

Speech enhancement system design is based on FastICA blind source separation algorithm

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Abstract. Based on the analysis of existing speech enhancement based on the drawbacks of the traditional algorithm is adopted in the system is proposed based on FastICA blind source separation algorithm design of speech enhancement algorithm, and the transplanted to embedded speech enhancement system. The system real-time speech enhancement, by four element microphone arrays is used to sample the space of sound signal and through the built-in speech enhancement algorithm to the voice source signal and noise source signal separation, capable of suppressing co channel noise, active noise and residual background noise. Based on the test results of the algorithm on PC platform, the system meets the design requirements and the effect is good.

Keywords: Blind source separation algorithm; Speech enhancement system; Active noise and noise suppression.

1. Introduction

At present, speech enhancement is an important research topic in multimedia technology. Traditional speech enhancement methods: spectrum subtraction and the adaptive filtering method and minimum mean square error is mostly for voice and with the parameters of the channel noise outside noise characteristics, the characteristics of statistical properties of noise suppression, the active noise, especially with channel noise (target speech signal of the voice signal) inhibition of apparently helpless.

Here fore, this paper designs a speech enhancement system based on microphone array and blind source separation algorithm, which has the following advantages compared with the speech enhancement system composed of a single microphone and general speech enhancement algorithm:

1) The microphone array has the function of spatial filtering, which can suppress the background noise.

2) After the microphone array is collected by the blind source separation algorithm, its mixed signal can be separated into multiple independent signal components. Then, LPC complex cepstrum (LPCC) parameter characteristics can be used to separate the target signal components, so as to suppress the active noise and achieve speech enhancement.

According to the needs of the project and the comparison of blind source separation algorithms, FastICA algorithm is adopted as the blind source separation algorithm of this system. It is easy to implement in embedded system.

2. System hardware block diagram and principle

The main modules of this system are shown in Figure 1. Hardware design mainly includes microphone array, analog-to-digital converter, DSP module, digital-to-analog converter, audio equipment interface, display module and keyboard module.

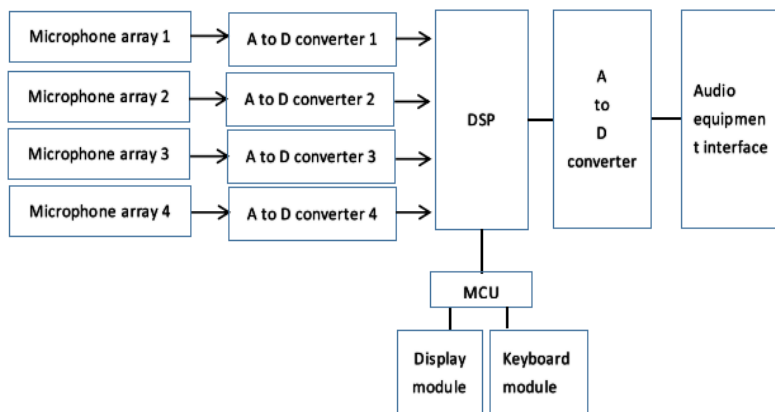


Fig. 1 System hardware block diagram

3. Design of main system hardware unit circuit

1) Microphone array module: it is composed of 4 microphone array elements, using op amplifier NE5532 as the front follower to reduce the output impedance. The design is shown in Figure 2.

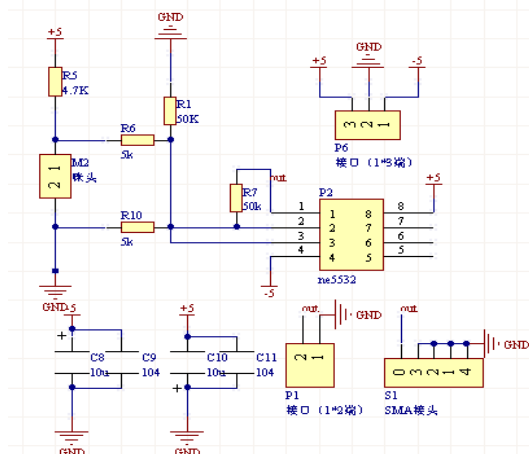


Fig. 2 Microphone array module block diagram

2) The design of analog-to-digital conversion module: PCM3000 is adopted to realize the simultaneous sampling and conversion of two channels of signals. The built-in filter has the highest precision up to 18 bits.

3) Audio equipment interface module: 3.5 dual-channel headphone socket is used to output audio signals. According to the selected signal frequency band, a passive low-pass filter of 4KHz is made for the output of the audio codec. Then, low-noise op amplifier OP07 is selected to connect to the 3.5 dual-channel headphone socket for output signals. The design is shown in Figure 3.

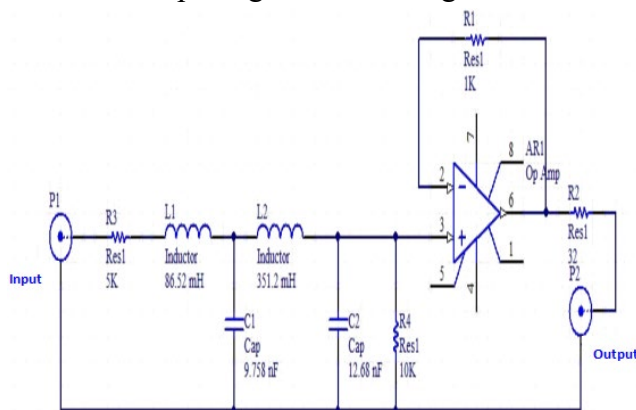
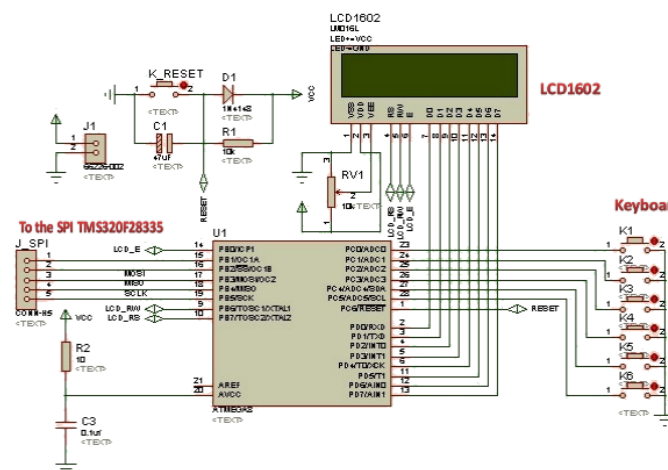


Fig. 3 Audio equipment interface diagram

4) Display module: LCD1602 is selected and can display 32 characters to meet the performance requirements of the system.

5) Keyboard module: 1X6 keyboard composed of six key switches. The circuit design of the display module and the keyboard module is shown in Figure 4.



The FIR low-pass filter with the center frequency of 3000Hz in Matlab is used for the pre-filter. Before the array signal is processed further, the pre-filter filters the signal collected by each array element to filter out the components outside the speech frequency and reduce the interference of the components outside the speech frequency, so that the subsequent algorithm can achieve better results.

3) FastICA blind source separator:

This article selects the FastICA the blind source separator is selection enough in the length of the microphone array signal analysis, the selection of money through pre - filter, to mean and bleaching, bleaching of array signal by FastICA algorithm can get the estimated separation matrix, multiplied by the treated microphone array signal vector, according to the results of its estimate of the independent source component.

LPCC Parameter Characteristics Analyzer:

The LPCC parameter feature analyzer conducts feature analysis on the previously separated multi-channel independent signal components one by one, and selects the signal components that are consistent with the target LPCC parameter characteristics as the target signal. Also according to the different application occasions, the number of signal components can be selected directly as the target signal.

5) Post filter:

In order to further suppress the residual background noise and improve the speech quality, the Karman filter is applied to the target signal obtained by the post filter.

5. Assessment and conclusion

According to the system design, the algorithm was tested on the PC platform, and the simulation test results were as follows:

1) Mute detector test

A speech signal with white noise is selected as the signal to be tested and detected by the silence detector. The results is shown in Figure 6.

The result of the pure white noise signal passing the mute detection is shown in Figure 7. By comparison, it can be concluded that the mute detector can accurately judge the speech signal.

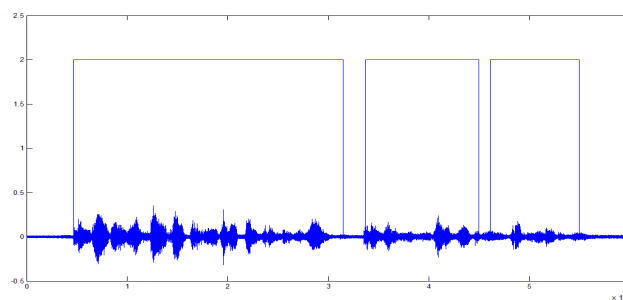


Fig. 6 Mute detection of noisy speech

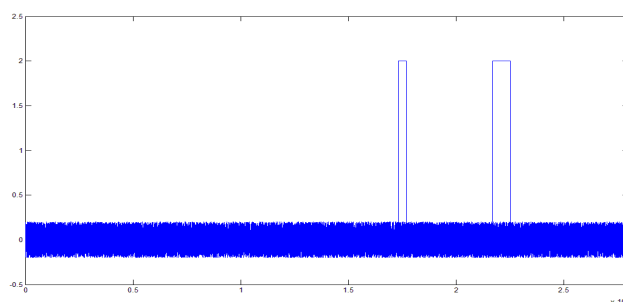


Fig. 7 White noise mute detection

2) FastICA blind source separation test

Select three speech files, read out three groups of speech signals, and form a speech matrix according to a straight array. The distance of the array element is set as 20cm, and its incident

directions are assumed to be 10, 20 and 50 degrees respectively. After generating array manifold and integrating with the signal matrix, a group of microphone array signals is obtained. The results after blind source separation by FastICA are shown in the figure.

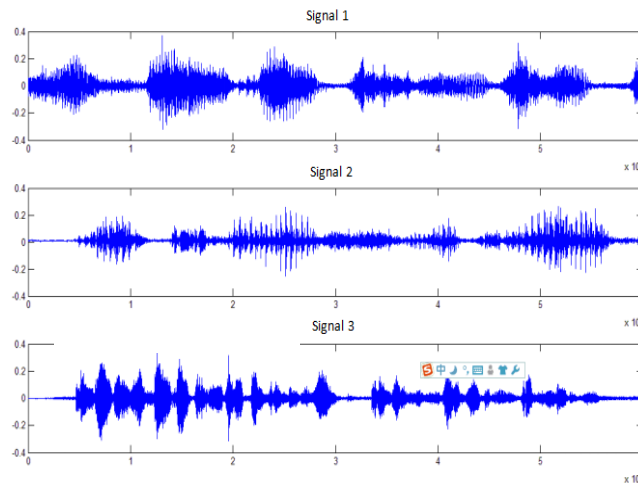


Fig. 8 Original signal

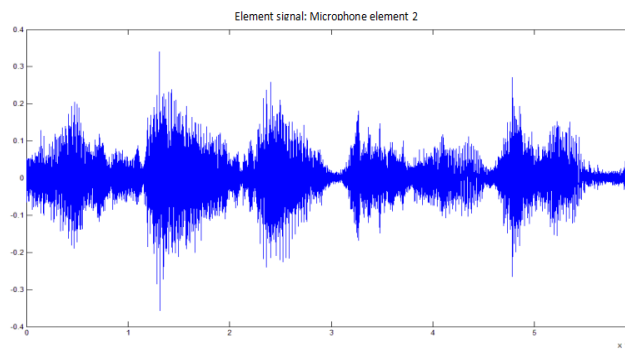


Fig. 9 Signals collected by the array element

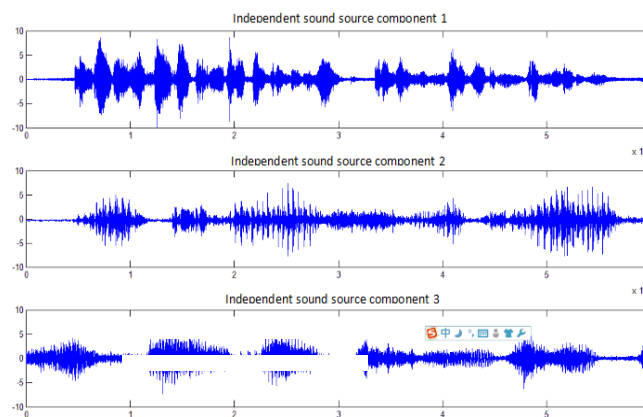


Fig. 10 The independent source component of the isolated array signal

The results showed that the FastICA blind source separator successfully separated the mixed signals. And independent sound source components 1, 2 and 3 correspond to signals 2, 3 and 1, respectively.

3) Post filter test

Gaussian noise is added to the 200Hz sinusoidal signal with a sampling rate of 9375Hz. After filtering by the post-filter, the result is shown in Figure 11. The post-filter has good noise suppression ability and can greatly improve the signal-to-noise ratio of the signal.

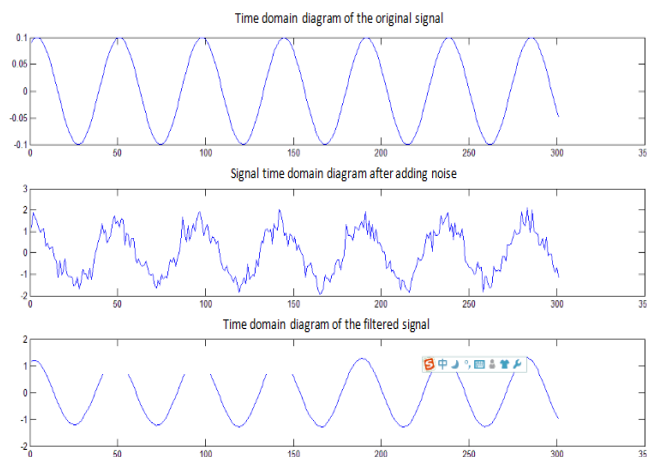


Fig. 11 Effect of post filter

Acknowledgements

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