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Research on Speech Signal Enhancement Algorithm Based on Matlab

Yachao Xing ,Hongli Zhu
School of Information and Electronic Engineering
Zhejiang University City College, Hangzhou, China
Corresponding Author: zhuhl@zucc.edu.cn

***Abstract*— Voice is another form of language. It is sent by a series of organs such as human lungs, vocal cords, oral cavity, etc. It plays an important role in the process of remote communication between people. But voice and language are very different. In people's lives and work, people communicate through language. In the process of remote communication, we have to reduce the loss in the transmission of voice signals, so as to achieve better language communication. For the enhancement of voice signals, there are many processing technologies, and each processing technology has its advantages and disadvantages. Therefore, enhancing the voice signal and maintaining the clarity of the voice signal are of great significance in real life. In the process of making the graduation project this time, first open the music through the quick audio converter and intercept part of it, and save the intercepted music to the desktop in the format of .wav. Read the voice signal through the audioread function in matlab and draw the time and frequency domain waveforms of the original signal, play back the original voice signal through the sound function, and then generate a high-frequency noise of a sine function through matlab and add the noise On the original signal, observe the changes in the time and frequency domain waveforms before and after noise addition and use the sound function to play back. Compare whether the speech signal has obvious changes before and after noise addition. Finally, design a filter, perform filter processing, and use the sound function to play back the voice.**

Keywords—voice signal; Signal processing; matlab

I. INTRODUCTION

The collection and processing of voice signals are two closely related disciplines, and they often appear at the same time. Voice signals are the most basic form of remote communication between people. However, in the process of long-distance transmission, it will be interfered by many factors, which will cause the effective voice information that people to extract at the receiving end will be reduced, which reduces the clarity of the voice signal. These factors that affect the voice signal can be collectively called noise.

In the process of voice signal transmission, it will be affected by many external factors. For example, during the propagation process, when encountering obstacles, a part of the voice signal will be lost, and some other unrelated voice signals will be lost. In severe weather, part of the voice signal will be lost. These are all we can think of, and there are many losses that we can't think of. At the receiving end, the size of the voice signal will change according to changes in the environment. However, these effects will cause the loss of the voice signal, resulting in a difference between the signal received at the receiving end and the transmitting end. At the same time, the effective voice information will decrease as the propagation distance increases. This is due to the interference of external noise. Therefore, it is very necessary to reduce the loss in the transmission route. However, the enhancement processing of the speech signal is one of the effective methods to solve the above problems.

II. COMMONLY USED SPEECH ENHANCEMENT ALGORITHMS

A. *Adaptive noise cancellation method*

The adaptive noise cancellation method can track the constantly changing speech signal in an unfamiliar signal environment and estimate the characteristics of the noise through its own filter. So as to achieve the filtering of the noise. The principle of the method is as follows Shown:

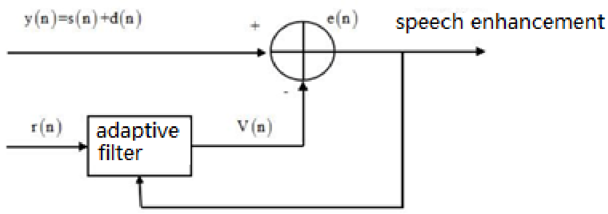

Fig. 1 Structure diagram of adaptive noise cancellation method

s(n) is the original speech signal, d(n) is the noise, y(n) is the speech signal after adding noise, r(n) is the input value of the filter, r(n) is output through the adaptive filter v(n), that is, v(n) is the output value of the filter. By tracking the noised signal, the change of d(n) can roughly estimate the various change characteristics of the noise d(n), thereby changing the filter's Input and output to achieve the effect of voice signal enhancement. In this method, the more accurate the noise signal d(n) is estimated, the better the effect of speech signal enhancement and the more obvious the noise signal elimination. When the output v(n) produced by the adaptive filter is exactly the same as the noise signal d(n), the noise signal can be completely cancelled out. At this time, the noise reduction effect is the best and the intelligibility of the speech signal is improved.

B. Wavelet transform

Wavelet transform appeared after 1980 as a time-domain analysis method. Its appearance has a certain relationship with Fourier transform, which is equivalent to Fourier transform. Wavelet transform has essential progress. The Fourier transform cannot combine the time domain and the frequency domain of the speech signal. But separate them separately and study them one by one. Therefore, the Fourier transform cannot describe the overall characteristics of a signal and can only study one characteristic alone. After the research is over, we can study another characteristic. For signals that are not very stable, the best way is to put all the characteristics together to see the overall characteristics of the speech signal more intuitively. The leaf transform is reformed, and then the wavelet transform is produced. With the generation of wavelet transform, the shortcomings and shortcomings of Fourier transform are overcome. Wavelet transform has the characteristics of multi-resolution analysis, that is, no matter in the high-frequency or low-frequency part of any signal, you can get a resolution you want through analysis, so as to extract more effective information. For those speech signals that are not stationary, wavelet transform is a very good research method. It can overcome the problems that Fourier transform cannot solve. With the emergence of wavelet transform, people have got better development in the fields of speech signal, digital graphics and image.

C. Spectral subtraction

Spectral subtraction is to roughly estimate the power spectrum of noise and subtract the estimated power spectrum from the noise-added speech signal, so as to achieve the equalization effect of the noise in the speech signal. However, in practical applications, spectral subtraction has many shortcomings and drawbacks. If the power spectrum of the original speech signal and the noise signal are close or similar, it is difficult to accurately estimate the power spectrum of the noise signal. Accurate, and may have a great impact on the original signal. Therefore, the traditional spectral subtraction noise removal effect is not very ideal.

Traditional spectral subtraction is very important for the estimation of noise power spectrum, because once the noise power spectrum estimation is inaccurate, it will affect the effect of speech signal enhancement and the quality of speech signal. In severe cases, additional noise signals will be generated. This extra noise signal is caused by inaccurate frequency spectrum estimation. Therefore, traditional spectral subtraction has many limitations.

Later, in order to improve the problems of traditional spectral subtraction and reduce the generation of this extra noise as much as possible, they proposed auditory masking effects, minimum mean square error estimation methods, logarithmic spectrum estimation and other methods, making the original spectrum subtraction method With a certain improvement effect, the application is relatively wider than before.

III. FILTER DESIGN

By observing the time-domain and frequency-domain waveforms of the original speech signal, the time-domain and frequency-domain waveforms of the speech signal after adding noise, see what changes in the frequency domain before and after adding noise, and then design a filter by adjusting the cutoff frequency of the filter , Change the characteristics of the filter, perform noise reduction processing on the noise-added speech signal, and obtain the time-domain and frequency-domain waveforms of the noise-reduced speech signal. Compare with the one before filtering, analyze the change of the voice signal, and play it back through the sound function. By changing the cut-off frequency, a set of parameters with better noise reduction processing effect can be obtained.

IV. VOICE SIGNAL PROCESSING AND RESULTS

Through self-designed filters, noise reduction is performed on the noisy speech signal. This function can be achieved through the filter function. The time-domain and frequency-domain waveforms of the speech signal after noise reduction are compared with the time-domain and frequency-domain waveforms before filtering, and the changes between them are analyzed, and the filtering is felt through the sound function. Before and after the audio signal changes, if the voice signal after adding noise does not change significantly, it means that the effect of speech enhancement is not particularly good. You can change the effect of noise reduction by changing the cut-off frequency until a large amount of noise can be eliminated.

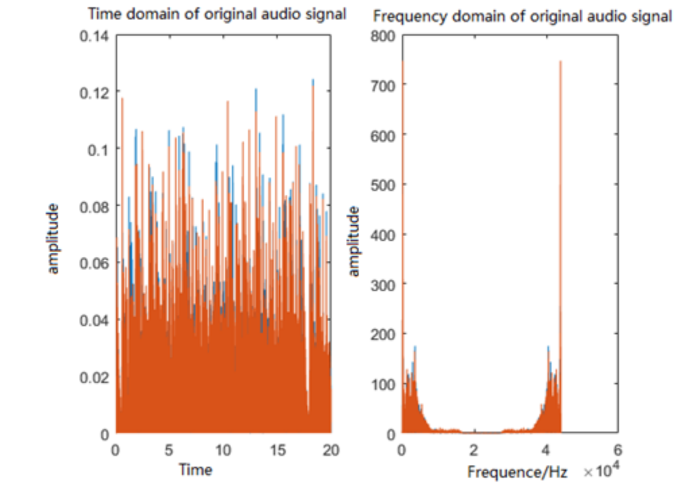

Fig.2 Time domain and frequency domain waveforms of the original audio signal

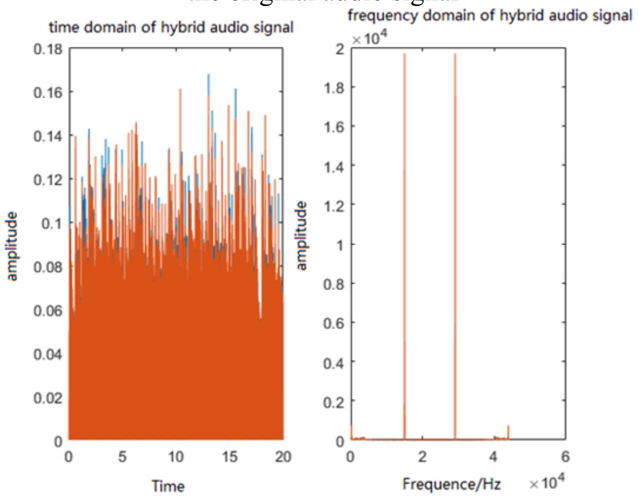

Fig. 3 Time domain and frequency domain waveforms of sinusoidal noise

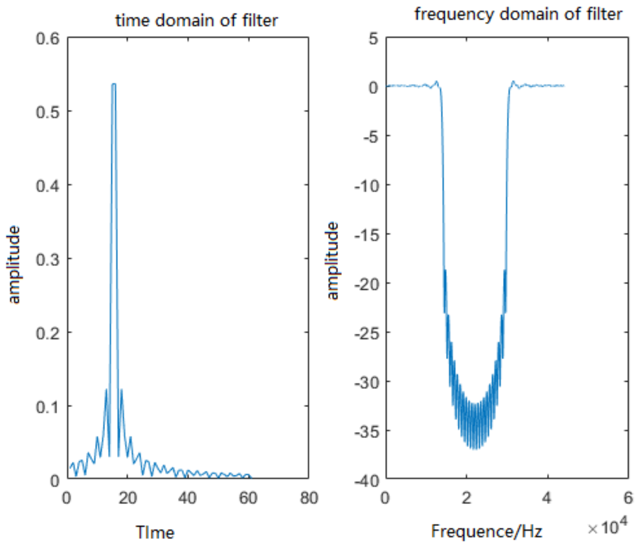

Fig. 4 Time domain and frequency domain waveforms of the filter

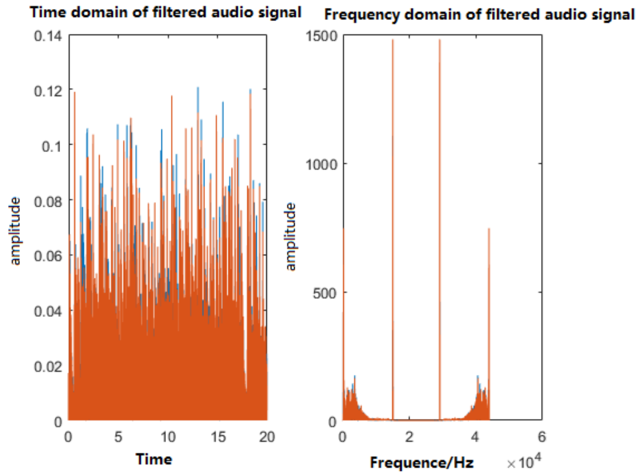

Fig. 5 Time domain and frequency domain waveforms after filtering

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