Intelligent Recognition Method of Short Wave Communication Transmission Signal Based on the Blind Separation Algorithm

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Abstract. The traditional signal recognition method can not be quickly and efficiently identified by noise interference. In order to avoid the drawbacks of traditional methods, an intelligent identification method for short-wave communication transmission signals based on blind separation algorithm is proposed. According to the mathematical model, all the transmission signals in short-wave communication are modally decomposed, and the signal can be decomposed into functions of several different feature scales, and the time and frequency are extracted as the physical quantities of the signal characteristics. The blind separation algorithm is used for signal preprocessing. The short-time energy, short-term average amplitude and short-time zero-crossing rate are used as the starting point of the recognized speech signal. Under the fixed background noise, the normal signal and the noise signal are identified. It can be seen from the experimental results that the method has short recognition time and fast rate, which lays a foundation for short-wave communication transmission.

Keywords: Blind separation algorithm · Shortwave communication · Transmission signal · Intelligent identification

1 Introduction

The rapid development of communication technology, the specification of communication is also constantly improving, and short-wave is the earliest developed and utilized radio frequency band. Short-wave communication is one of the oldest modern communication means. When people continue to develop radio frequency bands such as ultrashort waves and microwaves in pursuit of communication capacity and various services, traditional long-wave and medium-wave communication can only be applied in special occasions because they lose their original advantages [1]. As a modern communication technology, short-wave communication, despite decades of development, still maintains its vitality with its flexibility, simple equipment and long communication distance. However, due to the large number of bad noises and many types of signals in short-wave communication, the work of the receivers is seriously affected [2].

The early signal identification method is a demodulator using a series of different modulation methods. After receiving the high frequency signal, the high frequency signal is converted into an intermediate frequency signal, and then input to each

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demodulator to obtain an observable or audible signal, and then the operator uses the earphone. Analytical recognition by oscilloscope or spectrum analyzer. The identification of manual participation requires an experienced operator. Generally, it can successfully identify the amplitude keying signal with longer duration and lower symbol rate, and can modulate the frequency-shifted keying signal with larger index, but cannot recognize the phase shift. Keying signal. This kind of manual participation identification method, the judgment result including the subjective factors of people, will vary from person to person, and the types that can be identified are also limited [3].

In order to achieve a satisfactory communication effect for short-wave communication, an intelligent identification method for short-wave communication transmission signals based on blind separation algorithm is proposed. The method has advantages in communication, biomedical signal processing, speech signal processing, array signal processing and general signal analysis. Wide application prospects [4]. It not only can effectively process signals, but also plays an active role in the development of neural network theory.

2 Research on Intelligent Recognition Method of Shortwave Communication Transmission Signal

Blind separation algorithm identification is based on the blind separation principle to identify each signal. In the recognition process, the combination of signal detection and estimation, feature selection, classification and recognition, etc., and the content itself constitute a huge theoretical system. The traditional recognition method has more constraints, so that the disturbance signal can be effectively recognized [5].

2.1 Mathematical Model

The research contents of blind separation can be divided into four parts: instantaneous linear mixing blind separation, convolutional aliasing blind separation, nonlinear aliasing blind separation and the application of blind separation. When the aliasing model is nonlinear, it is difficult to recover the source signal from the aliasing data unless there is further prior knowledge about the signal and the aliasing model [6].

Figure 1 is a schematic diagram of an instantaneous linear mixed-blind separation signal model.

In Fig. 1, the source signal $S = [\mathrm{s}_1(t), \mathrm{s}_2(t), \ldots, \mathrm{s}_\mathrm{n}(t)]^T$ is an unknown n-dimensional source signal vector, and A is an unknown mixing matrix, $m = [\mathrm{m}_1(t), \mathrm{m}_2(t), \ldots, \mathrm{m}_\mathrm{n}(t)]^T$ is an m-dimensional noise vector, $X = [\mathrm{x}_1(t), \mathrm{x}_2(t), \ldots, \mathrm{x}_\mathrm{n}(t)]^T$ is the m-dimensional observed signal vector of the sensor output, which has:

$$X = AS + \mathrm{m} \tag{1}$$

The blind separation algorithm requires that only X be known to determine S or A. Independent component analysis is a kind of base station subsystem, and its basic meaning is to decompose the signal into several independent components [7]. In Fig. 1, the goal of independent component analysis is to find a separation matrix, and then

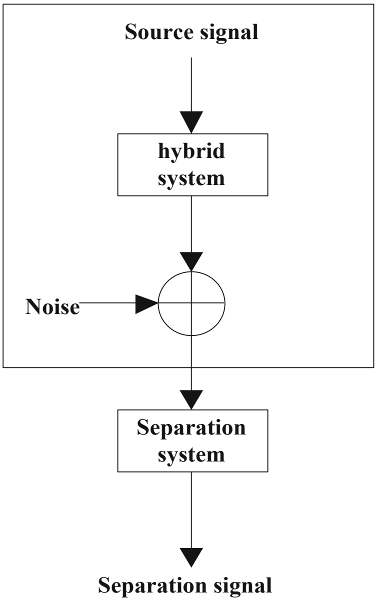

Fig. 1. Schematic diagram of instantaneous linear aliasing blind separation signal model

transform the matrix to obtain a new vector whose components are as independent as possible. The separated signal vector to be sought, that is, the estimated value of the source signal [8].

2.2 Mode Decomposition

Modal decomposition of all transmitted signals in short-wave communication can be used to decompose the signal into functions of several different feature scales. These functions satisfy the following conditions:

① The number of extreme points and zero crossings cannot exceed 1;

② At any point position, the average value of the data extremum is zero [9].

The established signal decomposition process is as follows:

(1) The upper and lower envelopes can be obtained by fitting all the extreme points of the signal, obtain the difference between the fitted signal and the envelope mean value;
(2) The difference obtained in step (1) is taken as a new fitting signal, and the above steps are repeated until the difference satisfies the eigenmode function condition, and the first decomposed modal component is consistent with the difference size.
(3) Classify the modal components from the original signal, obtain a new fitting signal, repeat steps (1) and (2), and obtain modal components of different feature scales until the new fitting signal shows a monotonic trend [10, 11].

2.3 Feature Selection

Short-wave communication is a time-varying, non-stationary random process. Short-wave communication has relative stability in a short time, and its characteristics are considered to be constant. Therefore, short-wave communication transmission signals have short-term stability [12]. In the actual environment, in addition to broadband white noise, there are periodic noises or impulse noises such as machine roar, gun sound, etc., so the audio signal can be divided into mode noise, non-white noise and broadband white noise. The real-time waveforms of noise and normal signals can be seen to be significantly different and can be identified exactly. This indicates that the transmitted signal and noise occupy the same frequency band in the frequency domain, but there is a significant difference in the time domain. Therefore, the difference between the transmitted signal and the noise short-time autocorrelation function value can be used to extract the pattern features.

When the modal decomposition signal satisfies the routing information protocol standard, the acquired source signal can retain the original signal characteristics. Since the acquired source signal is a modally decomposed signal, the coefficient vector can be restored by orthogonal matching to obtain a conjugate complex number. Before the feature extraction, the source signal is subjected to a modulus conversion process to remove the conjugate complex number. The modulus values in the array are arranged from large to small and combined into a new source signal.

Using time and frequency as the physical quantities to characterize the signal, based on the new source signal, reveal the essential characteristics of the disturbance signal with fast frequency change, and construct the time domain and frequency domain bridge by blind separation to ensure that the signal characteristics can be in the region. show. The abnormal signal is introduced in the short-wave communication in time, and the normal signal and the abnormal signal can be distinguished in the aliasing phenomenon, and the blind disturbance signal pre-processing is performed on the normal disturbance signal that is not affected by the noise.

2.4 Blind Separation Signal Preprocessing

In the process of blind separation signal processing, in order to reduce the amount of calculation and improve the transmission efficiency, it is usually necessary to undergo preprocessing. Pre-processing generally includes centralization and whitening. Centralization is to make the mean value of the signal zero. Since the data obtained under normal conditions are related, it is usually required to perform preliminary whitening of the data because the whitening process can be removed. The correlation between the observed signals, thereby simplifying the extraction process of subsequent independent components.

Generally, the whitening of the data is better than the whitening of the data, and the algorithm has better convergence, less workload, and higher efficiency. The linear aliasing blind separation signal model generally adopts the method of independent component analysis. The main basis and premise of ICA is to assume that the source signal is independent. Therefore, it is naturally conceivable that the first step of the ICA algorithm is to establish an objective function to characterize the separation result. The degree of independence. After the objective function is determined, it can be optimized by various optimization algorithms to determine the separation matrix. The representative algorithms mainly include the maximum information method, the natural gradient method, the fast independent element analysis algorithm, the matrix eigenvalue decomposition method, etc. In the blind separation, the optimization operation is often used. As far as the optimization method is concerned, the algorithm and the natural gradient algorithm belong to the gradient descent (rise) optimization algorithm. The convergence speed is linear and the speed is slightly slower, but it belongs to the adaptive method and has Real-time online processing capability; independent component analysis algorithm is a fast and numerically stable method. It uses the quasi-Newton algorithm to achieve optimization and has super-linear convergence speed. Generally, the convergence speed is much faster than the gradient-decreasing optimization algorithm; Matrix Eigenvalue decomposition methods usually estimate separated Matrices by Eigen-decomposition or Generalized Eigen-decomposition of Matrices. This is an analytical method, which can directly find the formal solution. Since it has no iterative optimization process, it runs fastest.

2.5 Recognize the Starting Point of a Speech Signal

The basic short-time parameters of transmission signal are: short-time energy, short-time average amplitude and short-time zero-crossing rate.

$$\begin{aligned} E &= \sum_{\mathrm{n}=0}^{S=1} L_{\mathrm{g}}^{2}(n) \\ F &= \sum_{\mathrm{n}=0}^{S=1} \left| L_{\mathrm{g}}(n) \right| \end{aligned} \tag{2}$$

$$Z = \frac{1}{2}\left\{ \sum_{\mathrm{n}=1}^{S=1} \left| \mathrm{sgn}\left[L_{\mathrm{g}}(n)\right] - \mathrm{sgn}\left[L_{\mathrm{g}}(n-1)\right] \right| \right\} \tag{3}$$

In formulas (2) and (3): $L_{\mathrm{g}}(n)$ is windowed transmission; S is window length; and E, F, Z represent short-term energy, short-term average amplitude and short-time zero-crossing rate, respectively.

The short-time parameter can be used to identify the starting point position of the short-wave communication transmission signal. Firstly, the probability density function in the case of a turbid transmission signal can be known through training. According to this, a threshold parameter can be determined. When the short-term parameter value of a frame input signal exceeds the threshold parameter, it can be confirmed that the frame signal is not silent, but May be voiced. According to the threshold parameter, it can be determined that the two points in the input signal are definitely between the two points, but the precise start and end points of the voice are also searched before and after the two points. To do this, set a low threshold parameter, which is forwarded by one of the points. This point can be determined when the short-term parameter value is reduced from large to small to the threshold parameter. Similar to this, by looking backwards from another point, you can determine the point, and it is still a voice segment between the two points.

Then, the search is performed with the short-term zero-crossing rate from the first point forward and the second point backward. According to the mean value of the short-time zero-crossing rate under the silent condition, a new parameter of short-time zero-crossing rate is set. If the short-time zero-crossing rate is always greater than 3 times of the new parameter when searching for the first point forward, these signal frames are considered to be It still belongs to the speech segment until the short-term zero-crossing rate suddenly drops to a new parameter below 3 short-time zero-crossing rate, at which point the exact starting point of the transmitted signal can be determined, and similar processing can be performed for the end point.

Recognizing the origin of speech is not only important for extracting data features, but also can separate speech more correctly. Because if the signal source has a silent part during the blind separation of the audio signal, the separated sound signal will have a silent part. The aliasing of other source signals is not conducive to the extraction of pure sound.

2.6 Transmission Signal Intelligent Identification

The purpose of transmitting signals in short-wave communication is to separate the transmitted signal from the received signal. Under fixed background noise, as long as the relative difference between the transmitted signal and the specific noise is extracted, the transmitted signal can be well recognized, but under varying background noise, it is necessary to find out between the transmitted signal and all other various noises. The difference in characteristics is not easy, because short-wave communication transmits signals and noise as a type of wave, and they have many commonalities. Especially in the case of low signal-to-noise ratio, the energy of most short-wave communication signals is very low relative to noise. Therefore, among the characteristics reflected by the analyzed signals, the characteristics of noise are the main components. In this case, it is particularly difficult to identify short-wave communication signals. Because it is a real-time short-wave communication signal identification method, the requirements for calculation amount are strict, and it is impossible to adopt an overly complex algorithm. According to this situation, the characteristics of noise can be classified according to the fact that some noise is in the time domain. The characteristics are obvious, and some noise features are more obvious in the frequency domain.

Under the optimal window, the blind separation algorithm can be used to cluster all the disturbance signals to the central position. Through signal conversion, all the transmission signals are collected together, which is convenient for users to obtain the demand signals. The specific implementation steps are as follows:

Assuming that the total number of extracted signals is m, and the amount of information contained in each signal is t, then the weight coefficient in the k th signal is x_k, and a coordinate point in the three-dimensional space is described according to the following formula. Disturbance signal recognition function under the influence of noise.

The disturbance signal identification function under the influence of noise is minimized to ensure that each signal has a fixed weight coefficient and fuzzy exponent in the process of clustering. The specific processing process of blind signal separation is as follows:

① Statistically disturb the number of signals, using blind separation processing, so that the signal blind separation index and weight coefficient reach the extreme value, and the processing threshold and the number of iterations reach the maximum value, and establish the signal initial matrix and the weight coefficient matrix;

② According to the weight coefficient matrix, the value of the objective function is set, and the number of times after the iterative processing is set is greater than the maximum value, then the selected objective function value will be smaller than the signal threshold, and the calculation can be stopped;

③ Error compensation for the center position of the blind separation space;

④ Using an iterative processing method to obtain a new blind separation center;

⑤ Update the signal attribute weight coefficient and return to step ②.

According to the above steps, the signal can be subjected to blind separation processing, and a blind separation algorithm is used to effectively identify the noise disturbance signal.

3 Check Analysis

In order to verify the rationality of the intelligent identification method of the short-wave communication transmission signal based on the blind separation algorithm, the following experiments are carried out.

3.1 Experimental Results and Analysis

3.1.1 Recognition Time

The identification time of the short-wave communication transmission signal is used as the standard of the transmission quality detection, and the recognition time is compared and analyzed by using the traditional identification technology and the identification technology based on the blind separation algorithm, and the result is shown in Fig. 2.

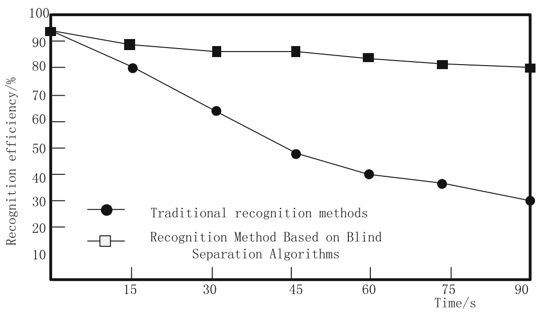

Fig. 2. Analysis of comparison of recognition time between two techniques

It can be seen from Fig. 2 that when the time is 15 s, the recognition efficiency by traditional technology is 80%, the recognition efficiency based on the blind separation algorithm identification technology is 90%; when the time is 45 s, the recognition efficiency by the traditional technology is 55%, based on blindness. The recognition efficiency of the separation algorithm identification technology is 88%; when the time is 75 s, the recognition efficiency by the traditional technology is 38%, and the recognition efficiency based on the blind separation algorithm identification technology is 82%. It can be seen that the recognition efficiency based on the blind separation algorithm identification technology is high, indicating that the communication signal transmission quality is good.

3.1.2. Recognition Rate

In the same way, the recognition time is compared and analyzed by using the traditional recognition technology and the identification technology based on the blind separation algorithm, and the result is shown in Fig. 3.

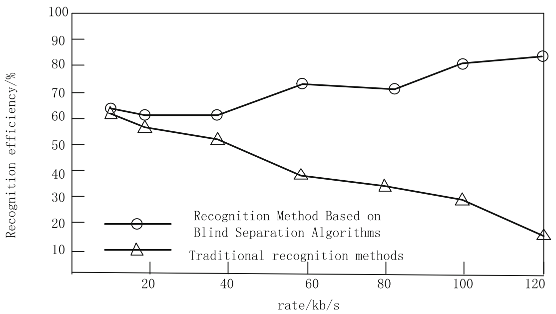

Fig. 3. Analysis of comparison results of recognition rate between two techniques

It can be seen from Fig. 3 that when the rate is 20 kb/s, the recognition efficiency is 55% by the conventional technique, the recognition efficiency based on the blind separation algorithm recognition technique is 65%, and when the rate is 60 kb/s, the recognition efficiency is determined by the conventional technique. For 38%, the recognition efficiency based on blind separation algorithm recognition technology is 75%; when the rate is 100 kb/s, the recognition efficiency is 32% by traditional technology, and the recognition efficiency based on blind separation algorithm recognition technology is 80%. It can be seen that the recognition efficiency based on the blind separation algorithm identification technology is high, indicating that the communication signal transmission quality is good.

3.2 Experimental Conclusions

The efficiency of communication signal recognition is verified and analyzed in two aspects: recognition time and recognition rate. The comparison results show that the recognition efficiency based on blind separation algorithm is higher, which shows that the communication signal transmission quality is better.

To sum up: the research on intelligent identification method of short-wave communication transmission signal based on blind separation algorithm is reasonable.

4 Conclusion

According to the actual environment of short-wave communication signal transmission, the short-wave communication signal recognition situation is analyzed. Therefore, researching signal recognition efficiency becomes a necessary condition for short-wave communication. Although this technology has higher recognition efficiency, it is still to be investigated for signal recognition in different environments. Therefore, in the future research work, the signal recognition in different environments is deeply studied.

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