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Denoising of a Speech Signal using Wiener Filter

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***Abstract*—Noise reduction is one of the major problems faced over the past several decades. Among the various techniques that were developed, the optimal Wiener filter can be considered to be one of the important noise reduction approaches, which has been used in different forms and developed in various applications. Although the Wiener filter may cause some detrimental effects to the speech signal , few efforts have been reported to show the inherent relationship between noise reduction and speech distortion. This project tells the quantitative performance behavior of the Wiener filter in the context of noise reduction.**

***Index Terms*— Minimum mean square error (MMSE), denoising, Automatic speech recognition, deconvolution, noise reduction, stationary noise.**

I. INTRODUCTION

Denoising of a speech signal is a method used to recover an original signal from noisy signals corrupted by different types of noises. Noise may be of different types such as white noise, pink noise, babble noise and many other types of noise that is present in the environment can be considered. Over the last decades, noise removal from speech signals is an area of interest of researchers in speech processing. Various methods are involved in the removal of the different noise that is present. The application areas of denoising are 1D or 2D biomedical signal analysis, producing & analyzing irregular signals or images.

Noise reduction techniques have a broad range of applications from hearing aids to cellular phones, voice-controlled systems, multiparty teleconferencing and automatic speech recognition systems. The choice between using and not using a noise reduction technique may be a significant impact on the functioning of these systems. In this context, noise reduction is a very challenging and complex problem due to several reasons. First of all, the nature and the characteristics of the noise signal change significantly from application to application., and moreover vary in time. It is therefore very difficult, if not impossible to develop a versatile algorithm that works in diversified environments. Secondly, the objective of a noise reduction system is heavily dependent on the specific context and application. In some scenarios, for example, we want to increase the intelligibility or improve the overall speech perception quality, while in other scenarios, we expext to ameliorate the accuracy of a communication system, or simply reduce the listener's fatigue. It is very hard to satisfy all the objectives at the same time. In addition, the complex characteristics of speech and the broad spectrum of constraints make the problem even more complicated. Research on noise reduction/ speech enhancement can be traced back to forty years ago with two patents by Schroeder, where an analog implementation of the spectral magnitude subtraction method was described. Since then it has become an area of active research. Over the past several decades, researchers and engineers

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have approached this challenging problem by exploiting different facts of the properties of the speech and noisy signals.

II. Methodology/Procedure

The inverse filtering is a restoration technique for deconvolution, i.e., when the signal is faded by a known low pass filter, it is possible to recover the image by inverse filtering or generalized inverse filtering. However, inverse filtering is very sensitive to additive noise. The approach of reducing one degradation at a time follows us to develop restoration algorithm for each type of degradation and simply combine them. The Wiener filter executes an optimal tradeoff between inverse filtering and noise smoothing. It removes the additive noise and inverts the blurring simultaneously.

The Wiener filter is optimal in terms of the mean square error. In other words, it minimizes the overall mean square error in the process of inverse filtering and noise smoothing. The Wiener filtering is a linear estimation of the original image. The approach is based on a stochastic framework. It is easy to see that the Wiener filter ha stwo parts, an inverse filtering part and the noise smoothing part. It not only performs the deconvolution by inverse filtering (high pass filtering) but also removes the noise with a compression operation (low pass filtering). For example, the known signal might consist of an unknown signal of interest that has been corrupted by additive noise. The Wiener filter can be used to filter out the noise from the corrupted signal to provide an estimate of the underlying signal of interest. The Wiener filter is based on a statistical approach, and a more statistical account of the theory is given in the Minimum Mean Square Error (MMSE) article.

In general, the more microphones are available, the easier the task of noise reduction. For example, when multiple realizations of the signal can be accessed, beamforming, source separation, or spatio- temporal filtering techniques can be applied to extract the desired speech signal or to attenuate the unwanted noise. If we have two microphones, where the first microphone picks up the noisy signal, and the Microphone ps able to measure the noise field, we cause the second Microphone signal as a noise reference and eliminate the noise in the first microphone by means of Wiener filtering. However, in most situations, such as mobile communications, only one microphone is available. In this case, noise reduction techniques need to rely on assumptions about the speech and noise signals, or need to exploit aspects of speech perception, speech production, or a speech model. A common assumption is that the noise is additive and slowly varying, so that the noise characteristics estimated in the absence of speech can be used subsequently in the presence of speech. If in reality this premise does not hold, or only partially holds, the system will either have less noise reduction, or introduce more speech distortion. Even with the limitations outlined above, single-channel noise reduction has attracted a tremendous amount of research attention because of its wide range of applications and relatively low cost. A variety of approaches have been developed, including spectral or cepstral restoration, signal subspace, parametric model based method, and statistical model based method. Most of these algorithms were developed independently of each other and generally their noise reduction performance was evaluated by accessing the improvement of signal to noise ratio (SNR), subjective speech quality. Almost with no exception, these algorithms achieve noise reduction by introducing some distortion to the speech signal. Some algorithms, such as subspace method are even explicitly based on tradeoff between noise reduction and speech distortion. Since there are so many algorithms in the literature, it is extremely difficult , if not impossible to find a universal analytical tool that can be applied to any algorithm. In this work, we choose the Wiener filter as the basis since it is one of the most fundamental approaches, and many algorithms are closely connected to this technique. It is then showed that the single channel Wiener filter, the amount of noise reduction is in general proportional to the amount of speech degradation, implying that when the noise reduction is maximized, the speech is maximized as well. Depending on the nature of application, some practical noise reduction systems require the speech signal to be as clean as possible, but may allow some degree of speech distortion. Therefore, it is necessary that we have some management scheme to control the compromise between the noise reduction and speech distortion in the context of Wiener filtering.

A. Block Diagram

We can approach the problem of designing the Wiener filter estimate of the unknown system in a slightly different way.

Consider the block diagram given below:

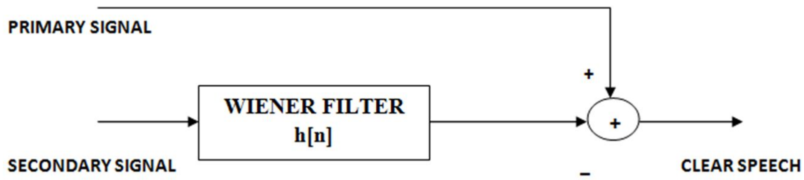

Figure 1: Block diagram of Wiener Filter

A good estimate of the unknown filter impulse response h(n) can be obtained, if the difference or error signal between the two outputs(real and estimated) is minimal (ideally zero). The use of LTI systems in order to perform minimum mean square error (MMSE) estimation of a WSS random process of interest, given measurements of another related process. The measurements are applied to the input of the LTI system, and the system is designed to produce as its output the MMSE estimate of the process of interest. We first develop the results in discrete time, and for convenience assume that the processes we deal with zero mean. We will then show exactly analogous results apply in continuos time, although their derivation is slightly different in certain parts.

III. Experimental Results

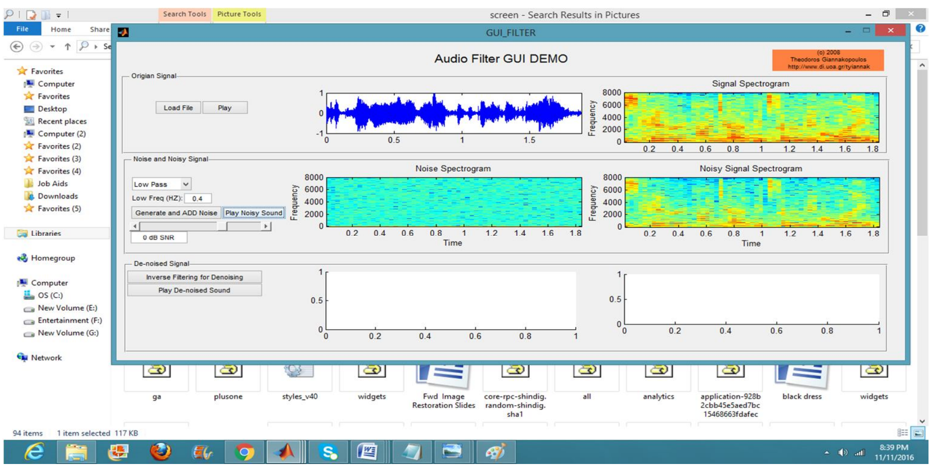

Figure 2: spectrogram of an input primary speech signal

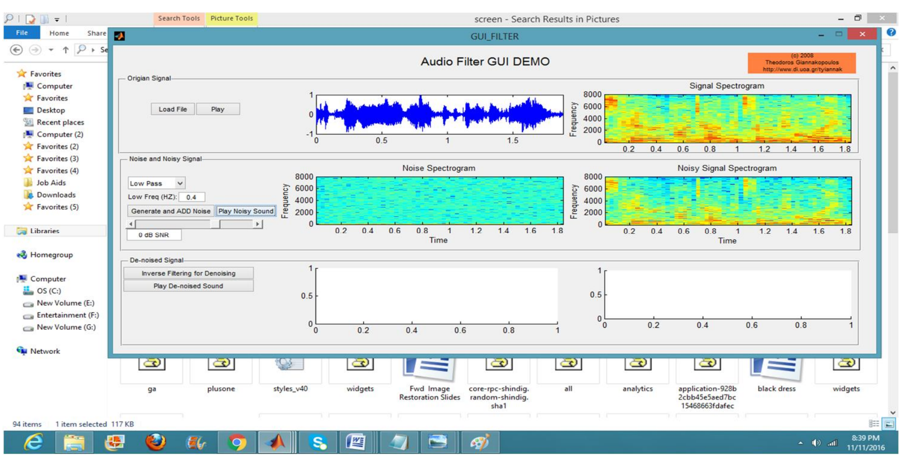

Figure 3: Spectrogram of a secondary speech signal

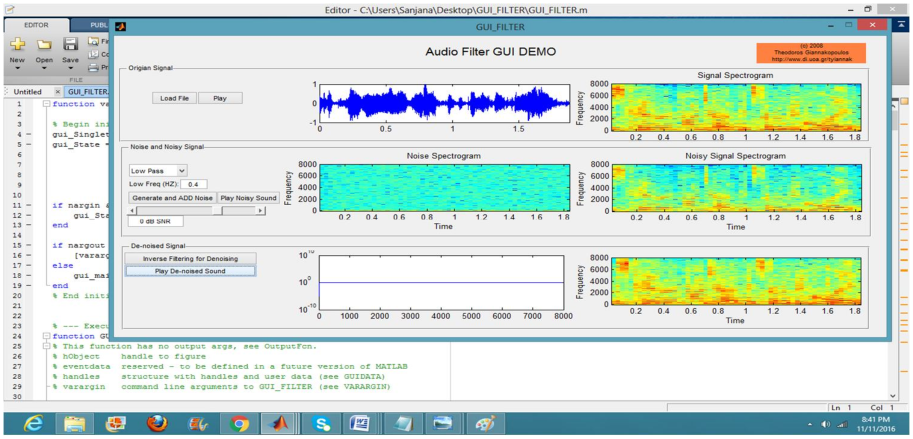

Figure 4: Spectrogram of Denoised speech signal

IV. Discussion

Wiener filter can be considered as one of the most fundamental noise reduction approaches. It is widely known that the Wiener filter achieves noise reduction by deforming the speech signal. Experimental results demonstrate that the proposed framework can achieve significant improvements in both objective and subjective measures over the conventional MMSE based technique. It is also interesting to observe that the proposed Wiener filtering approach can well suppress stationary noise, which is tough to handle in general.

A. Advantages

- Wiener filters are used to attenuate the noise present in the signal of interest and sharpen the signal.
- The highly non-stationary noise could be well suppressed in the off-line learning framework.
- Nearly no empirical thresholds to avoid the non-linear distortion in DNN- based speech enhancement.
- Nearly no guassian or independent assumptions.
- The deep architecture could well fit the non-linear relationship for regression function approximation.

B. Applications

- Mobile phones
- VoIP (Voice Over Internet Protocol)
- Teleconferencing system
- Hearing aids

V. Conclusion

In this work, we analyzed the inherent relationship between noise reduction and speech distortion with the Wiener filter, starting from the speech and noise estimation using Wiener theory, we introduced a speech distortion index and two noise reduction factors, and showed that for the single-channel Wiener filter, the amount of noise reduction is in general proportional to the amount of speech degradation, i.e., more noise reduction incurs more speech distortion. Hence with the use of Wiener filter it is possible to get rid of the background noise that is present in the signal of interest and thereby enhance the quality of the degraded speech.

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