 | 2.23 | 5.86 | 5.61 |

Table 4 Comparison of system throughput (byte/s)

| Number of tests | The system of this paper | System based on Moodle platform | System based on Web Service |
|-----------------|--------------------------|---------------------------------|-----------------------------|
| 1 | 18264.25 | 15060.12 | 14105.17 |
| 2 | 20226.48 | 14224.14 | 13081.68 |
| 3 | 21615.20 | 15507.16 | 12460.36 |
| 4 | 20549.54 | 13618.58 | 12524.34 |
| 5 | 19822.62 | 13242.26 | 13230.21 |

According to the above test, the average response time and throughput of each system can be obtained under the concurrent state of 200 users. According to Table 3, the average response time of the system in this paper is 2.04s, which is 3.30s and 3.42s less than that of the vocal music teaching system based on Moodle platform and web service. According to Table 4, the average throughput of the system in this paper is 20095.62 bytes/s, which is 5765.17 bytes/s and 7015.27 bytes/s more than the vocal music teaching system based on Moodle platform and web service. Based on the above performance test results, the actual system can meet the expected design objectives. Considering some client hardware configurations, it can meet the performance requirements of the actual operation within 1 second. Therefore, the designed vocal music teaching system meets the requirements of function and performance, and has certain practicability.

4 Conclusion

This paper integrates the characteristic teaching of vocal music specialty into the design and implementation of the system, and designs a set of more effective methods and models of music teaching and learning system. The follow-up work will continue to be deeply studied from the following three aspects. Instructional design is a complex process. Different disciplines have different instructional design strategies and methods. We should further optimize the key links in the design and deeply integrate with the concept of performance, so as to better reflect the teaching law and improve learning effect. The software design method has the advantages of strong reusability and high reliability. In the design process of the system, the research on the software

design pattern should be further strengthened, and the interface and aggregation degree of components should be further studied to improve the performance and security of the system. It is proposed to introduce the online examination system to check the learning effect, and take the examination score as the evaluation index of performance learning.

References

- [1] Xu Nan, Fan Wen-hui. (2020). Research on Interactive Augmented Reality Teaching System for Numerical Optimization Teaching. *Computer Simulation*, 37(11), 203-206,298.
- [2] Onwubiko, S. G., & Calilhanna, A. . (2020). Interdisciplinary physical music: a blind spot in education on acoustics. *The Journal of the Acoustical Society of America*, 148(4), 2697-2697.
- [3] Chen Jin-yin, Wang Zhen, Chen Jin-yu, Chen Zhi-qing, & Zhen Hai-bin. (2019). Design and Research on Intelligent Teaching System Based on Deep Learning. *Computer Science*, 46(z1), 550-554,576.
- [4] Albashtawi, A. H., & Bataineh, K. . (2020). The effectiveness of google classroom among efl students in jordan: an innovative teaching and learning online platform. *International Journal of Emerging Technologies in Learning (iJET)*, 15(11), 78-89.
- [5] Paek, H., Siebein, G. W., Roa, M., Miller, J. R., & Vetterick, M. . (2019). The evolution and creation of schools for music education. *The Journal of the Acoustical Society of America*, 145(3), 1739-1739.
- [6] Jaeger, H. . (2020). An acoustic impedance probe for the teaching of musical acoustics to non-majors. *The Journal of the Acoustical Society of America*, 148(4), 2563-2563.
- [7] Neilsen, T. B., Vongsawad, C. T., & Onwubiko, S. G. . (2020). Teaching musical acoustics with simple models and active learning. *The Journal of the Acoustical Society of America*, 148(4), 2528-2528.
- [8] Cooper, R. A., & Holden, M. . (2019). The acoustic design of higher education music rehearsal spaces. *The Journal of the Acoustical Society of America*, 146(4), 2851-2851.
- [9] HE Hongyan. (2020). Research on Vocal Music Performance Evaluation System Based on Neural Network and Its Application in Piano Teaching. *Microcomputer Applications*, 36(8), 125-128.
- [10] Abboud, R., & Tekli, J. . (2020). Integration of nonparametric fuzzy classification with an evolutionary-developmental framework to perform music sentiment-based analysis and composition. *Soft Computing*, 24(13), 9875-9925.
- [11] Linz, J. A. . (2020). Atom music: acoustical realizations of the atomic world through sonification. *The Journal of the Acoustical Society of America*, 148(4), 2749-2749.
- [12] Peretz, I., Ayotte, J., Zatorre, R. J., Mehler, J., Ahad, P., & Penhune, V. B., et al. (2020). Effects of vocal training in a musicophile with congenital amusia. *Neuron*, 33(2), 0-191.
- [13] Kim, J. H., & Larson, C. R. . (2019). Modulation of auditory-vocal feedback control due to planned changes in voice f o. *The Journal of the Acoustical Society of America*, 145(3), 1482-1492.

- [14] Ma Duarte-García, & Sigal-Sefchovich, J. R. . (2019). Working with electroacoustic music in rural communities: the use of an interactive music system in the creative process in primary and secondary school education. *Organised Sound*, 24(3), 228-239.
- [15] Ma Duarte-García, Wilde, E., Cortez, R., & Sigal, J. R. . (2020). The use of an interactive music system as an aid for exploring sound in music education in a rural area. *Revista Música*, 20(1), 357-380.
- [16] Kim, H. G., Kim, G. Y., & Kim, J. Y. . (2019). Music recommendation system using human activity recognition from accelerometer data. *IEEE Transactions on Consumer Electronics*, 65(3), 349-358.
- [17] Zhu Qing. (2020). Design of information teaching management system integrating association rule mining algorithm. *Modern Electronics Technique*, 43(23), 159-163.
- [18] Cao Lianjiang. (2020). Interactive computer aided teaching system based on .NET platform. *Modern Electronics Technique*, 43(3), 134-137,141.
- [19] Kariapper, R., Samsudeen, S. N., & Fathima, S. . (2020). Quantifying the impact of online educational system in teaching and learning environment among the teachers and students. *Solid State Technology*, 63(6), 12118-12132.
- [20] Kanth, R., Skoen, J. P., Toppinen, A., Lehtomaeki, K., Laakso, M. J., & Heikkonen, J. . (2019). Innovative and efficient teaching methodology for digital communication systems using an e-learning platform. *Journal of Communications*, 14(8), 689-695.