Diminishing of Noise in Mobile Contiguous

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***Abstract*— This paper discusses about the application of noise diminishing process for mobile phones. Mobile phones are often used in noisy environments such as noisy street, cafe, or in a moving car/train. The attribute of speech can be degraded by the background noise. The spectral subtraction method is considered to be the popular method of noise reduction in speech signal. The performance of the original speech signal is improved by processing the signal based on the type of background noise, through that the performance can be enhanced. The objective of the paper is to separate the noise that is present in the original speech signal in various environments of mobile phone users. To enrich the speech signal, Wavelet based Spectral Subtraction (WSS) is introduced to reduce the background interference. The extracted signal will be used to detect the type of noise. The WSS utilizes both low and high frequency components. The performance of WSS will be evaluated and calculated by Signal to Noise Ratio (SNR).**

Keywords— Diminishing of noise , Mobile Phone, Signal to Noise Ratio(SNR), Wavelet based Spectral Subtraction (WSS).

I. INTRODUCTION

Speech is a coarse mode of communication in the midst of human being and also the most effective and reliable form of exchanging information among human. Speech has been implanted into many applications like speech recognition, development of hearing aid, mobile and other sorts of personal communication.

In environment undesirable noise causes undesired effects in speech signal transmission and reception. Noise can be any types like random or white noise with no coherence or coherent noise introduced by the device mechanism. Suppression of undesired signal present in the background, such as babble or traffic noise.

Speech enhancement techniques have been widely employed for minimizing these undesirable background noises. According to a specific application, the requirement of speech enhancement technique varies to increase speech attribute and articulation of speech.

The presence of noise in speech signals can result in appreciable degradation in both the speech attribute and articulation. The background acoustic noise can corrupt speech signals in communication systems, interfering with communication interpretation and discernment.

In general noise reduction or speech enhancement algorithms attempt to improve the performance of communication systems when their input or output signals are corrupted by noise. The main aim of speech enhancement is to improve one or more perceptual aspects of speech, such as the articulation or speech attribute, [1].

The background noise is acoustically or digitally added to the speech thus obtaining noisy speech signal.It is normally hard to reduce noise without distorting speech and thus, the functioning of speech enhancement systems is determined by the tradeoff between speech distortion and interference reduction, [2]. Amongst the speech enhancement techniques, Discrete Fourier Transform (DFT) based transforms domain techniques have been widely spread in the form of spectral subtraction.

Noise reduction methods such as spectral subtraction detect the non-speech frames from the speech signals, then estimate the noise components in the frames, and use the estimate to reduce the noise components included in the speech frames, [3]. To reduce the influence of the background noise and increase the definition of the speech, the algorithm wavelet based spectral subtraction (WSS) of speech enhancement method is introduced in this paper.

Performance of the feature vector is evaluated in the context of environment detection and classification in clean, noisy, reverberant, and noisy reverberant conditions. Three features are chosen for the environment classification task, [4]. The noise reduction scheme in mobile environments, where several types of noises affects the accuracy of the speech signal and it verifies the effectiveness of the noise reduction approach. The unsupervised noise reduction that improves the performance of voice-based information retrieval tasks in mobile atmospheres, [5].

The enhanced speech is feature extracted by the most important attributes such as Magnitude Spectrum, Spectral Flux, Spectral Variability, Zero Crossing and Power Spectrum using audio segmentation features, [6]. By using Frond-End Processing SNR is very low in Subspace based Speech enhancement over mobile devices, [7]. In Statistical evaluation of the product was carried out by calculating theSNR. Frame averaging technique was used to recover the SNR.

In further sections of this paper, Section II discusses about Conventional Spectral Subtraction, Section III discusses about Diminishing of Noise Structure, Section IV shows the Simulation Results, Section V shows the Statistical Analysis, Section VI contains the Results and Discussions and Section VII contains the Conclusion.

II. CONVENTIONAL SPECTRAL SUBTRACTION

Speech enhancement techniques have been widely employed for minimizing these undesirable background noises. According to a specific application, the requirement of speech enhancement technique varies to increase speech attribute, articulation and performance of speech communication devices.

To diminish the influence of the background noise and increase the definition of the Speech filtering techniques is used such as Spectral subtraction, Wiener filtering, Signal subspace approach.

Spectral subtraction technique is historically one of the first algorithms proposed for background noise reduction. Spectral subtraction is performed by subtracting an estimate of the noise spectrum from the noisy speech spectrum.

Spectral subtraction is a method for restoration of the power spectrum or the magnitude spectrum of a signal perceived in additive noise, through subtraction of an appraisal of the average noise spectrum from the noisy signal spectrum.

The Fourier transform, provides information about the frequency domain, however time localized information is essentially lost in the process. The drawback is the inability to associate features in the frequency domain with their locality in time, as a modification in the frequency spectrum will result in changes throughout the time domain.

In divergence to the Fourier transform, the wavelet transform allows unique localization in both the time domain via translation of mother wavelet, and in the scale (frequency) domain via dilations. The translation and dilation operations applied to the mother wavelet are performed to estimate the wavelet quantities, which characterize the relationship between the wavelet and a localized section of the signal. The wavelet coefficients are premeditated for each wavelet segment, giving a time-scale function concerning the wavelets correlation to the signal.

III. DIMINISHING OF NOISE STRUCTURE

The background noise is the most common factor degrading the speech attribute and articulation in mobile atmosphere. The diminishing of noise module intends to lower the noise level without affecting the speech signal attribute. The module is based on the wavelet packet decomposition is performed both in low and high frequency components.

WSS focuses to separate the background noise in various environments of mobile phone users and the block diagram is illustrated in the Figure 1.

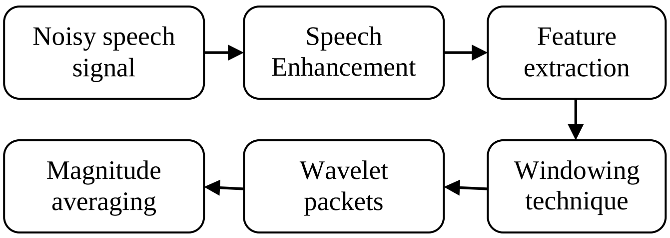

Figure 1 WSS Noise Diminishing Structure

The input signal is taken as random signal which is a combination of speech signal and noisy signal. The signal is then enhanced using Enhancement technique. The enhanced signal is then processed by Feature extraction. This includes the measure of statistical parameter such as Magnitude Spectrum, Power Spectrum, Spectral Centroid and Root Mean Square (RMS).The statistical parameters are used to detect the type of noise in mobile contiguous.

In order to eliminate that portion of noise the methodology of WSS is carried out with that noisy speech signal. The WSS module consists of windowing technique, wavelet packets and magnitude averaging.

Windowing technique is carried out with the extracted signal. The processed signal is given as an input to wavelet packets. In wavelet, the frequency and time can be analyzed in fast and reliable process. The magnitude is taken for the discretized signal to obtain the noise diminishing speech signal.

A. Windowing technique

The undesired transition that occurs before the signal settles to its desired value. Glitches can be reduced by shaping the signal so that its ends match more smoothly data cannot be lost. Multiplying by a window function (windowing) suppresses glitches and avoids the broadening of the frequency spectrum caused by the glitches. By using windowing spectral resolution of frequency domain will increase.

B. Wavelet packets

Discrete time signal passed through more filter than Discrete Wavelet Transform (DWT). The versatility and power of the DWT can be significantly increased by using its generalized form, wavelet packets. Unlike the DWT which only decomposes the low frequency components, a Wavelet packet utilizes both the low and high frequency components. Wavelet packet is known as sub band tree structure

C. Magnitude averaging

Averaging of spectral magnitude can be used to reduce the error. Averaging is taken for number of frames in the spectrum to retain the speech signal.

Noise diminishing speech signal is achieved by using WSS and the articulation of speech signal is improved by subtracting the noise spectrum from the noisy speech spectrum.

IV. SIMULATION RESULTS

Speech Signal with Noise as sample 1

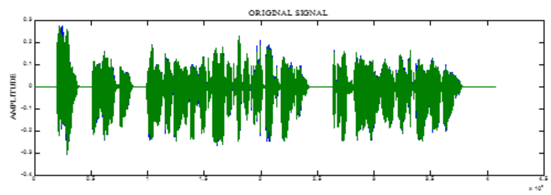

Figure 2 Input signal of speech

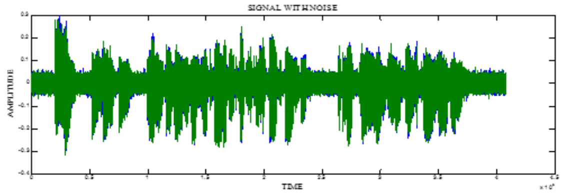

Figure 3 Convoluted signal of noise with speech

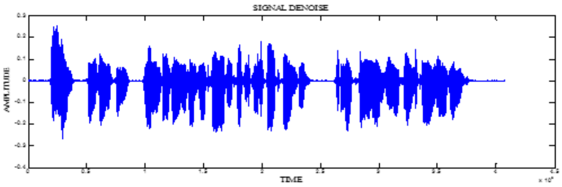

Figure 4 Subtracted signal from Noisy signal

Speech Signal with Birds Noise as sample 2

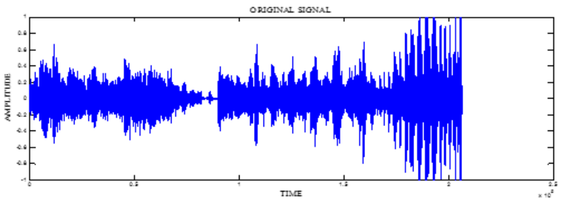

Figure 5 Input signal of speech

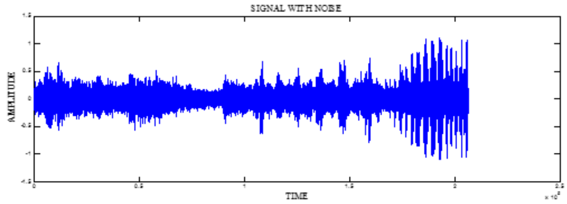

Figure 6 Convoluted signal of noise with speech

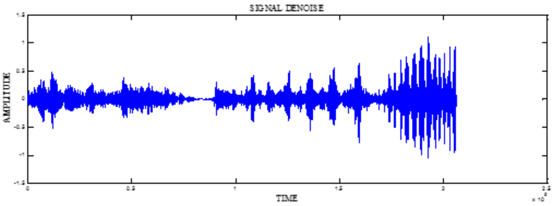

Figure 7 Subtracted signal from Noisy signal

V. STATISTICAL RESULTS

The SNR values of Coiflet and Haar wavelet is calculated for various Noise levels are shown in Table 1.

S No.	Noise Level	SNR	
		Coiflet	Haar
1	10	19.2	17.2
2	20	22.3	21.4
3	30	23.2	22.3
4	40	23.6	23.9
5	50	23.8	24.1

Table 1 SNR Values of Sample 1

The Characteristic response of Coiflet and Haar wavelet is plotted in the form of graph as shown in Figure 8.

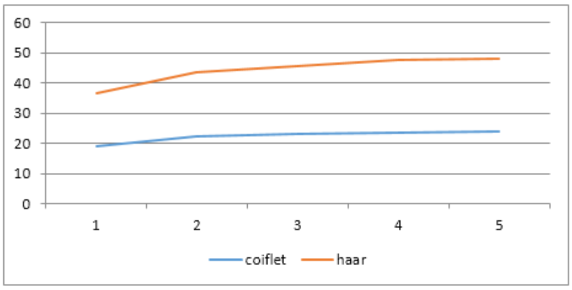

Figure 8 Coiflet and Haar Characteristics of Noise for Sample 1

The SNR values of Coiflet and Haar wavelet is calculated for various Noise levels are shown in Table 2.

S No.	Noise Level	SNR	
		Coiflet	Haar
1	10	13.4	10.0
2	20	16.9	12.5
3	30	17.1	13.1
4	40	17.6	13.2
5	50	17.4	13.4

Table 2 SNR Values of Sample 2

The Characteristic response of Coiflet and Haar wavelet is plotted in the form of graph as shown in Figure 9.

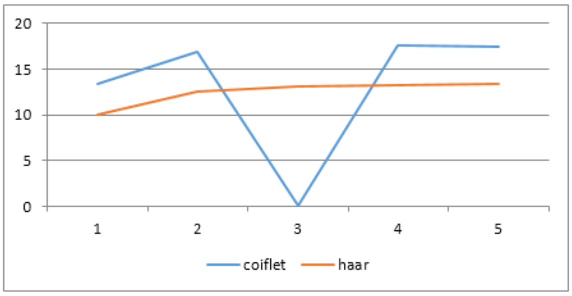

Figure 9 Coiflet and Haar Characteristics of Noise for Sample 2

VI. RESULTS AND DISCUSSIONS

The Simulations results shown in Figure 2-7 are carried out in MATLAB. Figure 8 and 9 are the comparative sheets which describes the comparison between the two different noise samples. Samples differ from the occurrence of bird noise at the back ground.

Figure 2 shows the characteristics between time and amplitude of the original speech signal which is considered as input signal. The number of iteration is taken as 100.

Figure 3 shows the characteristics of convolution of two signals, one is the input speech signal and the other is the noise signal (i.e.), additive white Gaussian noise.

Figure 4 is the resulting noise diminishing signal. The original speech signal is obtained from removing noise from the convoluted signal.

Figure 5 shows the characteristics between time and amplitude of the noisy speech signal which is the combination of speech signal with bird's noise as input signal. The number of iteration is taken as 100.

Figure 6 shows the characteristics of convolution of two signals, one is the input noisy speech signal and the other is the noise signal (i.e.), additive white Gaussian noise.

Figure 7 is the resulting noise diminishing signal. This signal is obtained from removing noise from the convoluted signal.

Figure 8 is the characteristic response of coiflet and haar wavelet of sample 1.

Figure 9 is the characteristic response of coiflet and haar wavelet of sample 2.

VII. CONCLUSION

In this paper, Wavelet packets are applied to the noisy speech signal. This method removes the noise from noisy speeches. It is observed that the noise diminishing signal provides better performance over all kinds of background noise conditions. By using Wavelet packets, considerable amount of noise is reduced further by using WSS the speech attribute will be improved. In future work the SNR will be improved by the incorporation of wavelet packets with windowing technique. Based on the measure of Statistical Parameters the analysis is carried out in all kinds of background noises in mobile environment.

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