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A Study on Optimizing English Teaching in Colleges and Universities by Integrating Computer-Assisted Language Learning and Traditional Teaching Methods

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Abstract With the development of information technology, computer-assisted language learning plays an increasingly important role in English teaching in colleges and universities. Traditional English teaching methods have problems such as insufficient interactivity and low degree of personalization, while computer-assisted teaching can provide a more flexible and diverse learning environment. In this study, a speech enhancement algorithm based on improved CTF-GSC and posterior Wiener filtering is proposed for the speech enhancement problem in computer-assisted language learning and applied to the practice of English teaching in colleges and universities. Methodologically, the basic principle of Wiener filtering is first analyzed, and it is proposed to optimize the Wiener filtering enhancement effect by combining the wavelet thresholding multi-window spectral estimation algorithm with the VAD algorithm; Second, the VSS-NLMS algorithm is introduced to improve the CTF-GSC algorithm to further enhance the speech enhancement effect; Finally, two teaching modes, interactive and collaborative, were designed to realize a new model of computer-assisted English teaching in colleges and universities. The experimental results show that under Gaussian white noise environment, the speech enhancement algorithm proposed in this study improves the signal-to-noise ratio to 32.0173 dB, which is 20.5848 dB and 15.9994 dB higher than the spectral subtraction method and the traditional Wiener filtering algorithm, respectively; the improved algorithm obtains an average score of 4.02 in the subjective scoring test, which is higher than the other comparative algorithms; and in the actual teaching application, the experimental class students' English scores on the posttest mean improved by 3.2 points over the pretest, while the control class only improved by 1.44 points. The study shows that the model of integrating computer-assisted language learning and traditional pedagogy can effectively improve the quality of speech and the effect of English teaching, which provides new ideas and methods for the reform of English teaching in colleges and universities.

Index Terms computer-assisted language learning, Wiener filtering, speech enhancement, CTF-GSC algorithm, interactive teaching mode, collaborative teaching mode

I. Introduction

Computer-assisted language learning (CALL) originated in the United States, referring to the exploration and research on the application of computers in language teaching and learning [1], [2]. Since the 1960s, CALL, as a cross-cutting field between information technology and linguistics, has made great progress, emerged with a large number of research results, and showed a trend of disciplining [3]. Language teaching and computer-assisted language learning have quite a lot in common with regard to the understanding of the way to develop language skills, from the beginning of the separation of listening, speaking, reading and writing and the emphasis on sub-practice to the gradual realization that language skills are a complex of skills that are closely interconnected with each other [4], [5]. Accordingly, the curriculum of English majors in some schools has changed the listening class to listening and speaking or audio-visual speaking class, reading to reading and writing, and combined speaking and writing, etc. [6]. However, in order to facilitate the sorting out of the developmental history of domestic and foreign scholars' research on computer-assisted language learning, we still start from four perspectives: listening, speaking, reading and writing.

In English listening teaching, beginner learners need to be able to recognize and master basic pronunciation, including intonation, rhythm and stress [7]. They also need to understand the pronunciation of everyday language based on real-life reading demonstrations, and through continuous practice, they need to be able to recognize and guess the meaning of the speaker even when they do not hear the words clearly [8]. CALL technology can achieve that learning purpose by supporting repeated listening, adjusting the speed of speech, simulating interactive situations and additional links to explain difficult points [9], [10]. After the introduction of Podcasts, some researchers began to pay attention to their value in listening learning and conducted teaching experiments because of their ability to push audio and video playback content to learners on a regular basis, and the ability for learners to create their own channels [11].

Oral skill development is the focus of CALL research, which contains channels such as human-computer dialog, audio and video conversations, voicemail, video podcasts and voice BBS [12]. Such channels require better network access to ensure the quality of audio and video, and the representative software includes Skype and so on. However, such software is commercially available and not designed for speaking practice, so their application to speaking practice requires careful instructional design [13], [14]. At the same time, and because these Internet instant chat tools, which allow both voice conversation and communication via text, provide a new perspective for research on improving speaking proficiency, researchers have begun to think about whether text conversations have an effect on improving speaking [15]. In recent years, speech recognition and synthesis technologies have also developed rapidly, enabling speech dictionaries and read-aloud texts to be embedded in various CALL systems [16]. Typical of these is the Candletalk program, which uses automatic speech recognition in human-computer conversations to improve participants' oral expression [17].

Computer-assisted reading applications, including electronic dictionaries, software that provides text, scenario, and multimedia annotations, computer-based reading training software, and Web-based reading activity design provide extremely valuable forms of activity for reading [18]. A related study in which the development of English reading skills was compared showed that computer-assisted reading classes improved reading levels significantly more than control classes [19]. A well-structured hypertext model is more helpful to learners with low reading levels because it fits the learners' cognitive processes and directs their attention to the important content [20]. There is also the use of software such as corpora to assess the level of difficulty of reading materials and to provide learners with additional aids to improve their reading skills by allowing them to repeat specific words in different contexts [21], [22].

Starting from the popularization of computers in the 1990s, word editors have been recognized by various industries as the most accepted assistive writing technology application due to their ease of use and simplicity of the final presentation form, but their word spelling and grammar tools are targeted at native speakers, and they are unable to recognize the errors of second language learners [23]-[25]. Here some researchers have developed software for second-language assignments, such as Fips Ortho, which is designed for beginner French speakers and uses a corpus of second-language learners' incorrect usage as a sample to prompt and modify word and grammar errors in writing [26]. Another example is the iWRITE system featuring a multimodal interactive environment, which provides online grammar resources based on a corpus. The technological development of CALL is rapidly changing, and it is more important for language learners and language teachers to choose the right technology to apply. However, it is not necessary to seek for the new, the existing application functions of the in-depth excavation, the existing technology applied to new areas, can have a new discovery. For example, Word text editor is familiar to many users, but its components such as annotations, revision process reproduction bookmarks and hyperlinks can be applied in writing learning [27]. As another example, Inputlog is built on learners' use of Word for writing and digs deeper into the cognitive process of learners' writing [28].

With the rapid development of information technology and the continuous updating of educational concepts, Computer-Assisted Language Learning (CALL) is increasingly widely used in English teaching in colleges and universities. The traditional English teaching mode mainly relies on teachers' lecturing and students' passive acceptance, and there are problems such as insufficient interactivity, low degree of personalization, and difficulty in guaranteeing the learning effect. Computer-assisted language learning provides rich learning resources and flexible and diverse learning modes, which can meet students' individualized learning needs and improve learning efficiency and quality. However, in the computer-assisted language learning system, the speech signal is often interfered by various noises, which affects the learning effect. Therefore, how to improve the quality of speech signals has become an important technical issue in computer-assisted language learning research. On the other hand, relying solely on computer-assisted language learning and ignoring the advantages of traditional teaching methods are also likely to lead to unsatisfactory teaching results. Integrating computer-assisted language learning and traditional teaching methods, giving full play to the advantages of both, and constructing a new type of college English teaching mode has become an important direction for the current reform of college English teaching.

Speech enhancement, as a key technology of computer-assisted language learning system, is of great significance to improve the quality of speech and enhance the learning experience. Wiener filtering is a classical speech enhancement algorithm, but in practical applications, the enhancement effect is limited due to the fact that the signal deviation cannot be completely eliminated. CTF-GSC algorithms are also widely used in the field of speech enhancement, but there are problems such as slow convergence speed and large mean square error.

Therefore, how to improve these algorithms to enhance the speech enhancement effect is a technical challenge to be solved in this study. In addition, how the computer-assisted language learning system can be organically integrated with the traditional teaching method to build a more efficient teaching mode is also the focus of this study.

In the practice of teaching English in colleges and universities, how to evaluate the teaching effect of the integration of computer-assisted language learning and traditional pedagogy is also a problem that needs to be studied in depth. Through experimental comparative analysis, it can provide empirical support and theoretical guidance for English teaching reform in colleges and universities. Considering the above issues comprehensively, this study aims to explore the optimization strategies for teaching English in colleges and universities with the integration of computer-assisted language learning and traditional pedagogy to improve the quality and effectiveness of teaching.

In this study, firstly, the basic principle of Wiener filtering is analyzed and the improvement strategy is proposed; secondly, the VSS-NLMS algorithm is introduced to improve the CTF-GSC algorithm; then, the improved CTF-GSC algorithm is combined with the improved Wiener filtering algorithm to construct the speech enhancement algorithm; then, the interactive teaching mode and the collaborative teaching mode based on the computer-assisted teaching are designed; finally, the effectiveness of the algorithm and the validity of the teaching mode are verified through experiments. Finally, the effect of the algorithm and the effectiveness of the teaching mode are verified. Through this research on the integration of technology and teaching methods, a new model of college English teaching combining computer-assisted language learning and traditional teaching methods is explored, providing theoretical basis and practical guidance for the reform of college English teaching.

II. Computer-assisted language learning system based on Wiener filter enhancement algorithm

II. A. Wiener filtering fundamentals

Wiener filtering can be used to extract speech signals in a smooth noise environment. Let the original speech be $y(n)$, the pure speech be $s(n)$, the noise be $d(n)$, and the estimated speech be $\hat{s}(n)$, which can be obtained after Wiener filtering with the system function $h(n)$:

$$y(n) = s(n) + d(n) \tag{1}$$

$$\hat{s}(n) = y(n) \times h(n) \tag{2}$$

Following the mean-square error criterion, the $s(n)$ and $\hat{s}(n)$ mean-square errors $\varepsilon = E\left[\left\{s(n) - \hat{s}(n)\right\}^2\right]$ are minimized and the Fourier transform yields the spectral estimation function of the Wiener filter as:

$$H(k) = P_{sy}(k) / P_y(k) \tag{3}$$

where, $P_y(k)$ is the power spectral density of $y(n)$; $P_{sy}(k)$ is the reciprocal power spectral density of $s(n)$ and $y(n)$.

Since $s(n)$ and $d(n)$ are uncorrelated, it can be obtained:

$$P_{sy}(k) = P_s(k) \tag{4}$$

$$P_y(k) = P_s(k) + P_d(k) \tag{5}$$

$$H(k) = P_s(k) / (P_s(k) + P_d(k)) \tag{6}$$

where, $P_s(k)$ is the speech power spectral density; $P_d(k)$ is the noise power spectral density [29].

Transform Eq. The numerator and denominator of the right side of the equation are divided by $P_d(k)$ in the same way to obtain:

$$H_i(k) = \xi(k) / (1 + \xi(k)) \tag{7}$$

where, $H_i(k)$ is the Wiener filter transfer function for the i th frame; $\xi(k) = P_s(k) / P_d(k)$ denotes the estimated a priori SNR at the k th frequency point.

Let $\hat{\xi}_i(k)$ be the a priori signal-to-noise ratio at the i th frame and the Wiener filter transfer function at the i th frame:

$$H_i(k) = \hat{\xi}_i(k) / (1 + \hat{\xi}_i(k)) \tag{8}$$

The Wiener filter output function can be obtained as:

$$\hat{S}_i(k) = H_i(k) Y_i(k) \tag{9}$$

where, $Y_i(k)$ is the speech power spectrum with noise.

II. B. Computer-assisted language learning system design

Aiming at the deficiencies of the CTF-GSC speech enhancement algorithm, the study further improves it, and at the same time combines it with the improved Wiener filtering algorithm to further enhance the denoising effect of the speech enhancement algorithm, and then enhance the effect of speech enhancement.

II. B. 1) Improved Wiener filtering algorithm

The Wiener filtering algorithm is one of the more commonly used speech enhancement algorithms, which mainly carries out the estimation of the minimum mean square value of the signal error and uses it as the best criterion [30]. The principle of Wiener filtering algorithm is shown in Figure 1.

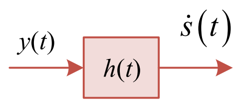


Figure 1: Wiener filtering algorithm principle

The noisy speech signal is set as $y(t)$, the pure speech signal as (t), and the background noise as $n(t)$, respectively. Then the a priori and a posteriori signal-to-noise ratios of the speech signal can be defined as respectively:

$$SNR_{\text{prio}}(k) = \frac{E[S^2(k)]}{E[N^2(k)]} \tag{10}$$

$$SNR_{post}(k) = \frac{X^2(k)}{E[N^2(K)]} \tag{11}$$

where $S(k)$ and $N(k)$ denote the variation form of the STFT for $s(t)$ and $n(t)$, respectively, and $X(k)$ is the best estimate of $S(k)$.

The gain function can be deformed as:

$$D(k) = \frac{SNR_{prio}(k)}{SNR_{prio}(k) + 1} \tag{12}$$

In practical scenarios, completely pure speech signals are difficult to obtain, and in traditional Wiener filtering algorithms, the a posteriori signal-to-noise ratio is usually computed in priority before the a priori signal-to-noise ratio is computed [31]. The a priori SNR of the signal in frame k is:

$$SNR_{prio}(k) = \varepsilon \frac{E[S^2(k-1)]}{E[N^2(k)]} + (1-\varepsilon)\max[SNR_{post}(k) - 1, 0] \tag{13}$$

where ε denotes the smoothing coefficient, which usually takes the value of 0.96.

Although the traditional Wiener filter enhancement algorithm is able to achieve a more accurate a priori signal-to-noise ratio when performing signal processing, the signal bias in the real situation cannot be completely eliminated and behaves differently under different estimation algorithms.

Since the deviation in the algorithm is unavoidable, it is considered to reduce it, and the optimization is carried out by estimating the noise signal, using the method of VAD algorithm, and at the same time, in order to further enhance the optimization effect, the wavelet threshold multi-window spectral estimation algorithm is further introduced on the basis of the wavelet thresholding, and after the optimization and improvement of the algorithm mentioned above, the gain function of the Wiener algorithm is converted to:

$$D(k)=\left(\frac{SNR_{prio}(k)}{SNR_{prio}(k)+\alpha}\right)^{\beta} \tag{14}$$

In the formula, α , β all represent the noise suppression factor, which is usually taken as a fixed value of 1, but taking this value will have an impact on the estimation accuracy of the a priori signal-to-noise ratio. Therefore, the study adjusts the value of α to:

$$\alpha=\begin{cases}1, & SNR_{post}(k)\geq 0.5\\ 1.5, & SNR_{post}(k)<0.5\end{cases} \tag{15}$$

Also set the value of β to change with the change of α , corresponding to the formula $\beta=\frac{1}{\alpha}$.

II. B. 2) Improved CTF-GSC algorithm

The study introduces the VSS-NLMS algorithm for step factor optimization and the improved step factor is calculated as:

$$u(k)=\theta u(k-1)+(1-\theta)\frac{\hat{\sigma}_e^2(k)}{\lambda\hat{\sigma}_e^2(k)} \tag{16}$$

where θ denotes the forgetting factor, λ denotes the positive parameter with added degrees of freedom design, $\hat{\sigma}_e^2(p,k)$ denotes the mean-square error, and $\hat{\sigma}_e^2(p,k)=\hat{\sigma}_e^2$ denotes the system noise power.

$$\hat{\sigma}_e^2(p,k)=\theta\hat{\sigma}_e^2(k-1)+(1-\theta)y^2(k) \tag{17}$$

$$\hat{\sigma}_v^2(p,k)=\hat{\sigma}_e^2(k)-\frac{1}{\hat{\sigma}_y^2(k)}\hat{r}_{ey}(k)^T\hat{r}_{ey}(k) \tag{18}$$

where $\hat{r}_{ey}(k)$ denotes the correlation between $y(k)$ and $e(k)$ and $\hat{\sigma}_y^2(k)$ denotes the input signal power, which can be estimated by Eqs. (19) and (20), respectively.

$$\hat{\sigma}_y^2(k)=\theta\hat{\sigma}_y^2(k-1)+(1-\theta)y^2(k) \tag{19}$$

$$\hat{r}_{ey}(k)=\theta\hat{r}_{ey}(k-1)+(1-\theta)y(k)e(k) \tag{20}$$

With the introduction of the VSS-NLMS algorithm, the influence of the speech signal on the noise estimation can be further reduced, and it also enables the Adaptive Noise Canceller (ANC) module in the algorithm to improve the convergence speed, stronger stability and interference suppression, which in turn further reduces the mean-square error and obtains the optimal estimation of the speech useful signal.

II. B. 3) Speech Enhancement Algorithm Based on Improved CTF-GSC and Posterior Wiener Filtering

The improved CTF-GSC algorithm is combined with the improved Wiener filtering algorithm to construct the final speech enhancement algorithm. The specific steps of the algorithm are: firstly, the output signal is processed using the wavelet threshold multispectral window estimation algorithm, which is obtained as the average noise spectrum of the speech signal, and then it is updated and computed, and the updating method is the VADS algorithm. The structure of the speech enhancement algorithm based on improved CTF-GSC and posterior Wiener filtering is schematically shown in Fig. 2.

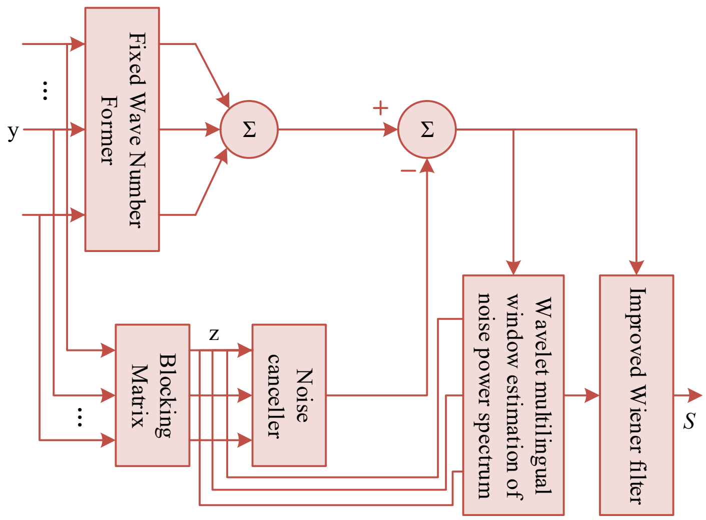


Figure 2: Speech enhancement algorithm structure schematic

The update of the average noise power spectrum is calculated as:

$$\bar{\lambda}_d(p,k)=\begin{cases}\alpha\bar{\lambda}_d(p-1,k)+(1-\alpha)\bar{\lambda}_y(p,k), & \beta(p)<T\\ \bar{\lambda}_d(p-1,k), & \beta(p)>T\end{cases} \tag{21}$$

The judgment parameter $\beta(p)$ is calculated as:

$$\beta(p)=\left[SNR_{post},\frac{SNR_{prio}}{1+SNR_{prio}}\right]-\log\left(\left[1+SNR_{prio}\right]\right) \tag{22}$$

where α denotes the smoothing coefficient, which is taken to be 0.96, T denotes the fixed threshold, which is taken to be 0.16, $[\cdot]$ denotes the mean computation, and $\bar{\lambda}_y(p,k)$ denotes the mean power spectrum.

The a priori signal-to-noise ratio of the subsequent signals is performed by a judgment algorithm:

$$SNR_{prio}(p+1)=\max\begin{Bmatrix}\tau\dfrac{\hat{V}hatS_k(n)^2}{\bar{\lambda}_d(p,k)}+(1-\tau)\max[SNR_{post}(p+1)-1,0]\\ SNR_{prio\ \min}\end{Bmatrix} \tag{23}$$

In Eq. (23), $SNR_{prio\min}$ generally takes the value of -21dB, and τ denotes the smoothing coefficient, which takes the value of 0.8.

The frequency domain speech signal after the enhancement process can be expressed as:

$$S(k)=D(k)\hat{s}(k) \tag{24}$$

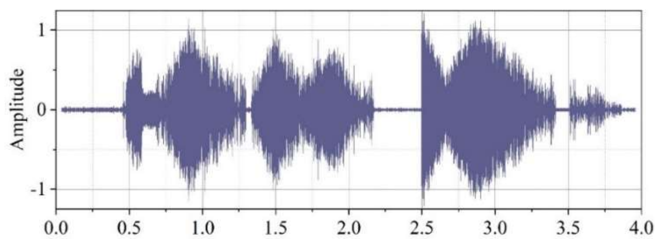
The frequency-domain speech signal $S(k)$ is subjected to a short-time Fourier inverse transform, and the resulting transformed signal is then subjected to frame merging to obtain the final enhanced speech signal.

III. Algorithm simulation and analysis

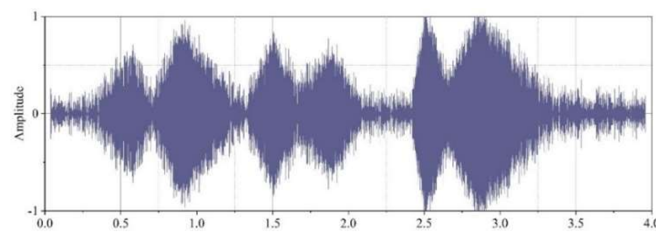
The pure voice used in the simulation experiment is the voice of a female for about 10 seconds, which adopts a sampling rate of 16KHz and a sampling precision of 20 bits. Through the audio software, we can see that there is about 0.5 seconds of mute section in 10 seconds of speech, so the length of the leading silent section is set to be 0.3 seconds, and in order to ensure that the speech is 10ms~30ms short-term smooth characteristics, the frame length is set to be 236 samples, because the sampling rate is 16Khz, the sampling period is 65.5us, then 236 samples is 16ms, which satisfies the short-term smooth conditions of the speech signal. In order to verify the noise reduction effect of the algorithm, the Gaussian white noise is mixed in the whole pure speech segment, and the signal-to-noise ratio of the speech signal with noise is 5dB.

Firstly, to verify the noise reduction effect of spectral subtraction method, in the spectral subtraction method written in Matlab, set the over-reducing factor a as 6, the gain compensation factor b as 0.001, and then calculate the signal-to-noise ratio of the speech after denoising by spectral subtraction method, and save the denoised speech signal with a waveform file through wavwrite function, and then ask five students to make a subjective evaluation. Then verify the voice enhancement algorithm based on a priori SNR Wiener filtering, set the weighting coefficients of the a posteriori SNR in the a priori SNR Wiener filtering algorithm written in Matlab, calculate the SNR of the speech signal after denoising based on the a priori SNR Wiener filtering algorithm through the formula of the SNR and save it with a waveform file through the wavwrite function of the denoised speech information. Then we ask five laboratory students to test and listen to the test and make a subjective evaluation. Finally verify the algorithm of this paper to set the sampling frequency Fs to 15KHz, calculate the signal-to-noise ratio of the speech after the improved algorithm, and save the denoised speech signal as a waveform file through the wavwrite function, and then ask five students to make a subjective evaluation of the quality of the denoised speech.

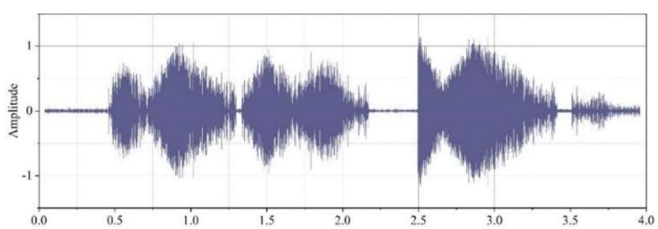
The graphs of pure speech, noise, and denoised speech signals are shown in Fig. 3. (Figures a~e are the waveforms of pure speech, the signal-to-noise ratio of 5db for noisy speech, the waveforms after spectral subtraction filtering, the waveforms based on the a priori signal-to-noise ratio Wiener filtering, and the waveforms after applying the algorithm of this paper, respectively)



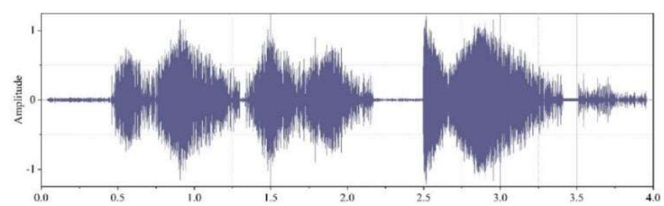
(a) Pure voice waveform



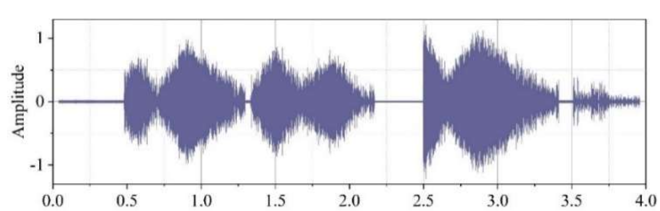
(b) Noise ratio 5db



(c) Spectral minus filter waveform



(d) Based on the waveform of the prior signal-to-noise filter



(e) Ours

Figure 3: Signal diagram

The signal-to-noise ratio of the denoised speech is shown in Table 1. By comparing and analyzing the signal-to-noise ratio data in the table, it is easy to see that the denoising effect of the algorithm in this paper is obviously much better than that based on the a priori signal-to-noise ratio of the Wiener filtering algorithm and spectral subtraction denoising effect. By listening to the waveform audio file saved after denoising, it can be found that there is still more noise left in the noiseless section of the speech lead after the spectral subtraction denoising method, and in the speech section, it can also be clearly heard with the sound of “clattering” water, the sound is not very crisp. And based on the a priori signal-to-noise ratio Wiener filtering algorithm denoising speech after the leading voiceless section of the basic noise clean, and from the whole section of the speech, the denoising effect is obviously better than the spectral subtraction method, you can hear the sound of the human voice, but the human voice no longer sounds so crisp, indicating that the noise in some frequency bands is not well eliminated. After this paper's algorithm denoising speech not only in the leading noiseless section of the noise is basically eliminated, and in the voice section can be clearly heard in the human voice, the sound is crisp and pleasant, a better restoration of pure speech signals, in the voice of the intelligibility and clarity of the two aspects have been greatly improved.

Table 1: Noise ratio

Noise type	Noise speech	Spectral subtraction	Based on the prior signal-to-noise, the voice is sound	Ours
Signal-to-noise ratio	SNR (dB)	SNR (dB)	SNR (dB)	SNR (dB)
Gaussian white noise	5	11.4325	16.0159	32.0173

After the subjective opinion scoring of the denoised speech signal by 5 lab students. The average opinion scores are shown in Table 2, from the table it can be seen that the average score of this paper's algorithm is the highest 4.02, and the 5 lab students have a high evaluation of this paper's algorithm.

Table 2: Average score

Test speech	Test the people's views score					Final score
	1	2	3	4	5	
Noise speech	1.4	1	1.2	1	1.3	1.18
Spectral subtraction	2.7	3.4	3.1	2.9	2.7	2.96
Based on the prior signal-to-noise, the voice is sound	3.2	3.6	3.1	3.3	3.4	3.32
Ours	4.2	3.9	3.8	4.4	3.8	4.02

IV. Designing a new model for teaching English in colleges and universities based on computer-assisted language learning

IV. A. Computer-assisted instruction program design

IV. A. 1) Interactive teaching models

Interactive teaching mode emphasizes the interaction between people and things, mainly human-to-human interaction, human-computer interaction, and interaction between learners and what they learn, and these interactions can be one-to-one or one-to-many or many-to-many. Computer-assisted language learning emphasizes personalized, interactive learning, independent learning and collaborative learning, and the composition of the interactive teaching model is shown in Figure 4.

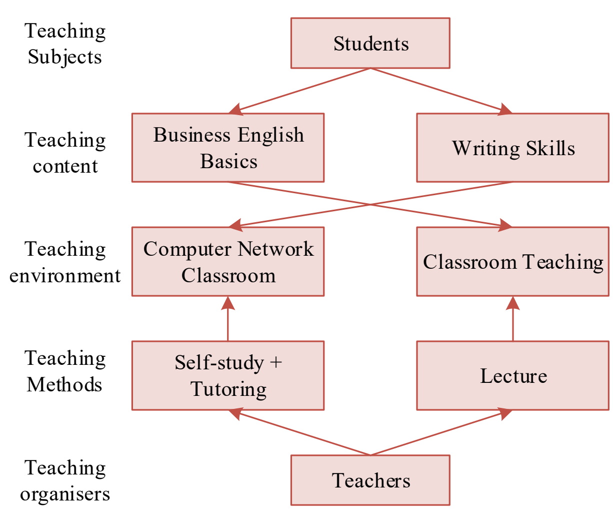


Figure 4: The interaction teaching pattern is made up

IV. A. 2) Collaborative teaching models

Computer-assisted collaborative teaching has four main components: teacher-guided learning, on-site discussion, data collection and collaborative learning, and the composition of the collaborative teaching model is shown in Figure 5.

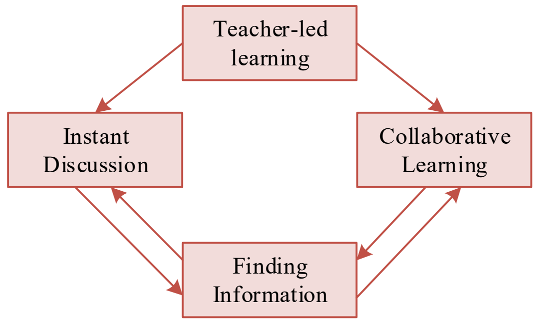


Figure 5: Composition of collaborative teaching mode

Among them, the process of teaching English with computer assistance has the following main aspects and processes, i.e., assigning homework, discussing problems, answering questions and counseling, and completing homework, and the process in writing teaching is shown in Figure 6.

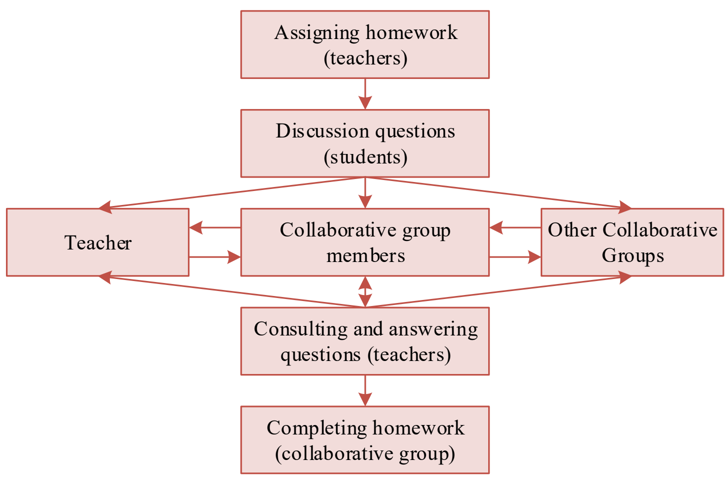


Figure 6: Process in writing teaching

IV. B. Analysis of the effect of the application of the English teaching model

In this study, two English teaching classes in a university were selected as experimental subjects, with class size of 20 students respectively, and experimental and control classes were set up. The experimental class used the teaching model proposed in this paper for English teaching, while the control class used the traditional teaching method for English teaching. This study needs to test the students' English performance during the experiment, utilizing Test Paper I for pre-test and Test Paper II for post-test respectively. The whole experimental process is divided into three stages, which are pre-test, mid-test and post-test. The time period is from September 2024 to December 2024, which is four months. The number of students in each score band of the pre-test of English scores of the experimental and control classes is shown in Figure 7. We can see that the experimental class and the control class do have a more consistent distribution of the number of people in each score band of the pre-test achievement, and combined with the results of the independent samples t-test of the data from the test paper 1, the overall level of the students is not significantly different, and can be used as an experimental subject.

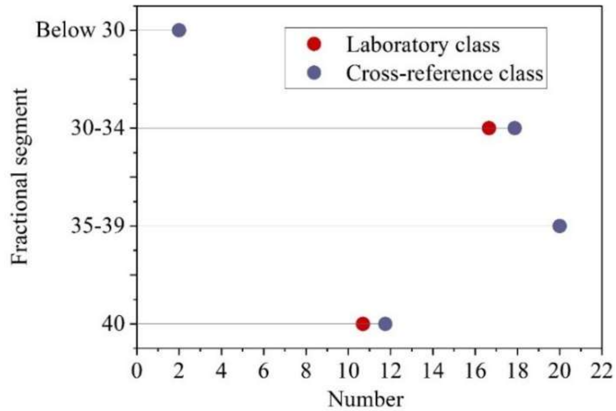


Figure 7: The number of points in the laboratory class and the cross-section English grades

After the test, Microsoft Excel was utilized to tally the students' scores on each item. The experimental class pre- and post-test scores are shown in Table 3. The pre- and post-test scores of the control class are shown in Table 4. From the table, it can be seen that after using the method proposed in this paper for English teaching, the English scores of the students in the experimental class improved more obviously, and the average of the post-test scores was 3.2 points higher than that of the pre-test, while the post-test English scores of the students in the control class were 1.44 points higher than that of the pre-test.

Table 3: Test results before and after the experimental class

	Premeasurement					Posttest				
School number	Problem 1	Problem 2	Problem 3	Problem 4	Total score	Problem 1	Problem 2	Problem 3	Problem 4	Total score
1	10	8	8	7	33	10	10	8	6	34
2	10	10	5	8	33	10	10	10	8	38
3	10	8	8	10	36	10	8	8	6	32
4	8	10	9	7	34	10	10	8	10	38
5	10	10	9	9	38	10	10	8	9	37
6	10	10	4	4	28	10	10	10	10	40
7	10	10	4	9	33	10	10	8	8	36
8	10	10	5	4	29	10	10	9	9	38
9	8	10	10	10	38	8	10	10	10	38
10	10	10	4	10	34	10	10	9	6	35
11	10	10	4	8	32	10	10	10	8	38
12	10	10	8	8	36	10	10	8	7	35
13	8	10	5	6	29	10	10	10	9	39
14	10	10	10	9	39	10	10	8	7	35
15	10	10	5	4	29	10	10	9	10	39
16	8	10	9	7	34	8	10	9	8	35
17	10	10	9	7	36	10	10	10	9	39
18	8	10	5	8	31	10	10	8	7	35
19	8	10	10	6	34	8	10	8	10	36
20	10	8	10	6	34	10	10	10	6	36
21	10	10	4	9	33	10	10	8	9	37
22	10	10	4	8	32	10	10	10	6	36
23	10	10	8	7	35	10	10	9	7	36
24	10	10	4	7	31	10	10	10	6	36
25	10	10	10	4	34	10	10	8	9	37

Table 4: Test results before and after the cross-section

	Premeasurement					Posttest				
School number	Problem 1	Problem 2	Problem 3	Problem 4	Total score	Problem 1	Problem 2	Problem 3	Problem 4	Total score
1	10	6	9	9	34	10	10	9	7	36
2	8	10	10	8	36	10	6	7	9	32
3	10	8	7	9	34	10	10	10	9	39
4	10	10	8	8	36	10	10	8	7	35
5	10	10	8	10	38	10	10	8	10	38
6	10	8	5	7	30	10	10	9	7	36
7	8	10	10	8	36	8	10	10	8	36
8	10	10	9	7	36	10	10	8	9	37
9	10	10	10	10	40	10	10	7	7	34
10	10	10	5	8	33	10	10	7	8	35
11	8	10	7	7	32	8	10	8	7	33
12	8	10	8	10	36	8	10	8	9	35
13	10	6	9	9	34	10	10	10	10	40
14	10	10	8	9	37	10	10	7	8	35
15	10	10	9	8	37	10	10	10	10	40
16	10	8	6	7	31	10	8	9	8	35
17	10	10	10	9	39	10	10	10	7	37
18	10	8	5	8	31	10	10	7	10	37
19	10	10	7	7	34	10	10	9	8	37
20	8	10	6	7	31	10	10	8	7	35
21	10	10	6	7	33	10	10	9	9	38
22	10	6	10	7	33	10	6	9	7	32
23	10	10	6	9	35	10	10	10	9	39
24	8	10	8	7	33	10	10	10	7	37
25	10	10	8	9	37	10	10	7	7	34

V. Conclusion

In this study, a speech enhancement algorithm based on improved CTF-GSC and posterior Wiener filtering is proposed and applied to computer-assisted college English teaching, which has achieved remarkable results. The experimental results show that the proposed speech enhancement algorithm achieves a signal-to-noise ratio of 32.0173 dB in a Gaussian white noise environment, which is significantly improved over the spectral subtraction method's 11.4325 dB and the a priori signal-to-noise ratio based Wiener filter's 16.0159 dB. In the subjective scoring test, the average rating of this algorithm by the five testers was 4.02, which was 1.06 and 0.7 points higher than the 2.96 of spectral subtraction and the 3.32 of traditional Wiener filtering, respectively, indicating that this algorithm has a significant advantage in terms of speech quality. A comparison of the teaching effects between the experimental class and the control class reveals that the students' performance in the experimental class, which adopts the teaching model proposed in this study for English teaching, has been improved by 3.2 points, while the students' performance in the control class, which adopts the traditional teaching method, has been improved by only 1.44 points, with a gap of 1.76 points. This fully proves the effectiveness and superiority of the new model of teaching English in colleges and universities that integrates computer-assisted language learning and traditional teaching methods.

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