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Research Article

Application Design of Linguistics in Computer Technology under Artificial Intelligence Background

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Computer technology is called the fourth industrial revolution, indicating the importance of computer technology. At the same time, as the most important way for our daily exchanges, it is also connected to computer technology. This article is intended to study the application design of linguistics in computer technology in controversy. This paper proposes a documentary conversion algorithm based on a BP neural network and has made corresponding analysis for linguistics. It designed a documentary conversion system for the application of language technology. The experimental results show that the language has more in-depth applications in computer technology, and the documentary conversion system designed herein can reach more than 97% of the accuracy of 20 people in the system.

1. Introduction

Since the introduction of systematic linguistics in the 1980s, systematic functional linguistics have been greatly developed in China. Students studying system functional linguists have introduced research center from previously pure communication to the point of view, which achieves theoretical partial innovation. Then put it to the application of the theory of Chinese, such as the discourse analysis of Chinese or the discourse comparative analysis between Chinese and English. The content of computer technology is very broad, which can be roughly divided into computer system technology, computer device technology, computer component technology, and computer assembly technology. Computer technology includes the basic principle of computing methods and computing unit design, instruction system, central processing unit (CPU) design, the principle of pipeline and its application in CPU design, storage system, bus, and input and output.

Language is a carrier of culture, the culture of the nation is often inherited by language, and the national language has an extremely close relationship with national culture. And computer technology can convert the text language in linguistics into digital codes for storage. Ethnic minorities learn Chinese, because in the multi-unified Chinese nation, Chinese learning is the premise of their integration into the Chinese nation. When it is integrated into the main body of the Chinese national culture, they can also make their own cultures have become part of Chinese culture, and they will continue to benefit common interests, joint progress, and common development. The main task of bilingual education in China is to learn Chinese on the basis of mastering the mother tongue, thus relying on the modernization of modern scientific cultural technology, participate in modern life, and realize the modernization of national culture.

This article has the following innovation: (1) in the algorithm module, the document system uses a better algorithm for data selection. The algorithm is selected in mode switching. (2) The corresponding length time of each radio station reserve has a total of 22 samples. The update of the subsequent material library and the mating upgrade of the sound module will increase the sound result diversity of the document system. (3) It adds more personalized parameter settings in sound playback and processing modules, such as sound image control. And it places a new parameter adjustment knob in the user operating interface for users to have more control selection when using.

2. Related Work

Language is the basis of communication, and there are many researches on language. Johann-Mattis et al. believe that the amount of data from all local languages is rapidly increasing the method of linguistics, which needs to face the challenges of these data. He tested the potential of the automatic method to detect the word source (homologous words) in cross-language data. The results show that the automatic method can identify homologues with very high accuracy, and the best performance of the INFOMAP reaches 89% [1]. Mccarthy has studied grammatical relations in linguistics and argues that preference theory is a general pattern of grammatical structures. He investigated the motivation of optimization theory, and its core principles and analysis of the analysis [2]. Stab and Gurevych proposed the first end-to-end method to parse the argument structure in the persuasive articles. He used two corpus assessments of the combined model that his method not only greatly improves the identification of the type and the relationship between the components and the main relationship, and is significantly better than the challenging heuristic baseline [3]. Isaacs and Harding believe that after the long-term marginal position, the second language (L2) pronunciation field has achieved many progresses in the past decade, which makes it possible to improve the activity and visibility of the subsequent study in phraseology [4]. Murasugi studied the grammar pronunciation in Japanese, and he proposed a uninterrupted rule to explain this fact. Based on the empirical evidence of the pure composite noun phrase structure, Japanese children inferred all prename modifications; especially, the relationship clauses are noun phrases. Therefore, IP hypothesis has been supported by syntactics, subjectivity, and acquisition research [5]. Kuroda will put and discuss such a point of view; in addition to this common relationship type, Japanese also has a relationship clause. This relationship is recently increasingly attracting the attention of linguists, often referred to as "headless relationship" [6]. However, comprehensive analysis of the current research results of the academic circles, we find that there is still a more theoretical discussion, and the actual analysis is less; there are much more analyses of English discourse, and there is less analysis of Chinese language; there are much more analyses of English discourse, and there is less analysis of Chinese language; there are many different discourse analyses, and there are few word comparisons, etc. And this small part of research results is compared to foreign discourse (mainly English language) and rarely carries out language comparisons within Chinese.

3. Linguistics in the Context of Artificial Intelligence

3.1. Linguistics under Artificial Intelligence

3.1.1. Language Is a Social Norm. Language is a sound (image) instruction with a unified coding and decoding standard developed by the same biological species due to the need for communication. Substantial definition language is the sound/symbol as the material shell and the meaning as the connotation, which is composed of vocabulary and grammar and can express the instruction system of human thought. Language is actually a social convention, and most scholars publicly emphasize its correctness. Language is a social convention; in fact, it contains two aspects: one is social, and the other is regularity. Some scholars believe that language should be a complete and fixed whole. So how to fix it? This requires specific rules, so the language must follow certain rules [7, 8].

3.1.2. Language Is Homogeneous. Language is a way of communication between people, and people cannot communicate with each other without language. Although people's thoughts can be conveyed through pictures, actions, expressions, etc., language is the most important and the most convenient medium. However, people around the world speak different languages, and it is difficult, if not impossible, to talk directly to each other. Taking Russian as an example, it can be seen that the "language" should be homogeneous. The Russian language is diverse, so the language is heterogeneous. And language cannot be used to refer to these heterogeneous speeches; in other words, it is the opposite of the heterogeneity of speech, so it is homogeneous. Moreover, although in the actual speech communication process, the speaker will be affected by a variety of factors, the homogeneity of language is still reflected in the specific implementation process [9, 10].

Other scholars believe that speech activities are mixed, and language is a very definite object in this mixed totality. Not only in different regions, there are different languages and dialects; that is, in the same region, different social classes and people of different ages will have special vocabulary to express their unique feelings. This makes it difficult for people of another class or age to understand. Judging from the classification of research, the confusion of speech activity is mainly caused by its inclusion of "speech," because "speech" is individual, it has many uncertain factors, and it is the sum of many special circumstances. This is the heterogeneity of speech activity [11, 12].

The production of language means that people speak or write the thoughts to be expressed through the activities of language organs or hands. It includes two forms of speaking and writing. The main units of language production are phonemes, syllables, morphemes, words, phrases, and sentences. As shown in Figure 1, the highly developed computer technology and artificial intelligence have changed traditional linguistics. Whether from the classroom or from social media, the languages people are exposed to are becoming more diverse and smarter.

3.2. BP Neural Network Algorithm

3.2.1. Node Design. BP (back propagation) neural network is a multi-layer feedforward neural network trained according to the error back propagation algorithm, and it is one of the most widely used neural network models.

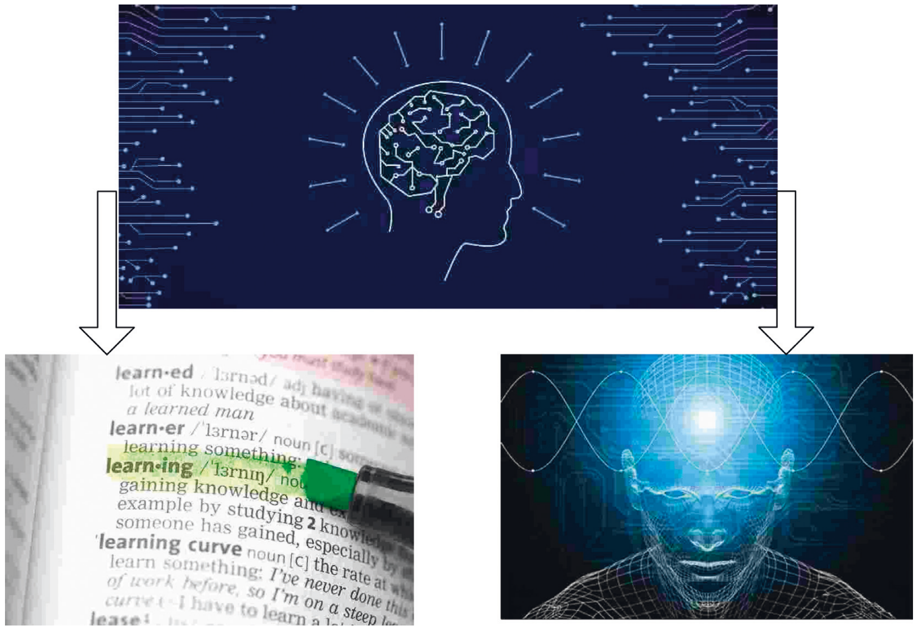

Figure 1: Combination of artificial intelligence and linguistics.

Artificial neural network does not need to determine the mathematical equation of the mapping relationship between input and output in advance. It only learns certain rules through its own training and gets the result closest to the expected output value given the input value. As an intelligent information processing system, the core of artificial neural network to realize its function is algorithm. BP neural network is a multi-layer feedforward network trained by error back propagation (referred to as error back propagation), and its algorithm is called BP algorithm. Its basic idea is the gradient descent method, which uses the gradient search technology to minimize the error mean square error between the actual output value and the expected output value of the network.

BP neural network has arbitrarily complex pattern classification ability and excellent multi-dimensional function mapping ability, and solves the exclusive or (XOR) and some other problems that cannot be solved by simple perceptrons. Structurally, the BP network has an input layer, a hidden layer, and an output layer; in essence, the BP algorithm takes the square of the network error as the objective function and uses the gradient descent method to calculate the minimum value of the objective function.

As shown in Figure 2, input layer node design is determined by the number of risk evaluation indicators. By performing principal component processing on the original data of the sample, the input layer nodes are affected by the data after the principal component processing of the original data, and the specific value should be the number of principal components after the principal component processing [13].

Output layer node design: The output layer node is determined by the demand form of the evaluation results, and the risk level is distinguished according to the risk (0–10) of the samples divided by 10 experts. [0, 2) is very low risk; [2, 4) is low risk; [4, 6) is high risk; [6, 8) is high risk; [8, 10) is very high risk. Therefore, the output value is based on the range of the number for risk rating, the output value is unique, and the output node should be 1.

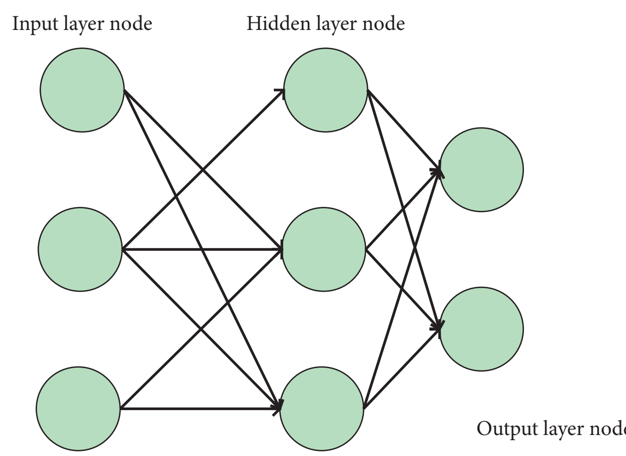

Figure 2: Node design.

Design of the number of hidden layer nodes: The input nodes were not uniformly specified in the previous model construction. At the same time, the fitting accuracy is affected by the number of hidden layer nodes, but the more the better, the complexity of the network and the training speed should be considered under the premise of ensuring a small error. It needs to adjust the number of nodes in the middle layer M according to the different research contents and select the number of nodes within a range according to the empirical formula. The commonly used empirical formula is as follows:

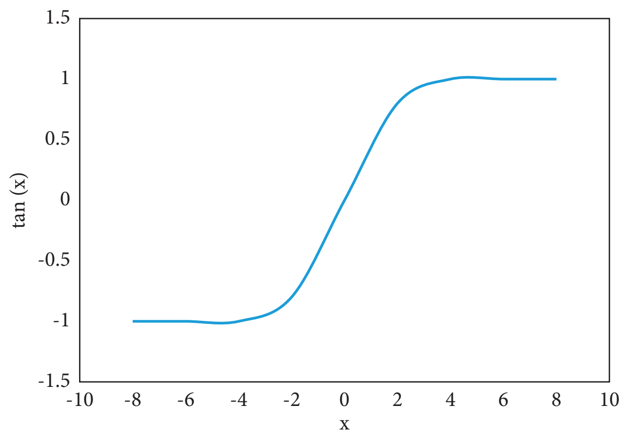

Figure 3: Sigmoid function.

$$M = \sqrt{PQ},$$
$$M = \log_2 P, \tag{1}$$
$$M = \sqrt{P+Q} + a,$$

where P is the number of nodes in the input layer, Q is the number of nodes in the output layer, and a is between [0, 10].

3.2.2. BP Neural Network Function Selection. The sigmoid function is a sigmoid function commonly seen in biology, also known as the sigmoid growth curve. In information science, the sigmoid function is often used as the activation function of a neural network to map variables between 0 and 1 due to its mono-increase and inverse function mono-increase properties. The transfer function is mainly the hidden layer function and the output layer function. As shown in Figure 3, the sigmoid function, as an excitation function, is not easy to diverge during the transfer process and maps the variable output range between (0-1) [14]. The participation of the sigmoid function enables the model to handle nonlinear data, and when using the sigmoid function for classification, probabilistic predictions can be calculated. This is of great significance to probabilistic decision-making, and the essence of risk assessment we do is the prediction of probability. In addition, the two ends of the sigmoid function are flat, and it is differentiable during calculation, which is of good help to the input and output of data, and improves the network's ability to map nonlinear data. The description formula is as follows:

$$f(x) = \frac{1}{1+\exp(-x)}, f(x) \in (0,1),$$
$$f'(x) = \frac{ae^{-ax}}{(1+e^{-ax})}, \tag{2}$$
$$f'(x) = af(x)[1-f(x)].$$

The function from the hidden layer to the output layer selects the purelin function. The function of this function is to output the training results quickly and select purelin function as the linear transfer function, so that the simulation steps are simplified and the transmission efficiency is high. The hidden layer sample function is shown in formula:

$$y_{ak} = f\left(\sum_{j=1}^{n} w_{jk} r_{pj} - \theta_k\right). \tag{3}$$

In the formula, the $k = 1, 2, \cdots, m$ and θ_k parameters represent the offset set in the hidden layer node, which is used to adjust the function. The output layer function is shown in formula:

$$b_p' = f\left(\sum_{k=1}^{m} w_k y_{ak} - \theta\right). \tag{4}$$

In the formula, θ represents the offset value set at the output layer node, and the calculation process of the inverse function representing its error is shown in formula:

$$E = \sum_{a=1}^{q} \left(b_p' - b_p\right)^2 \div 2. \tag{5}$$

In the formula, b_p' represents the actual output error, b_p represents the expected error, and the function calculation is performed to minimize the value of E to meet the output requirements.

The training function (trainlm function) is mainly a training algorithm for weights, which refers to the L-M (Levenberg–Marquardt) optimization algorithm. It has the fastest convergence speed for medium-scale BP neural network and is the default algorithm of the system. Since it avoids the direct calculation of the Hessian matrix, it reduces the amount of computation in training, but requires a larger amount of memory. From the literature research point of view, trainlm is easy to operate and fast to train. In the process of network training, the error curve can be quickly reduced and the accuracy can be improved, which has the advantages of fast training convergence speed and fewer steps. As long as there is a large memory, the training speed of trainlm can be guaranteed.

3.2.3. Evaluation Model Function Algorithm. The input evaluation sample data $p_k = (a_1^k, a_2^k, \cdots, a_n^k)$ calculate the input s_j of each unit in the middle layer through the connection weight w_{ij} and the threshold θ_j. The transfer function then calculates the output value of each layer b_j, of which $j = (1, 2, \cdots, p)$, and the calculation process is as follows:

$$s_j = \sum_{i=1}^{n} w_{ij} a_i - \theta_j,$$
$$b_j = f(s_j). \tag{6}$$

According to the output value b_j of the intermediate layer, the weight v_{jt} and the threshold value γ_t are connected, and the output value L_t of each unit is obtained. The output layer value C_t is obtained through the transfer function. This process is the calculation process of the output value of the input data of the meat product cold chain logistics quality risk assessment sample in the model; see

$$L_t = \sum_{i=1}^{p} v_{jk} b_j - \gamma_t, \tag{7}$$

$$C_t = f(L_t). \tag{8}$$

Comparing the target output value T_k with the calculated output value C_t, the error d_t^k of the calculated output layer is shown in formula :

$$d_l^k = \left(y_t^k - C_t\right) \cdot C_t (1 - C_t). \tag{9}$$

It uses the output layer error d_t^k and the output value b_j of the middle layer to correct the connection weight v_{jt} and the threshold γ_t to achieve the purpose of simulating experts, as shown in formulas in detail:

$$v_{jt}(N+1) = v_{jt}(N) + a \cdot d_t^k \cdot b_j, \tag{10}$$

$$\gamma_t(N+1) = \gamma_t(N) + a \cdot d_t^k. \tag{11}$$

3.2.4. Dimensionality Reduction of Principal Component Data. Before the data are input into the neural network model, the data are subjected to principal component dimension reduction [15, 16]. In the evaluation process of literature research, after reducing the dimension of the principal component of the data, this part of the data is brought into the network model and the original data without dimension reduction are substituted into the model for comparison. The results show that the BP network model after dimensionality reduction with principal component data can achieve higher accuracy with fewer iterations, while the relative error is lower [17]. The reason for the length of this paper briefly explains the comparison process between the BP network model after principal component dimension reduction and the original BP model. It first performs principal component processing on the data and then enters the model for use.

Principal component analysis uses the relevant variables to reduce the dimensionality so that the data can be explained with the main variable. The main variable can explain most of the information of the original variable, which has a great effect on preventing data overlap. The definition and derivation formula of the principal component are as follows: it sets a q-dimensional random variable, and it is second-order derivable at the same time. Mean and variance, according to linear transformation, are shown in formula :

$$\begin{cases} y_1 = a_{11}x_1 + a_{12}x_2 + \cdots + a_{1q}x_q = a_1'x, \\ y_2 = a_{21}x_1 + a_{22}x_2 + \cdots + a_{2q}x_q = a_2'x, \\ \vdots \\ y_q = a_{q1}x_1 + a_{q2}x_2 + \cdots + a_{qq}x_q = a_q'x. \end{cases} \tag{12}$$

Table 1: Encoding using Wubi input method.

Chinese character	Wubi code	Code list
ren	WWWW	23-23-23-23
sheng	TGD	20-7-4
ruo	ADKF	1-4-11-6
zhi	KWU	11-23-21
ru	VKG	22-11-7
chu	PUVN	16-21-22-14
jian	MQB	13-17-2

In order to obtain the vector y_1 with the largest variance in the linear function of $x_1, x_2, \cdots, x_q$, there is $V(ka_1'x) = k^2 V(a_1'x)$ for any constant k, set a_1 as the unit vector, and $a_1 a_1' = 1$. On this basis, a_1 is obtained when $V(y_1) = a_1' \sum a_1$ is maximized, and y_1 obtained at this time is the first principal component. $\sum T \wedge T'$. The specific calculation is shown in formula :

$$\sum = \sum_{i=1}^{q} \lambda_i t_i t_i'. \tag{13}$$

4. Language into Data

4.1. The Way of Chinese Character Dataization. The input method is the encoding method used to input text and symbols. The encoding methods of the Chinese input method are basically divided into two categories: based on pronunciation, such as pinyin input method; based on the meaning of font, such as Wubi font input method, Zheng code input method, and Cangjie input method; these types are collectively referred to as shape codes. Each shape code input method has its own original Chinese character dismantling logic and coding method, and is equipped with a radical table that can correspond to the "QWERTY" keyboard. Its characteristics are that it more accurately reflects the glyph structure and semantics of the Chinese character itself, and compared with the pinyin input method, it has a lower repetition rate [18, 19]. In view of this, this paper intends to use the graphic code input method as the data input source to form the framework foundation from text analysis to text-data conversion. The following will select the input method that will be used in this research through the introduction and comparison of the three common input methods.

4.1.1. Wubi Font Input Method. The root distribution method of the Wubi input method is based on the starting strokes of Chinese characters into five basic stroke areas: "horizontal," "vertical," "skimming," "squeeze," and "folding." The Wubi input method decomposes Chinese characters into three types: left-right type, up-down type, and hybrid type.

The verse "If life is only as first seen" is encoded using the Wubi input method, as shown in Table 1.

4.1.2. Zheng Code Input Method. According to the function of use and the ability to form words, Zheng code divides the

Table 2: Encoding using Zheng code.

Chinese character	Zheng code	Code list
ren	OD	15-4
sheng	MC	13-3
ruo	EGJ	5-7-10
zhi	JOVV	10-15-22-22
ru	ZMJ	26-13-10
chu	WTYD	23-20-25-4
jian	LR	12-18

Table 3: Encoding using Cangjie input method.

Chinese character	Cang Xie coding	Code list
ren	0	15
sheng	HQM	8-17-13
ruo	TKR	20-11-18
zhi	RC	18-3
ru	VR	22-18
chu	LSH	12-19-8
jian	BLU	2-12-21

basic root (hereinafter referred to as "basic root") into the main root and the auxiliary root, and the main root is further divided into the first main root and the second main root. The first main root is "one-code root"; that is, it is only necessary to type out the corresponding letter code alone, and it is the basic root with the strongest ability to group 26 words. The second main root and secondary root are designed as "two-code roots" in order to reduce the duplication code; that is, they need to be determined by the two codes of the area code and the bit code. The Zheng code encoding method is to disassemble the Chinese characters, according to the order of their strokes, to represent the basic root with the alphabetic code successively, but the quantity is limited to at most four-character codes [20]. The verse "If life is only as first seen" is encoded using Zheng code as shown in Table 2.

4.1.3. Cangjie Input Method. Cangjie input method is a popular Chinese input method in Taiwan. Its initial version was originally called "Xingyi Jianzi method" and was later renamed "Cangjie input method" to commemorate the spirit of Cangjie's character creation. At the beginning of its establishment, Mr. Zhu Bangfu's original intention was to develop a set of Chinese retrieval methods, so that Chinese characters, like pinyin characters, have an order, which is convenient for computer program retrieval.

The original intention of Cangjie input method is to input not only Chinese characters, but a more complete Chinese system. The developer means to use Cangjie coding to uniformly handle the six major issues of Chinese character code, word order, font shape, word pronunciation, word meaning, and word recognition. Therefore, Cangjie roots have been devoted to the subtle features that can reflect the structure of Chinese characters very intuitively at the beginning of the design. The Cangjie input method divides the roots into four categories: philosophy, strokes, personalities, and glyphs. Compared with Zhengma and Wubi, the Cangjie character splitting scheme is not aimed at typing speed in the use of the input method. However, from the perspective of the classification of the root table, its meaning is figurative, accurate, and refined, which is consistent with the roots of all things and the human body, and retains the ideographic characteristics of Chinese characters derived from pictographs to the greatest extent. The code acquisition principle of Cangjie input method is to take the first, second, third, and last code of the single character, and the split character is divided into the prefix and the body. The order of fetching code also follows three principles: from top to bottom, from left to right, and from outside to inside. When encoding, enter the encoding of the prefix and the font body in turn. The shortest of a single Chinese character is one yard, and the longest is five yards.

To sum up, the verse "If life is only as first seen" is encoded using the Cangjie input method, as shown in Table 3.

The three input methods listed above have their own advantages and disadvantages. Wubi coding and Zhengma have relatively fast input speed compared to Cangjie input method because of their character splitting logic and root distribution. However, given that the text-to-sound system was designed to have as good a mapping as possible between code and sound samples, it was decided to play back the samples through a combination of algorithms and a set of rules. This is bound to require that the used Chinese character code has a clear root-figurative meaning correspondence between its radical classification and the design of the radical table. From this point of view, due to the design of the division of strokes in Zhengma and Wubi coding, in their radical table, the main radicals represented by letters are not clear enough and irregular in meaning. If it maps to the sample, the performance will not be very smooth. Therefore, in view of the unique advantages of Cangjie input method, this paper intends to select Cangjie input method as the main encoding method of Chinese language text based on glyph data.

4.2. Design Ideas of Text and Sound System. This chapter will describe the basic structure of the text-acoustic system, the functions to be implemented, and discuss the problems and solutions that may be involved. The architecture of the text and sound system is divided into four parts: data preprocessing, data selection rules, sound material mapping rules, and sound processing rules, as shown in Figure 4.

Max/MSP is an interactive programming language and development environment for audio and media production. Max has been one of the most expensive environmental programs for developing interactive media and live performances for over a decade. Max lets the user build the sound, by connecting graphic objects to do the job. MSP adds a large number of objects for audio programming that makes the signal consistent. The Jitter object package allows users to create video or 3D graphics programming, or run

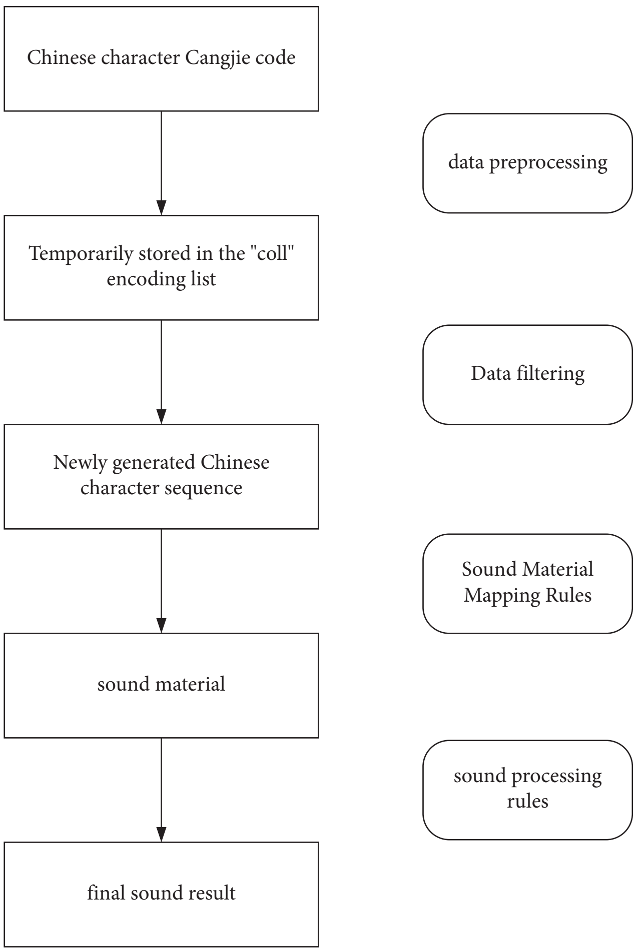

FIGURE 4: Basic architecture diagram of the text-acoustic system.

any matrix-based data. The text and sound system is based on modular thinking, and sub-modules (subpatch) are built in Max/MSP to realize the corresponding functions of each part.

4.2.1. Data Preprocessing. The Chinese text is disassembled by the Cangjie input method to obtain a sequence of numbers, which consists of a number of coded lists (lists) of at most five numbers. The text and sound system intends to build a sub-module as the input module, which is responsible for preprocessing the encoding list to realize the input, storage, and calling of the list. At the same time, the system is intended to be presented in a relatively concise and interactive logic operation mode, so that users can type Cangjie codes in the text-to-sound system just like typing on a keyboard in daily life, and all operations can be completed with a few keys. The input component is intended to contain the following functions:

Simulate the process of typing text using the keyboard, and only use the necessary keys such as letters and spaces to complete the complete data input process.

It uses the "coll" component to temporarily store the data entered by the user in the form of an argument plus a coded list (list) and realizes the function of changing and reading, so as to facilitate the mathematical arrangement and calculation required by the subsequent modules.

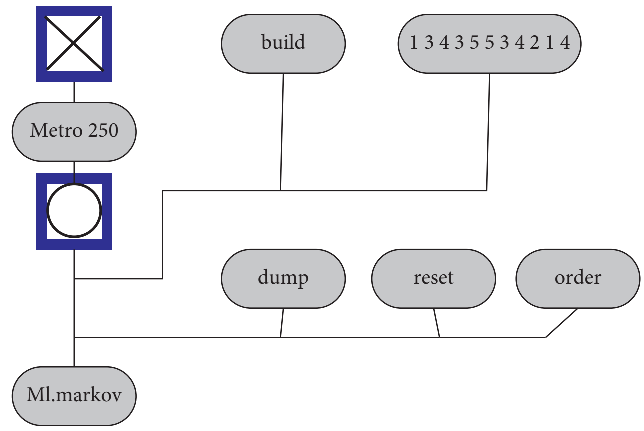

FIGURE 5: "Ml.markov" component.

4.2.2. Data Filtering Rules. The Markov Chain (MarkovChain) algorithm component "ml.markov" to be used by the text-acoustic system in Max/MSP is shown in Figure 5. This component comes from a set of machine learning toolbox "ml.*" written for Max, which can build a multi-order Markov chain model according to the input list (list) and trigger the generation of a new sequence with a "bang" message.

The Wensheng system plans to arrange the coded numbers composed of Chinese characters in the verses in order to build a list, and import them into the "ml.markov" component to establish a Markov chain model about the coded numbers. At the same time, each Chinese character in the verse is counted according to the number of codes of "one-code word," "two-code word," "three-code word," "four-code word," and "five-code word" quantity and established into a database. When "ml.markov" starts to generate a new sequence, the new number it generates determines the next Chinese character code number. It randomly selects Chinese characters in the font library of the corresponding coded numbers and uses the Chinese character code list to enter the subsequent modules of the system, as shown in Figure 6.

4.2.3. Sound Processing Rules. This section intends to build a sub-module responsible for sound playback and processing to receive the data output from the algorithm component "ml.markov." This sub-module will use a single Chinese character as the output unit to complete the final mapping of encoding. In other words, the encoding of the sound material area triggers the playback of the corresponding material, the encoding of the sound form area will control the duration, playback direction, and volume envelope, and the encoding of the real-time audio processing area will trigger the corresponding effector. And a single Chinese character, regardless of the number of codes, will only trigger the sound result module to work once, and the sound result takes a

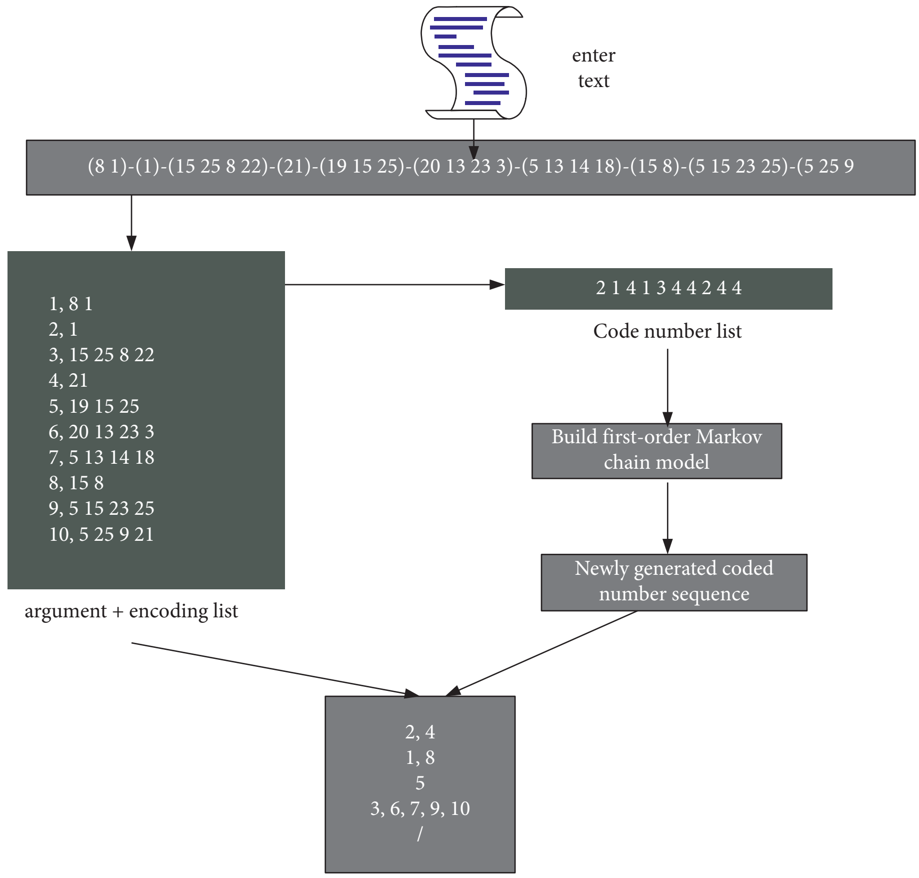

Figure 6: Schematic diagram of data selection rules.

Table 4: Coding—sound shape control mapping comparison.

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Table 5: Encoding—audio real-time processing mapping comparison.

Digital coding	Cang Xie coding	Audio real-time processing means		
19	*S*	Sending effect device	Frequency domain	Filter low cut
20	*T*			Filter high cut
21	*U*		Time domain	Reverb
22	*V*			Delay
23	*W*	Dry processing	Time domain	Time stretch
25	*Y*		Frequency domain	Pitch shift

single Chinese character as a whole and can achieve polyphonic output at the same time.

In order to enrich the diversity of sound results, the text-sound system uses the coding of stroke classes (*H* to *N*) according to the characteristics of their radical strokes and sets rules to determine the sound form, as shown in Table 4.

The "*H*" to "K" encoding will trigger the selection of the type of duration and playback direction of the sound material. The "*L*" to "*N*" codes determine the volume envelope of its playback. There are three levels of "fade in," "fade out," and "alternate between strong and weak" controlled by the "line~" component. The encodings between the material type and volume envelope do not conflict with each other.

The encoding of the glyph class (*S* to *Y*) triggers the audio real-time processing means through the rule setting, as shown in Table 5.

The "*S*" to "*V*" codes will trigger a send-type effect, and the sound signal processed in real time will be output to the main channel (Stereoout) in parallel. Among them, "*S*" and "*T*" trigger the low-cut and high-cut filters, respectively; "*U*" and "*V*" trigger the reverb and delay effects, respectively. Similarly, "*W*," "Y" encoding will trigger the processing of time stretching and pitch shifting. But unlike the former send effects, time stretching and pitch shifting are processed directly on the dry sound of the playback of the sound material.

To sum up, because the text-to-sound system has formulated three partitions on the coding mapping, different numbers of codes are combined in different mapping partitions, or there is a situation of logical conflict. For example, the encoding composition of the Chinese character "thing" is *J-L-L-N*, that is, the encoding list 10-12-12-14. Two problems immediately appeared: first, none of the four codes of the word belonged to the mapping area of the sound material, which resulted in no subsequent mapping options; second, there are two identical "*L*"s in the word code, and the codes "*L*" and "*N*" are both the volume envelope controls in the sound shape mapping area, so they conflict with each other. Therefore, in the sound playback and processing module, rules should be formulated for different encoding combinations. The following are divided into several situations for discussion:

If the Chinese character composition codes are all sound material areas, the sound playback processing module triggers the playback of the corresponding sound material of all composition codes.

If there is no encoding in the sound shape area, the default is to play back with a long duration value, forward playback, and a faded volume envelope. If there is no encoding in the real-time audio processing area, the processing module is disabled by default.

If none of the Chinese character composition codes belong to the sound material area, it is planned to sum up all the codes and divide it by the total number of categories of the sound material area 11, and the remainder determines the sound material selected for the character. The encoding of the sound morphology area and the audio real-time processing area is normally mapped.

If the number of codes in the sound shape area is greater than 1, the codes that determine the material type and control the volume envelope do not conflict with each other. If the code number of the material type is greater than 1, such as "8" and "9," it will be randomly selected among them to determine the material type. If the encoding number of the volume envelope is greater than 1, such as "12" and "13," it will be randomly selected to control the overall volume envelope.

The number of codes in the audio real-time processing area is not limited, and they can all be triggered without conflicting with each other.

4.3. Text and Sound System Test. In this paper, the system is tested by LoadRunner, and the performance of the system under high intensity is tested by simulating multiple users online at the same time through LoadRunner. The test results are as follows:

As shown in Figure 7, LoadRunner simulates 100 people online at the same time and records the response time of each person in the system. It can be seen that the response time of these 100 people is between 4 s and 5 s, which is in line with a stable time series. It can be considered that the response time of the system is relatively stable. The lowest response time is 4 s, the highest is 5 s, and the difference is 1 s. In actual situations, the average user's stay time will not exceed 5 s. If it exceeds 5 s, most users will choose to leave. From this, it can be found that the system in this paper is in line with the actual use requirements, and then the performance of the system is analyzed.

In order to test the maximum performance of the system, use LoadRunner to simulate 1 to 100 people online at the same time to test the transformation of the text-acoustic system. The article randomly selects 10 poems from the 300 Tang poems for literary-sound conversion and records the error rate. In order to eliminate the research error, the average value of the recognition error rate of 10 ancient poems was recorded. As shown in Figure 8, with the increase of the number of people, the error rate also increases, and the overall increase is stepped. This shows that for the best effect of the system, the situation where a large number of people use it at the same time should be avoided as much as

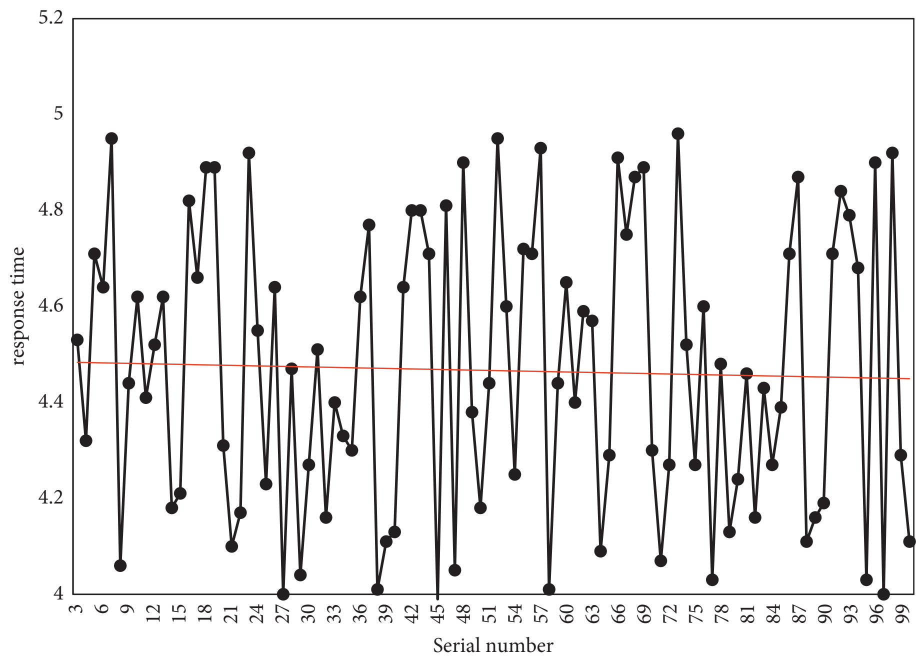

Figure 7: The response time of each person when 100 people are online.

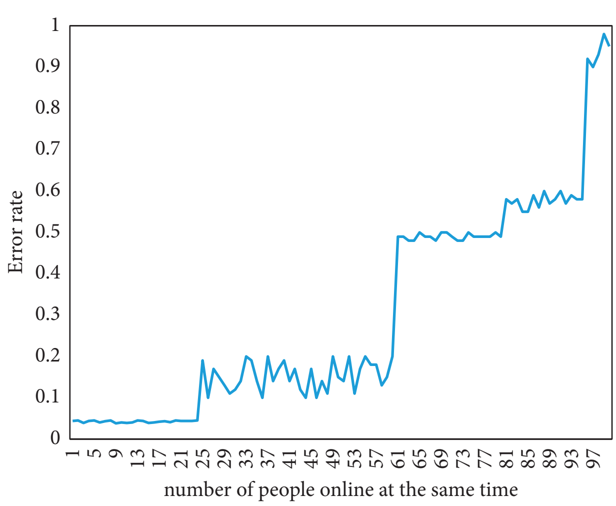

Figure 8: Performance analysis of text-acoustic system.

possible. Under 20 people, the error rate is about 4%, and the conversion effect is good, but between 21 and 57 people, the error rate increases to about 15%, which is not enough to meet the system requirements.

5. Conclusions

This paper makes a comparative analysis of two styles of Chinese narrative text and expository text using systematic functional linguistics. This paper makes detailed statistics on the components of the selected texts that reflect the three major meta-functions and textual cohesion and presents them in a table form, so that we have a real and concrete basis for our understanding of the similarities and differences between these two styles. It is no longer vague and empty, but truly "knows why." In addition, this paper also applies the results of discourse analysis and comparison in the practice of discourse teaching of Chinese as a foreign language, and puts forward some suggestions for improving teaching efficiency from the perspective of discourse teaching in the teaching of reading, writing, and listening in Chinese as a foreign language. Combined with the actual application effect of the text-acoustic system and the evaluation of the operation experience and sound of the text-acoustic system by several listeners, this paper believes that the text-acoustic system has the following possibilities for expansion and needs to be improved: in the input module, the text and sound system may have a real-time input encoding function to achieve the effect of generating sound similar to real-time encoding (Livecoding). In the actual creation and design of works, the data selected by the algorithm enter the sound module to process the sound, and the real-time encoding module can input additional Chinese characters into the sound module, so that the composer can use the system to create performances or creations. A higher-dimensional design and consideration can be carried out at the "voice" or sound level.

Data Availability

No data were used to support this study.

Conflicts of Interest

The authors declare that there are no potential conflicts of interest in this study.

References

[1] L. Johann-Mattis, S. J. Greenhill, and R. D. Gray, "The potential of automatic word comparison for historical linguistics," *PLoS One*, vol. 12, no. 1, pp. e0170046–e0170048, 2017.

[2] J. J. Mccarthy, "What is optimality theory?" *Language & Linguistics Compass*, vol. 1, no. 4, pp. 260–291, 2017.

[3] C. Stab and I. Gurevych, "Parsing argumentation structures in persuasive essays," *Computational Linguistics*, vol. 43, no. 3, pp. 619–659, 2017.

[4] T. Isaacs and L. Harding, "Pronunciation assessment," *Language Teaching*, vol. 50, no. 3, pp. 347–366, 2017.

[5] K. Murasugi, "Noun phrases in Japanese and English: a study in syntax, learnability and acquisition," *Journal of Japanese Linguistics*, vol. 13, no. 1, pp. 171-172, 2017.

[6] S. Y. Kuroda, "Pivot-Independent relativization in Japanese (iii)," *Journal of Japanese Linguistics*, vol. 4, no. 1-2, pp. 85–96, 2018.

[7] A. Williams, A. Drozdov, and S. R. Bowman, "Do latent tree learning models identify meaningful structure in sentences?" *Transactions of the Association for Computational Linguistics*, vol. 6, no. 1, pp. 253–267, 2018.

[8] H. A. Eva and V. L. S. Diana, "A closer look at formulaic language: prosodic characteristics of Swedish proverbs[J]," *Applied Linguistics*, vol. 38, no. 1, pp. 68–89, 2017.

[9] D. Jean-Marc, "Why the dichotomy 'L1 versus LX user' is better than 'native versus non-native speaker'," *Applied Linguistics*, vol. 39, no. 2, pp. 236–240, 2017.

[10] M. L. Vercellotti, "The development of complexity, accuracy, and fluency in second language performance: a longitudinal study," *Applied Linguistics*, vol. 38, no. 1, pp. 90–111, 2017.

[11] M. Popel and O. Bojar, "Training tips for the transformer model," *The Prague Bulletin of Mathematical Linguistics*, vol. 110, no. 1, pp. 43–70, 2018.

[12] A. Toral, "Fine-grained human evaluation of neural versus phrase-based machine translation," *The Prague Bulletin of Mathematical Linguistics*, vol. 108, no. 1, pp. 121–132, 2017.

[13] P. Durrant, "Lexical bundles and disciplinary variation in university students' writing: mapping the territories," *Applied Linguistics*, vol. 38, no. 2, pp. 165–193, 2017.

[14] H. G. Widdowson, "Conceptual and communicative functions in written discourse," *Applied Linguistics*, vol. 1, no. 3, pp. 234–243, 1980.

[15] Y. J. Kwon, H. J. Seo, J. N. Kim, and C. S. Lee, "Effect of grain boundary engineering on hydrogen embrittlement in Fe-Mn-C TWIP steel at various strain rates," *Corrosion Science*, vol. 142, pp. 213–221, 2018.

[16] N. K. Tewary, S. K. Ghosh, S. Chatterjee, and A. Ghosh, "Deformation and annealing behaviour of dual phase TWIP steel from the perspective of residual stress, faults, microstructures and mechanical properties," *Materials Science and Engineering A*, vol. 733, no. 22, pp. 43–58, 2018.

[17] J. Chen, F. T. Dong, H. L. Jiang, and Z. Liu, "Influence of final rolling temperature on microstructure and mechanical properties in a hot-rolled TWIP steel for cryogenic application," *Materials Science and Engineering A*, vol. 724, no. 2, pp. 330–334, 2018.

[18] M. Medvedeva, M. Kroon, and B. Plank, "15TH conference of the EUROPEAN chapter of the association for computational linguistics (eacl 2017)," *Bulletin of the Chemical Society of Japan*, vol. 34, no. 6, pp. 785–790, 2017.

[19] M. Coulthard, A. Johnson, and D. Wright, "An introduction to forensic linguistics: language in evidence by malcolm coulthard; alison johnson," *Studies in Linguistics*, vol. 30, no. 1, pp. 147–150, 2017.

[20] X. Feng, B. Qin, and T. Liu, "A language-independent neural network for event detection," *Science China Information Sciences*, vol. 61, no. 09, pp. 81–92, 2018.