

# Study of Speech Digital Signal Processing based on Matlab

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**Abstract.** Language is a common tool for information exchange. Speech acquisition, preservation and processing make great progress in the field of communication. The emergence and development of digital signal processing technology has a positive role in promoting the development of communication technology. As an interactive high-level language, MATLAB has been widely used in various scientific fields due to its advantages in signal processing and matrix budget. Therefore, the analysis of speech signal de-noising processing and simulation based on MATLAB has certain practical significance.

**Keywords:** digital signal processing; MATLAB.

## 1. Introduction

After the application of this software in speech signal processing, people can start with the generation mechanism and information characteristics of voice information, and build an integrated voice research method, which can provide some theoretical support for the development of voice online recognition technology.

## 2. Acquisition of Noise Signal

Select an arbitrary section of audio, use the audio read function in the MATLAB toolbox to read the audio file, and record the audio sampling frequency and sampling points at the same time. Intercepting audio files to a reasonable length is convenient for MATLAB digital signal processing. Gaussian white noise is generated by AWGN () or WAN () function; Gaussian white noise is used as narrow-band Gaussian white noise by narrow-band filter; and single-frequency sinusoidal wave is used as single-frequency interference. Three kinds of noise of corresponding amplitude are added to the original audio to produce corresponding noise signal. The time-frequency waveforms of three kinds of speech signals are drawn, and the similarities and differences between the original signal and the noised signal are compared.

## 3. Noise Signal Processing

### 3.1 Gauss White Noise

For speech signal processing with Gauss white noise, according to the spectrum characteristics of the original signal, a low-pass filter is used to denoise the Gauss white noise. The frequency components of the original audio signal are mostly below 3000Hz, so the low-pass filter with cut-off frequency below 3000Hz can be used to filter the high-frequency components of the noise signal. Here, IIR filters are used to design low-pass filters.

### 3.2 Narrow Band Gauss White Noise

The frequency range of narrowband noise concentration is obtained by comparing the original audio signal with the noise signal. The narrowband Gauss white noise is filtered out by designing a reasonable band-stop filter. The band-stop filter can be designed by band-pass filter.

### 3.3 Single Frequency Interference Noise

Observe the signal spectrum after adding noise and determine the frequency range of noise. A band-stop filter is designed to filter out the noise signal and obtain the pure audio signal. At this time,

the effect of FIR filter is not obvious enough. The amplitude-frequency response of Chebyshev band-stop filter attenuates quickly, which is suitable for filtering single-frequency noise interference.

### 3.4 Noise Elimination in Conference Site

For noise in special environment such as meeting place, because of the correlation of noise, adaptive filter based on LMS is usually used to denoise. Through iteration, the weights are revised continuously, and the noise in the noise frequency is filtered gradually. Finally, a relatively pure audio signal is output.

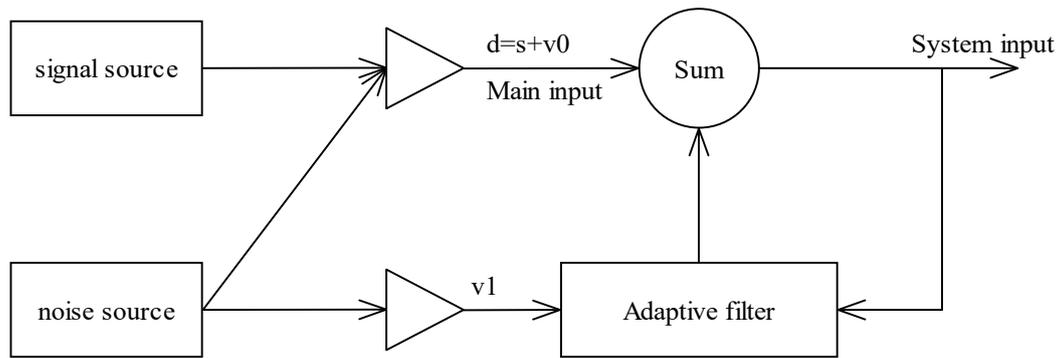


Fig.1 Noise Elimination in Conference Site

## 4. Simulation Implementations

Corresponding filters are used for different noise signals. For Gauss white noise, a reasonable low-pass filter is used; for narrow-band Gauss white noise, a band-stop filter is used; for single-frequency interference noise, a Chebyshev filter with very narrow stopband is used. By determining the cut-off frequencies of passband, stopband and attenuation of passband and stopband, corresponding filters can be designed, and then the signal with noise frequency can be filtered to get the de-drying audio.

$$[Bz,Az] = \text{lvboqi}(fp,fs,'low');$$

$$y1 = \text{filter}(Bz,Az,aa(:,1));$$

fp is the cut-off frequency of passband, fs is the cut-off frequency of stopband, FP and FS can be an array, thus the corresponding low-pass filter can be obtained.

After testing, when the real audio is mixed with strong noise or even submerged with the real signal, the real signal can still be recovered well through LMS adaptive filter.

## 5. Gui Design

In order to facilitate data testing, the GUI function in MATLAB is used to add reasonable controls to facilitate the experiment.

### 5.1 Signal Acquisition

Through the audioread () function, select a specific audio file in the folder, and intercept a specific length into the MATALB processing.

### 5.2 Noise Signal Realization

By using pop-up menu to select the type of noise to be added, the signal-to-noise ratio of Gauss white noise needs to be determined simultaneously when adding Gauss white noise. The input number is determined by reading function of GUI dynamic text box in MATLAB toolbox.

$$a = \text{get}(\text{handles.popupmenu1}, 'value');$$

The return value of the control is obtained by the function, and the noise type selected in the pop-up menu is determined by the return value. According to the noise type, the corresponding noise is added to the original audio signal.

### 5.3 Waveform Display in Time-Frequency Domain

By setting appropriate controls and utilizing the existing functions in the toolbox, the original signal, the noise-added signal and the signal spectrum after filtering and denoising are displayed, and their similarities and differences are observed in the time-frequency domain.

## 6. Epilogue

In a word, different kinds of filters can be selected for different noise audio signals, and the amplitude-frequency characteristics of filters corresponding to noise audio can be designed to filter noise signals at specific frequencies. For the noise in special scenes such as conference venues, LMS adaptive algorithm can be used because of its high correlation, so as to obtain a relatively pure audio signal.

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