

Research Article

Research on the Current Situation and Countermeasures of English Listening Teaching Based on Multimedia Intelligent-Embedded Processor

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In order to improve the effect of English listening teaching, this paper combines the interactive needs of English listening teaching to analyze the current problems in English listening teaching. Moreover, according to the actual needs of the intelligent English teaching system, this paper conducts research on the high-order cumulant signal-to-noise ratio estimation method and combines data analysis to improve the algorithm to obtain an intelligent algorithm suitable for English listening teaching. In addition, this paper combines the algorithm to construct an English listening teaching system based on a multimedia intelligent-embedded processor and applies the embedded processor to intelligent English listening teaching. Finally, this paper builds an intelligent system to solve the current problems in English listening teaching and improve the effect of English teaching.

1. Introduction

At present, there are still major problems in English listening teaching in some colleges and universities, but this teaching link is the key to improving students' English ability. Therefore, in the future development process, colleges and universities need to continuously improve and innovate teaching models and teaching ideas based on current problems and use advanced teaching methods to improve students' English skills. From the perspective of teachers, English teachers in colleges and universities do not pay enough attention to the teaching of English listening, and the formalization is obvious, and the recording of English texts is mainly played in the class, and the mode is relatively simple. Some teachers think that the English listening link is a waste of most of the class time, so they often omit this link in class, resulting in lower listening proficiency of students [1]. From the perspective of students, there are two main factors that affect the improvement of their English listening level: knowledge barriers and noninformation barriers. The so-called intellectual barriers are problems in cultural background knowledge and language knowledge, including grammatical knowledge, vocabulary knowledge, and phonetic knowledge. Having

good speech skills is the prerequisite and foundation of listening. Individual learners have been practicing English for many years, and their reading and writing skills have reached a certain level, but their listening skills are relatively low. The main reason is the lack of phonetic knowledge [2]. In recent years, the scale of enrollment of colleges and universities has been expanding, and the entrance threshold has been gradually reduced. Due to the poor English foundation of students, their pronunciation skills are even worse. Nonintellectual barriers refer to barriers in listening methods, skills, and psychology. College students have poor learning foundations and skills, so they will inevitably have emotions such as fear and anxiety during the learning process, which directly leads to a decrease in their interest in English learning. In addition, methods and skills are also key factors that affect the improvement of English listening ability, so they need to be paid attention to. From the perspective of college English courses, although this subject is a basic course, there is still a problem of a small amount of courses. However, the emergence of this problem will directly cause students to ignore this course and attach importance to other "professional" courses that involve more. In addition, even if the amount of courses is up to

the standard, some colleges and universities still involve less listening courses and cannot provide students with comprehensive listening training [3]. The emotional factors that affect college students' English listening learning include motivation, attitude, personality, belief, and self-confidence. The learning motivation is the inner force that promotes students to carry out learning activities and directly affects the learning effect. It mainly includes result motivation, integration motivation, and instrumental motivation. Self-confidence is the judgment and evaluation of students on their own learning process, which can guide students to actively participate in learning activities. All in all, emotion is the filter of knowledge learning, and the knowledge input to the brain can only be fully absorbed by students after being filtered. Interaction design has developed rapidly in recent years, and along with the cross-development of other professions and technologies, it has produced many amazing results, changing and subverting people's lifestyles, the behavioral relationship between "people and things," and "people and things." The group relationship of "people" and the cultural relationship of "people and society" have been redefined under the impetus of new technology. The revolution of information technology has brought tremendous changes to the development of human society and directly promoted the overall innovation of production methods, communication channels, and business models related to the design discipline. From the perspective of disciplinary development, we need to review interaction design from the perspective of evaluation and then reverse the evaluation method of interaction design. This paper combines the multimedia intelligent-embedded processor to analyze English listening teaching and builds an intelligent English listening model to improve the effect of English listening teaching.

2. Related Work

In the embedded field, embedded processors play an important role and are leading the development of the entire embedded field. Many domestic and foreign companies have participated in the development boom of embedded processors and developed various products. The development version of the commercial embedded processor has further promoted the research and application of embedded processors [4]. At present, the embedded processor system is widely used, and the smart terminal field is widely optimistic. Many domestic and foreign developers are increasing investment and developing new products. The embedded processor will be their preferred operating system. As an excellent representative of free software, Linux has developed rapidly in less than ten years due to its unique characteristics of high efficiency, safety, and dynamic loading [5]. At present, the scope of application of embedded systems at home and abroad mainly includes aerospace, NASA's Mars Climate Orbiter, "Polar Lander," and "Deep Space Two," and other Mars probes use embedded systems. In the VxWorks system, information appliances include digital set-top boxes, digital TVs, video phones, home networks, and mobile PDAs; the industrial market has control equip-

ment, industrial control boards, instruments, etc. [6]; in addition to advanced medical diagnostic equipment, smart houses, smart office, etc. are indispensable for the core technology of embedded systems [7]. Due to the rapid development of technology, the diversification of user needs, and the continuous advancement of differentiation, the development of embedded intelligent terminal systems has become an emerging scientific research field and industry. According to the definition of the British Institute of Electrical Engineers, an embedded system is a device that controls and monitors auxiliary equipment, machinery, or even factory operations [8]. Embedded systems mainly include hardware and software. With the increase in system complexity and the development of hardware integration technology, a higher-performance microprocessor has become the core hardware component of the embedded system; the software has only the program block with a single control function which has developed into a software system with layers and embedded operating systems [9]. With the deepening of the information age, the development of digitization has become an inevitable trend, and one of the core technologies of digitization is digital signal processing [10]. Digital signal processing is the use of computers or special processing equipment to collect, transform, filter, estimate, enhance, compress, and identify signals in digital form to obtain a signal form that meets people's needs. Digital signal processing (DSP) is an emerging discipline that involves many disciplines and is widely used in many fields at the same time [11]. For example, in the field of mathematics, calculus, probability and statistics, stochastic processes, and numerical analysis are all basic tools for digital signal processing and are closely related to network theory, signal and system, cybernetics, communication theory, and fault diagnosis. In recent years, some emerging disciplines, such as artificial intelligence, pattern recognition, and neural networks, are inseparable from digital signal processing [12]. It can be said that digital signal processing takes many classic theoretical systems as its theoretical basis and at the same time makes itself the theoretical basis of a series of emerging disciplines. With the rapid development of computer and information technology, digital signal processing technology emerged and developed rapidly. Especially in the application of embedded computers, DSP technology has become one of the mainstream directions of computer technology development today, mainly in the field of measurement and control. The purpose of the embedded system is to provide a complex digital system with multitasking and networking as the core which is easy to develop. From the perspective of digital technology and information technology, embedded systems have become the basic technology of modern information network technology applications and have become the basic technology in the field of modern industrial control [13]. Although the theory of digital signal processing has developed rapidly, it is no exaggeration to say that the birth and development of DSP chips have played a very important role in the technical development of communications, computers, and control in the past 20 years [14]. The task of digital signal processing needs to be completed by DSP devices to a large extent. DSP technology has become a cutting-edge

technology that people are paying more and more attention to and getting rapid development. DSP is a device that processes a large amount of information with digital signals. The new DSP not only has data processing capabilities but also integrates more and more other components inside, which can form a new DSP-embedded application system, which not only has other microprocessing. The advantages of the embedded system of the processor and the single-chip microcomputer have a unique high-speed digital signal processing capability [15].

3. High-Order Cumulant Signal-to-Noise Ratio Estimation Method

The main step of the high-order cumulant method is to calculate the result of the conversion of high-order quantities into signal moments. According to the connection of signal moment, noise power, and signal power, the estimated value of signal and noise power is calculated, respectively, and then, the corresponding signal-to-noise ratio is obtained. For a zero-mean k -order random process $x(n)$, the values of l have different time nodes: the higher-order moments and higher-order cumulants of $x(n), x(n-k_1), \dots, x(n-k_{l-1})$ are defined as equations (1) and (2):

$$m_l(k_1, k_2, \dots, k_{l-1}) = \text{mom}\{x(n), x(n-k_1), \dots, x(n-k_{l-1})\}, \quad (1)$$

$$c_l(k_1, k_2, \dots, k_{l-1}) = \text{com}\{x(n), x(n-k_1), \dots, x(n-k_{l-1})\}. \quad (2)$$

We assume that $X = (x_1, \dots, x_l)$ is a vector and $I = (1, 2, \dots, l)$ is the indicator set of vector X . If $I \subseteq I_x$, then X_I represents each component of IX vectors. Suppose the vector $X_I = (\chi_1, \dots, \chi_n)$, and if $\chi_i = 1$, then $i \in I$, and if $\chi_i = 0$, then $i \notin I$. These vectors have a one-to-one correspondence with the set $I \subseteq I_x$. Therefore, you can get [16]

$$\begin{aligned} m_x(I) &= m_x^{(X(I))}, \\ c_x(I) &= c_x^{(X(I))}. \end{aligned} \quad (3)$$

In other words, $m_x(I)$ and $c_x(I)$ are the moments and cumulants of the subvector X of X_I . According to the above formula, we get [17]

$$c_x(I) = \sum_{U_p I_p = I} (-1)^{q-1} (q-1)! \prod_{p=1}^q m_x(I_p). \quad (4)$$

Among them, $C_x(I)$ represents the sum in all divisions of I $1 \leq q \leq N(I)$.

For formula (4), $l=1$, $l=2$, and $l=3$ are as follows:

$$l=1 : c_1(x_1) = E\{x_1\}, \quad (5)$$

$$l=2 : c_2(x_1, x_2) = E\{x_1 x_2\} - E\{x_1\}E\{x_2\}, \quad (6)$$

$$\begin{aligned} l=3 : c_3(x_1, x_2, x_3) &= E\{x_1 x_2 x_3\} - E\{x_1\}E\{x_2 x_3\} \\ &\quad - E\{x_2\}E\{x_1 x_3\} - E\{x_3\}E\{x_1 x_2\} + 2E\{x_1\}E\{x_2\}E\{x_3\}. \end{aligned} \quad (7)$$

Based on the previous derivation of the formula and their respective connections, for a complex random variable with zero mean, the moment is defined as [18]

$$M_{p+q,p} = E[x^p (x^*)^q]. \quad (8)$$

We define its cumulative amount as

$$C_{p+q,p} = \text{cum}[x, \dots, x, x^*, \dots, x^*]. \quad (9)$$

Among them, x is the p item, and x^* is the q item.

For a complex random process x with zero mean, we have

$$\begin{aligned} \text{cum}_4^1(l_1, l_2, l_3) &= \text{cum}_4[X^*(k), X(k+l_1), X(k+l_2), X^*(k+l_3)] \\ &= E[X^*(k)X(k+l_1)X(k+l_2)X^*(k+l_3)] \\ &\quad - E[X^*(k)X(k+l_1)] \cdot E[X(k+l_2)X^*(k+l_3)] \\ &\quad - E[X^*(k)X(k+l_2)] \cdot E[X(k+l_1)X^*(k+l_3)] \\ &\quad - E[X^*(k)X(k+l_3)] \cdot E[X(k+l_1)X^*(k+l_2)], \end{aligned} \quad (10)$$

$$\begin{aligned} \text{cum}_4^2(l_1, l_2, l_3) &= \text{cum}_4[X(k), X(k+l_1), X(k+l_2), X(k+l_3)] \\ &= E[X(k)X(k+l_1)X(k+l_2)X(k+l_3)] \\ &\quad - E[X(k)X(k+l_1)] \cdot E[X(k+l_2)X(k+l_3)] \\ &\quad - E[X(k)X(k+l_2)] \cdot E[X(k+l_1)X(k+l_3)] \\ &\quad - E[X(k)X(k+l_3)] \cdot E[X(k+l_1)X(k+l_2)], \end{aligned} \quad (11)$$

$$\begin{aligned} \text{cum}_4^3(l_1, l_2, l_3) &= \text{cum}_4[X(k), X(k+l_1), X(k+l_2), X^*(k+l_3)] \\ &= E[X(k)X(k+l_1)X(k+l_2)X^*(k+l_3)] \\ &\quad - E[X(k)X(k+l_1)] \cdot E[X(k+l_2)X^*(k+l_3)] \\ &\quad - E[X(k)X(k+l_2)] \cdot E[X(k+l_1)X^*(k+l_3)] \\ &\quad - E[X(k)X^*(k+l_3)] \cdot E[X(k+l_1)X(k+l_2)]. \end{aligned} \quad (12)$$

According to the formula of the above algorithm, the MPSK signal is simulated below, assuming that the complex digital sequence of the MPSK signal is

$$r_k = \sqrt{E}a_k + n_k = x_k + n_k \quad k = 1, 2, \dots, N. \quad (13)$$

In the formula, the noise of the mean value is zero, the variance is the complex Gaussian random variable with the variance N_0 , E is the energy of the transmitted symbol, and $a_k \in \{\exp(j2\pi(m-1)/M), m = 1, \dots, M\}$, M is the signal modulation order.

It is defined that the signal $x_k = \sqrt{E}a_k$ and the noise $n_k = n_{1k} + jn_{2k}$ are not related to each other. According to the previous definition, we can get complex Gaussian random

variables with n_{lk} and n_{Qk} mean zero and variance $N_0/2$, and they are not related to each other. $\text{SNR} = E/N_0$ is the signal-to-noise ratio.

For MPSK signals with $\{A \exp(j2\pi(m-1)/M), m = 1, \dots, M\}, A = \sqrt{E}$. Normally, to define that the transmitted signals are not related to each other, it is only necessary to calculate the value of the cumulant function when $l_1 = l_2 = l_3 = 0$.

$$\begin{aligned} E[X(k)X(k)X(k)X(k)] &= \frac{1}{M} \sum_{m=1}^M A^4 \exp\left(\frac{4 \cdot j2\pi(m-1)}{M}\right) = \begin{cases} A^4 & M = 2, 4, \\ 0 & M = 8, 16, 32 \dots \end{cases} \\ E[X(k)X(k)X(k)X^*(k)] &= \frac{1}{M} \sum_{m=1}^M A^4 \exp\left(\frac{2 \cdot j2\pi(m-1)}{M}\right) = \begin{cases} A^4 & M = 2, \\ 0 & M = 4, 8 \dots \end{cases} \\ E[X^*(k)X(k)X(k)X^*(k)] &= \frac{1}{M} \sum_{m=1}^M A^4 \exp\left(\frac{0 \cdot j2\pi(m-1)}{M}\right) = A^2 \quad M = 2, 4, 8 \dots, \\ E[X^*(k)X(k)] &= E[|X(k)|^2] = \frac{1}{M} \sum_{m=1}^M A^4 \exp\left(\frac{0 \cdot j2\pi(m-1)}{M}\right) = A^2 \quad M = 2, 4, 8 \dots, \\ E[X(k)X(k)] &= E[X^2(k)] = \frac{1}{M} \sum_{m=1}^M A^4 \exp\left(\frac{2 \cdot j2\pi(m-1)}{M}\right) = \begin{cases} A^2 & M = 2, \\ 0 & M = 4, 8 \dots \end{cases} \end{aligned} \quad (14)$$

It can be obtained by formulas ((3)-(4)), ((3)-(5)), ((3)-(6)), ((3)-(7)), ((3)-(8)), ((3)-(9)), and ((3)-(10)).

$$\begin{aligned} C_{20} &= E[X^2], \\ C_{21} &= E[|X|^2], \end{aligned} \quad (15)$$

$$\begin{aligned} C_{41} &= \text{cum}(X^*, X, X, X) = E[X^*XXX] - 3E[X^2] \cdot E[|X|^2] \\ &= E[X^*XXX] - 3C_{20}C_{21}, \end{aligned} \quad (16)$$

$$\begin{aligned} C_{42} &= \text{cum}(|X|^*, |X, X, X|) = E[|X|^*|XXX|] - 2(E[|X|^2])^2 \\ &\quad \cdot E[X^2] \cdot E[|X|^*]^2 = E[|X^*|XXX|] - 2C_{21}^2 - |C_{20}|^2. \end{aligned} \quad (17)$$

Because the fourth-order cumulant is zero, noise and signal are not related to each other. According to the previous formula and properties, we can get the following:

$$C_{r,21} = C_x, \quad \frac{1}{21} C_n \quad (18)$$

$$C_{r,41} = C_x, \quad (19)$$

$$C_{r,42} = C_{x,42}, \quad (20)$$

σ_r^2 , σ_x^2 , and σ_n^2 represent the variances of received signal, sent signal, and noise, respectively. When $M = 2$, that is, BPSK signal, there are

$$C_{x,21} = \sigma_x^2 = E, \quad C_{x,42} = -2E^2, \quad C_{x,41} = -2E^2. \quad (21)$$

When $M \geq 4$, which is a high-order MPSK signal, there are

$$C_{x,21} = \sigma_x^2 = E, \quad C_{x,42} = E^2, \quad C_{x,41} = 0. \quad (22)$$

From the above, we can get the relationship between the cumulant of MPSK signal and its symbol energy, as shown below:

$$E = \sigma_x^2 = \begin{cases} \sqrt{C_{x,42}/2}, & M = 2, \\ \sqrt{|C_{x,42}|}, & M \geq 4. \end{cases} \quad (23)$$

From formula (20), $C_{r,42} = C_{x,42}$, and formula (23) can be written as follows:

$$E = \sigma_x^2 = \begin{cases} \sqrt{C_{r,42}/2}, & M = 2, \\ \sqrt{|C_{r,42}|}, & M \geq 4. \end{cases} \quad (24)$$

From formula (18), we can get the following:

$$\sigma_r^2 = \sigma_x^2 + \sigma. \quad (25)$$

In summary, we can give the steps of signal-to-noise ratio estimation as follows:

- (1) The algorithm obtains the cumulant estimated value $C_{r,42}$ and the variance estimated value $C_{r,21}$ according to the collected signal $r_k, k = 1, 2 \dots, N$
- (2) The algorithm calculates the noise variance estimate $\sigma_n^2 = \sigma_r^2 - \sigma_x^2$
- (3) The algorithm calculates the energy E of the signal

The following algorithm calculates the estimated signal-to-noise ratio.

$$\text{SNR} \frac{E}{N} = \frac{S_x^2}{S_n^2} = \frac{S_x^2}{S_r^2 - S_x^2} = \begin{cases} \frac{\sqrt{C_{r,42}/2}}{C_{r,21} - \sqrt{|C_{r,42}|/2}} & \text{BPSKsignal} \\ \frac{\sqrt{|C_{r,42}|}}{C_{r,21} - \sqrt{|C_{r,42}|}} & \text{MPSKsignal} \end{cases} \quad (26)$$

When the modulation order is unknown, certain calculations are required. The algorithm first performs a simple analysis of the order and uses the analysis results to make the following assumptions:

$$\Delta = \frac{|C_{r,41}|}{|C_{r,42}|} = \frac{|C_{x,41}|}{|C_{x,42}|} \begin{cases} > 0.5 \text{ The received sequence is BPSK signal,} \\ < 0.5 \text{ The received sequence is a non-BPSK signal.} \end{cases} \quad (27)$$

Theoretically, the results obtained by formula (26) and the second-order fourth-order estimation method ((3)-(28)) are the same.

$$\widehat{\text{SNR}}_{\text{complex}} = \frac{\sqrt{|2M_2^2 - M_4|}}{M_2 - \sqrt{|2M_2^2 - M_4|}}, \quad (28)$$

$$\text{SNR}_{\text{real}} = \frac{(1/2)\sqrt{6M_2^2 - 2M_4}}{(M_2 - (1/2))\sqrt{6M_2^2 - 2M_4}} = \frac{\sqrt{|M_4 - 3M_2|/2}}{M_2 - \sqrt{|M_4 - 3M_2|/2}}. \quad (29)$$

In the above formula, M_2 and M_4 are the second-order and fourth-order quantities of the received sequence $r(n)$. The following formulas M_2 and M_4 are related to the power of the collected signal.

$$M_2 = \frac{1}{N} \sum_{n=0}^{N-1} |y_n|^2, \quad (30)$$

$$M_4 = \frac{1}{N} \sum_{n=0}^{N-1} |y_n|^4.$$

For the above formula, there is such a relationship between the moment and the cumulant, as shown below:

$$M_2 = E[|X|^2] = C_{21}, \quad (31)$$

$$M_4 = E[|X|^4] = E[X^*XXX^*].$$

When $M \geq 4$, $C_{20} = 0$, $C_{r,42} = M_4 - 2M_2^2$. When ν , since the constellation diagram of the BPSK signal is one-dimensional, so $C_{20} = C_{21}$, then $C_{r,42} = M_4 - 3M_2^2$ at this time. Figure 1 shows the root-mean-square error of the signal-to-noise ratio estimation by the higher-order cumulative estimation method.

In actual situations, generally speaking, when perceiving the spectrum, the prior information of authorized users will not be obtained. Therefore, there are certain limitations to the estimation of the signal-to-noise ratio. In order to improve the accuracy of spectrum sensing, appropriate channel quality metrics and related reliable estimation algorithms are required. The following will expand the range of modulation methods to estimate the unified signal-to-noise ratio.

$$Y_i = X_i + W_i. \quad (32)$$

In the formula, Y_i is the received signal, X_i is the transmitter signal, and W_i is the channel noise.

The other ν represents the second moment of the received signal Y_i .

$$M_2 = E[|Y_i|^2] = E[|X_i|^2] + 2 \text{Re} \{E[|X_i|W_i^*]\} + E[|W_i|^2]. \quad (33)$$

$E[\]$ is the statistical mean value formula. The fourth moment of the received signal Y_i can be expressed as follows:

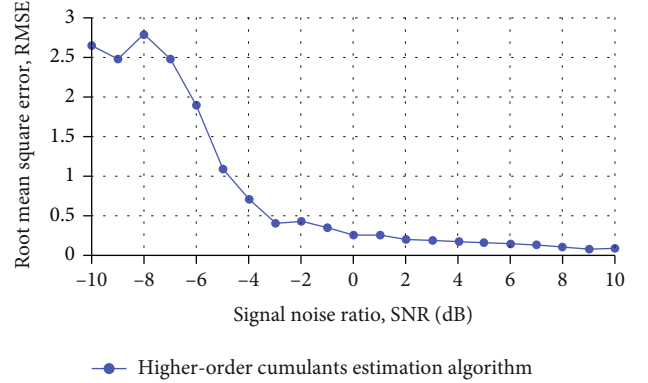


FIGURE 1: The root mean square error of the signal-to-noise ratio estimation method of the order cumulant estimation method.

$$\begin{aligned} M_4 &= E\left[|Y_i|^4\right] = E[|X_i|^4] + 4 \text{Re} \{E[|X_i|^2 X_i W_i^*]\} \\ &\quad + 2 \text{Re} \left\{E\left[(X_i W_i^*)^2\right] + 4E[|X_i|^2 |W_i|^2]\right\} \\ &\quad + 4 \text{Re} \{E[X_i |W_i|^2 W_i^*]\} + E[|W_i|^4]. \end{aligned} \quad (34)$$

We assume that the signal X_i and the noise W_i are independent zero-mean random processes, and W_i is a complex Gaussian; the kurtosis of Y_i is defined as follows:

$$C_Y = \frac{M_4}{M_2^2} = \frac{E[|Y_i|^4]}{E[|Y_i|^2]^2}. \quad (35)$$

Similarly, the kurtosis of X_i is represented by C_X . Through some algorithm operations, the estimated value $\hat{\gamma}$ of the average signal-to-noise ratio can be derived from (32)–(35).

$$\begin{aligned} \hat{\gamma} &= \frac{E[|X_i|^2]}{E[|W_i|^2]} = \frac{(C_Y - 2)}{(C_X - C_Y)} \\ &\quad + \frac{4.3\sqrt{(4 - 2C_Y)^2 - (C_X - C_Y)(2 - C_Y)}}{2(C_X - C_Y)}. \end{aligned} \quad (36)$$

The derivation of the above formula not only satisfies the low-order statistics BPSK and MPSK but is also suitable for 16QAM and 64QAM. This expands the range of modulation methods. Figure 2 shows that the improved SNR estimation algorithm can satisfy 16QAM SNR estimation. Figure 3 shows that the improved SNR estimation algorithm can satisfy 64QAM SNR estimation.

The above simulation results show that the improved SNR estimation algorithm can more accurately estimate the SNR for 16QAM and 64QAM modulated signals.

Figures 4 and 5 show the simulation results of BPSK and MPSK signals by the improved SNR estimation algorithm under the same conditions as the classical high-order cumulants.

Derived by the above formula, the simulation results obtained are shown in Figures 4 and 5. For BPSK and QPSK

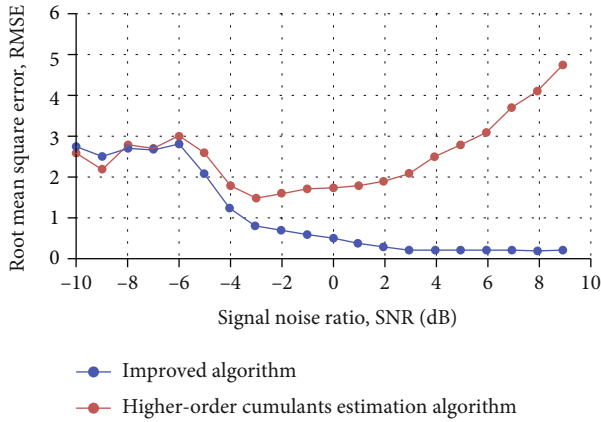


FIGURE 2: Comparison of improved 16QAM SNR estimation algorithm and high-order cumulant estimation.

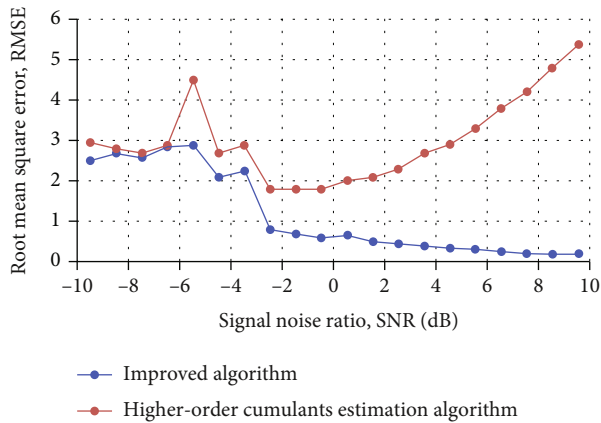


FIGURE 3: Comparison of improved 64QAM SNR estimation algorithm and high-order cumulant estimation.

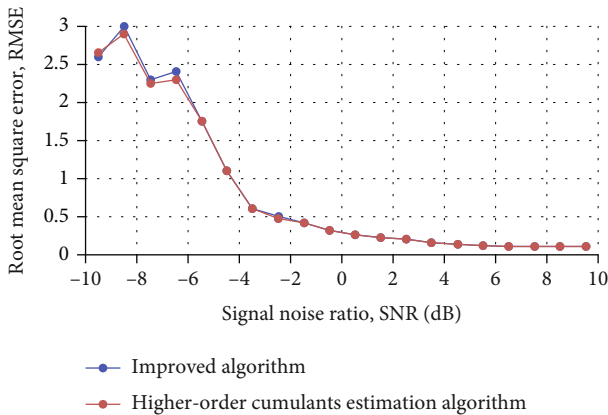


FIGURE 4: Comparison of improved BPSK signal-to-noise ratio estimation algorithm and high-order cumulant estimation.

signals, the improved SNR estimation algorithm is not only equivalent to the high-order cumulant, but the improved algorithm can estimate the SNR of 16QAM and 64QAM more accurately.

Through the above research, the embedded processor signal-to-noise ratio processing method of this paper is con-

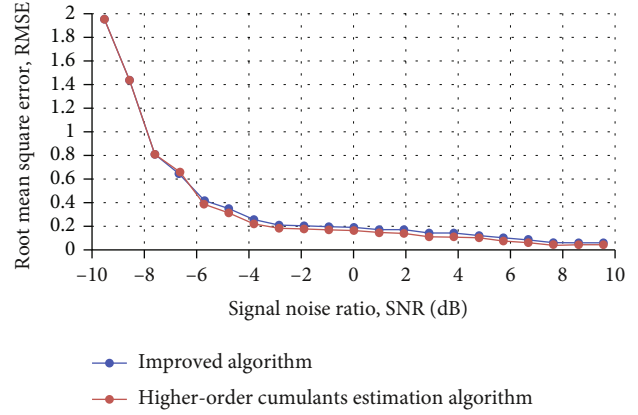


FIGURE 5: Comparison of improved QPSK signal-to-noise ratio estimation algorithm and high-order cumulant estimation.

structed. This method is applied to English listening teaching, and then the embedded system of English listening teaching can be constructed.

4. English Listening Teaching System Based on Multimedia Intelligent-Embedded Processor

As a basic compulsory course for college students, college English listening and speaking is essential to improve students' comprehensive ability and employability. In recent years, many scholars have conducted research on multimedia teaching and the newly emerged "hybrid" teaching, "flipped classroom" and "MOOC," and other computer-assisted teaching methods, but their focus and classification methods are not the same. The article regards "hybrid" teaching, "flipped classroom," "MOOC," and "online live broadcast" commonly adopted by Chinese universities and primary and secondary schools during the epidemic as "new multimedia teaching." However, the current English listening teaching in colleges and universities in my country has not fully utilized the advantages of the "new multimedia," which has caused a waste of valuable resources such as new multimedia resources and teacher and student time. Teachers combine their own years of teaching experience and summarized the questions collected in the form of tests, interviews, and questionnaires from the three aspects of students, teachers, and schools. According to previous researches, college students generally have problems in the cognitive and technical aspects of the multimedia teaching style of college English listening courses. First of all, at the cognitive level, the vast majority of students are basically ignorant of "new multimedia teaching." According to previous investigations, students are already familiar with multimedia teaching synonymous with PPT. However, the "new multimedia teaching" form synonymous with "hybrid" teaching, "flipped classroom," "MOOC," and "online live broadcast" has only begun to appear and mature in recent years. Even in colleges and universities with relatively better teaching conditions, it is still not popularized, especially in primary and secondary schools. Therefore, most college students have never heard of new multimedia forms such as "hybrid" teaching, "flipped classroom," and "MOOC" before

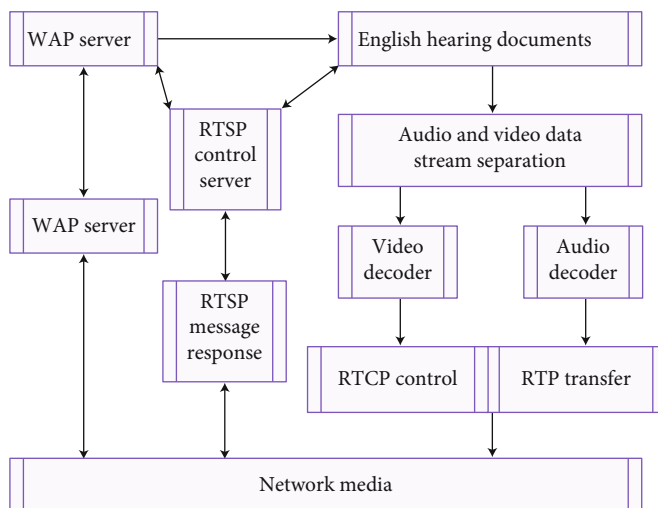


FIGURE 6: Framework structure of mobile streaming media service.

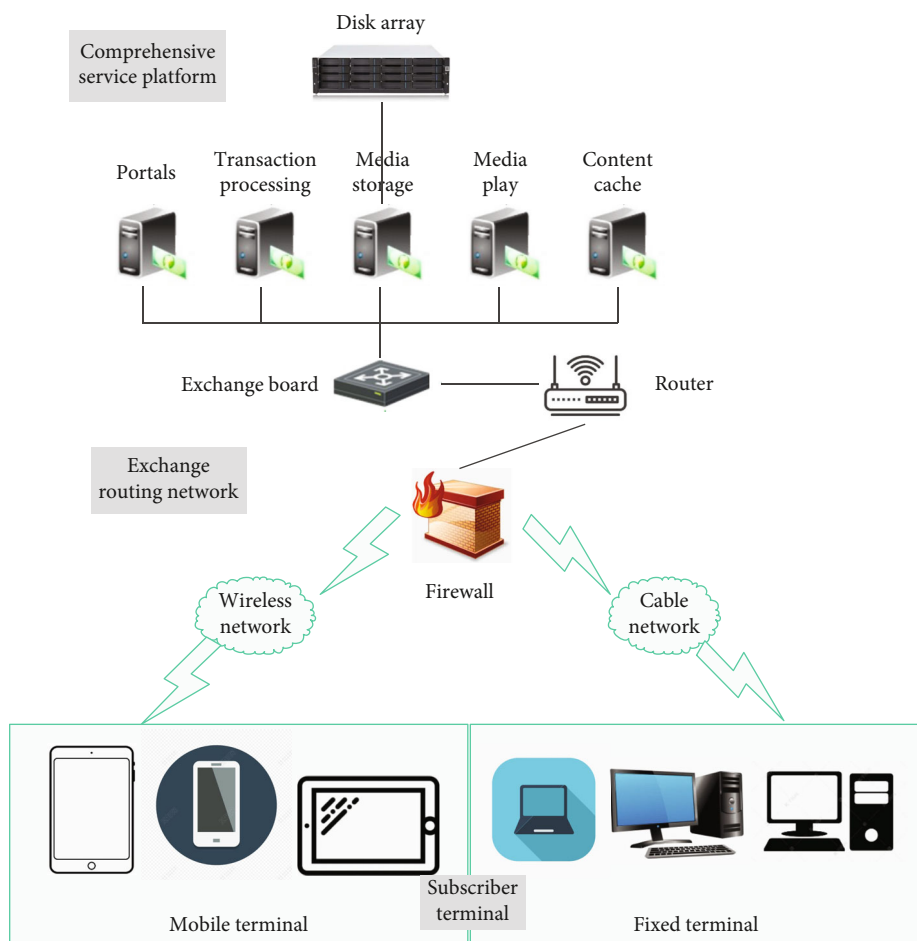


FIGURE 7: The structure of English listening streaming media on-demand system.

entering the school. The lack of understanding of new multimedia teaching methods will create obstacles for them to quickly adapt to college listening courses. Secondly, at the technical level, students have insufficient grasp of the technology and operation methods required by the new multi-

media. Although the current college students are called the “indigenous people” of the Internet age, there are still some students with poor family conditions who are unfamiliar with the operation of computers and mobile phones. In addition, even students who are proficient in the operation

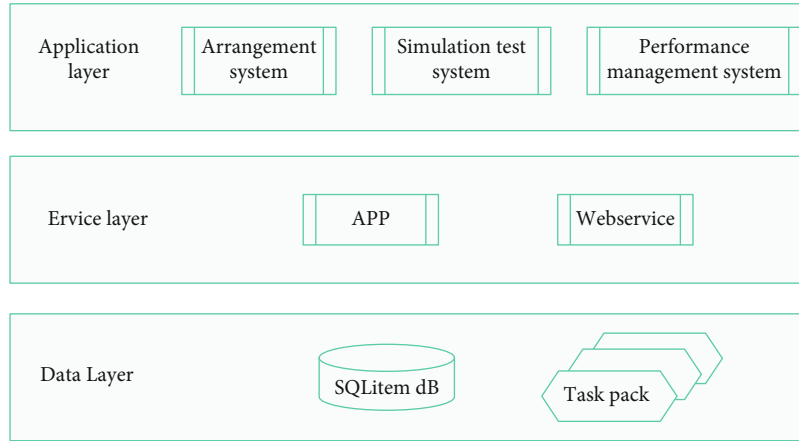


FIGURE 8: System overall architecture diagram.

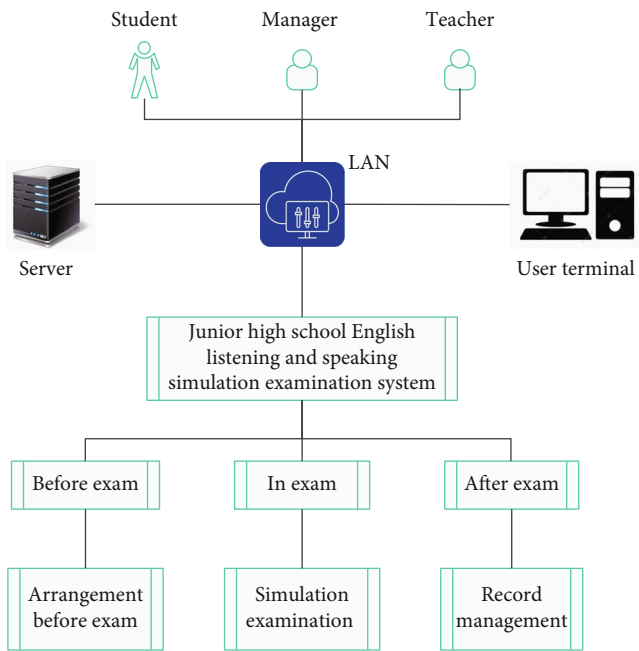


FIGURE 9: The functional overview of the English listening test system.

of computers and mobile devices are basically only familiar with a few instant messaging, social software, shopping, and film and television software. For a new client developed for learning purposes, students may not be able to operate it at all. In the new multimedia environment, few students have any knowledge of the audio clips, video clips, and subtitle addition technologies and corresponding software often assigned by listening teachers. Therefore, students currently lack the skills needed to complete multimedia assignments and activities.

The “new multimedia teaching” model represented by “mixed” teaching, “flipped classroom,” and “MOOC” has appeared for many years and has become the development trend of college English listening teaching in the future. Some problems in the school may cause the development of “new multimedia teaching” to fail to achieve the expected

TABLE 1: The effect of high-order cumulant signal-to-noise ratio estimation method in English listening speech processing.

Number	Noise removal evaluation	Number	Noise removal evaluation
1	96.575	21	93.920
2	95.868	22	95.849
3	95.864	23	94.595
4	94.477	24	94.248
5	97.208	25	96.740
6	95.225	26	95.717
7	93.223	27	93.508
8	94.886	28	93.193
9	94.097	29	94.106
10	96.675	30	93.495
11	94.100	31	97.959
12	96.788	32	93.903
13	97.437	33	97.317
14	95.712	34	94.710
15	97.076	35	97.268
16	94.022	36	95.471
17	93.720	37	93.140
18	95.436	38	94.428
19	93.175	39	93.752
20	96.474	40	95.167

results after years of development. First, the top-level design is not in place. The top-level design is one of the most critical influencing factors for the failure of universities to efficiently and smoothly carry out the new multimedia teaching reform. School management must fully realize the necessity, system, and complexity of “hybrid” teaching reform. According to the research team’s understanding, some colleges and universities have carried out “mixed” teaching reforms for many years, but they are still limited to a few classes of a certain profession led by a certain teacher in the initial experiment, and the reform effect has been minimal. The main reason is that the management only regards

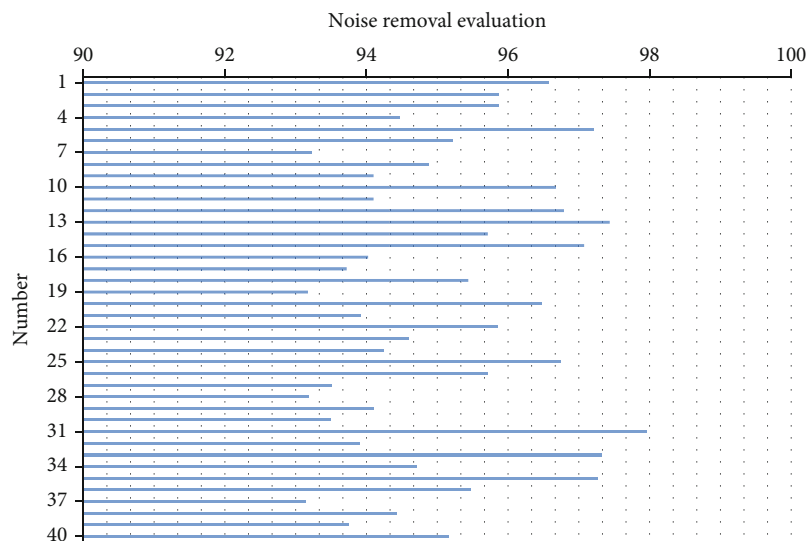


FIGURE 10: Statistical diagram of the effect of the high-order cumulant SNR estimation method.

blended teaching as an attempt and has not seen that “blended” teaching is the inevitable direction of teaching reform in colleges and universities. Second, the hardware and technical guarantees are insufficient. Compared with traditional classrooms, new multimedia teaching is highly dependent on the overall construction of “network+computer/mobile terminal.” At present, college students have basically realized that each person has a smart phone. Even if they do not have a computer, they can use the smart phone client for online learning. However, because ordinary mobile data access is often expensive, students using the mobile network to study are equivalent to increasing the financial burden of students. In addition, the implementation of new multimedia teaching puts forward higher technical requirements for teachers, especially older teachers with high professional titles and teaching management departments, and it undoubtedly increases the workload of teachers and administrators.

Based on the above analysis, this paper combines multimedia technology and applies embedded processors to intelligent English listening teaching to build an intelligent system to solve the current problems in English listening teaching and improve the effect of English teaching.

The frame structure of the mobile streaming media server is shown in Figure 6. The working process is briefly described as follows: The system separates the audio and video data streams of English listening. The audio encoder and the video encoder are used to encode, respectively, and the streaming media data is transmitted one-to-one or one-to-many by RTP. Real-time Transmission Control Protocol (RTCP) provides a reliable transmission mechanism for sequentially transmitting data packets and provides flow control or congestion control. The Wireless Application Communication Protocol (WAP) technology is a standard for mobile terminals to access wireless information services. The user obtains the audio file list, audio file introduction, and audio file data by communicating with the WAP server. Moreover, it responds to the WAP server through the Hypertext Transfer Protocol (HTTP).

TABLE 2: English listening teaching effect of multimedia intelligent-embedded processor.

Number	Teaching evaluation	Number	Teaching evaluation
1	89.730	21	83.676
2	87.008	22	89.250
3	88.109	23	89.125
4	86.330	24	89.903
5	89.669	25	84.251
6	91.903	26	89.314
7	90.358	27	83.889
8	90.043	28	90.088
9	85.478	29	85.581
10	87.510	30	92.598
11	91.046	31	85.521
12	88.561	32	87.329
13	89.814	33	84.669
14	85.498	34	87.509
15	83.896	35	83.304
16	90.657	36	91.257
17	89.571	37	92.427
18	90.425	38	82.733
19	84.397	39	89.935
20	88.743	40	89.445

On the basis of absorbing the advanced technology of other audio and video on-demand systems, this paper combines many years of English listening teaching experience and the characteristics of English listening learning to construct an English listening on-demand system based on streaming media technology to provide comprehensive solutions and technical support for software development. The English listening streaming media on-demand system is mainly composed of three parts: a comprehensive service

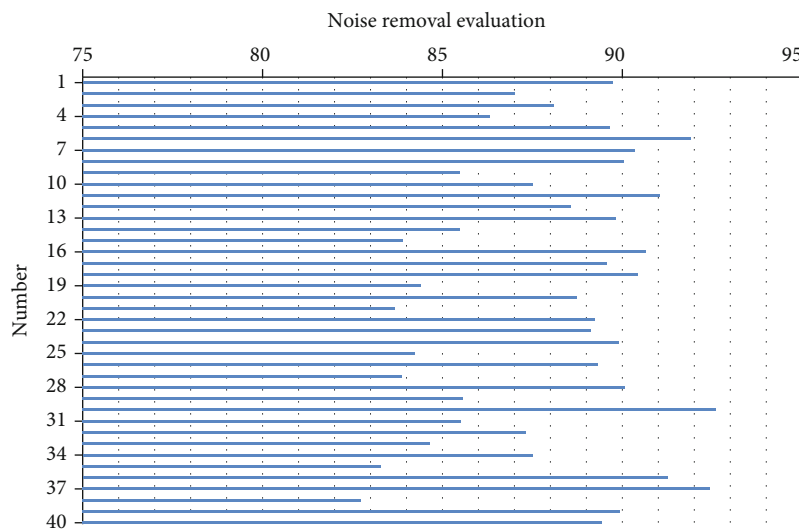


FIGURE 11: Statistical diagram of English listening teaching effect of multimedia intelligent-embedded processor.

platform, a switching routing network, and a user terminal. The system structure is shown in Figure 7.

The overall system architecture is shown in Figure 8. The front UI interface is displayed as three module entrances, which is a WPF design method. The realization of UI operation needs the support of BLL (business logic layer). DAL (data access layer) is the operating layer of the database, providing data for the business logic layer or presentation layer.

The function summary of the English listening test system is shown in Figure 9:

The system constructed above can effectively improve the current problems in English listening teaching and can test the teaching effect through the simulated listening test function. Next, the effect of the department of this paper will be evaluated through experimental research.

This paper evaluates the effect of the signal-to-noise ratio algorithm constructed in this paper in the processing of English listening sound waves through simulation experiments. The statistical results are shown in Table 1 and Figure 10.

The above experiments verify that the high-order cumulant signal-to-noise ratio estimation method is very effective in English listening speech processing. After that, the teaching effect of the English listening teaching system based on the multimedia intelligent-embedded processor constructed in this paper is evaluated, and the results shown in Table 2 and Figure 11 are obtained.

Through the above experimental analysis, it can be seen that the multimedia intelligent-embedded processor English listening teaching system constructed in this paper can play an important role in listening teaching and effectively improve the existing problems in listening teaching.

5. Conclusion

Listening teaching is a long-term comprehensive skill training process with strong continuous inertia. Relying on short-term intensive listening training cannot improve listening level in any way. Because of the different grades in middle

school and university, foreign language teaching is basically test-oriented education, focusing on the dissemination of foreign language knowledge, neglecting the cultivation of foreign language ability, and reading and listening and speaking. This has caused the students to have defects in listening. One of the most important manifestations is that there are very few listening classes at each stage, and at the internship stage, English learning has been completely put aside. Because there is no regular learning and training, it is impossible to improve listening skills in any way. This paper combines the multimedia intelligent-embedded processor to analyze English listening teaching and constructs an intelligent English listening model to improve the effect of English listening teaching. The experimental analysis shows that the multimedia intelligent-embedded processor English listening teaching system constructed in this paper can play an important role in listening teaching and effectively improve the existing problems in listening teaching.

Data Availability

The labeled dataset used to support the findings of this study are available from the corresponding author upon request.

Conflicts of Interest

The author declares no competing interests.

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