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# Source Separation and Echo Cancellation Using Independent Component Analysis and DWT

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**ABSTRACT**: In recent telecommunication hands free communication is used widely. This hands free communication suffers from many technical problems like room reverberation, acoustic echo, source interference, background noise. This paper proposes with the objective of blind source separation from mixture of many audio source signals , along with echo cancelation. Recent statistical and computational method ICA examined for source separation & echo cancellation. ICA along with DWT worked efficiently in source separation and denoising. An objective component of data is extracted from a linear representation of data, where that data must be non Gaussian and independent or partially dependant. DWTs are applicable for non-parametric signals.

KEYWORDS: Blind source separation, independent component analysis, discrete wavelet transform.

# **I**.INTRODUCTION

Recently, the demand for speech signals has increased .The use of hands-free communication has been increased tremendous. There are problems in recognizing speech in conference calls if there are more than two speakers and contamination of noise is more. Conventional or existing speech recognition techniques degrade by their performance as sound sources in increase or any background sound exists. It is difficult task for improving performance of speech recognition from mixture of speeches along with noise removal. Consider an indoor system like room where two people are speaking simultaneously, also there may be some unwanted background sound of telephone, fan, or music. You have two microphones which you hold in different locations each giving two recorded time signals which is denoted as  $S_1(k)$  and  $S_2(k)$ . Speaker output of these two signals will be  $x_1(k)$  and  $x_2(k)$ . These recorded signals may be mixture of original source signals, noise and some acoustic echo which is major problem. Feasible solution to recover original source signals from mixture signal without having knowledge of original source signal is blind source separation technique. From observed mixed speeches, this becoming prominent technique, extracts objective sources signals. Method is said blind source separation because it identifies original signal without the knowledge of source positioning, spectral construction, or a mixing system. For achieving this, source separation concentration have been made on technique based on Independent component analysis(ICA), which extracts independent sounds from among mixed sounds. The ICA is a statistical method. ICA Predicts the original signal from the mixture signal, even if the original signal and the transfer function are unknown. It predicts by assuming statistical independence of the object sound and other sound which is non-Gaussian. Wavelets eliminate or reduce the noise in speech signal in speech. Wavelet de-noising is a non parametric method.

# **II. LITERATURE SURCEY**

Blind source separation was detailed explained, along with basic BSS model in [5].Principal component analysis and its limitations, overcoming this PCA problems ICA giving robust solution were described.[5] Ica methods like neural network, Maximum entropy method, minimization of mutual information and fast ICA were comparatively explained in [5]. Recent Independent component technique was explained along with discreet wavelet transform, author also implemented this two techniques showing results that ICA together with DWT gives efficient results [1]. In this paper author explained two source un-mixing techniques Like BSS and adaptive beam formers (ABF), also explained



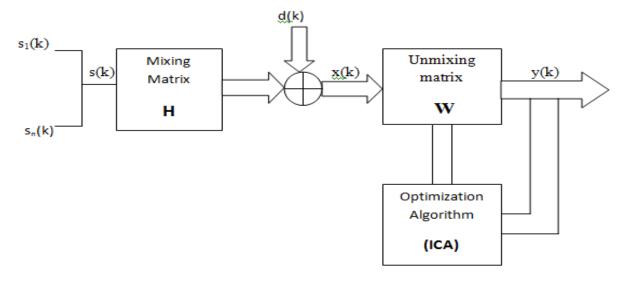
(An ISO 3297: 2007 Certified Organization)

# Vol. 4, Issue 5, May 2015

intelligence of BSS over ABF [4]. Approach of higher order statistics separation procedure of BSS were explained.[4]. This paper implemented acoustic echo cancellation and noise suppressing nonlinearities. [2] .De-noising techniques ,different thresholding and decomposition of signal methods using wavelets were detailed. [7]

#### **III. BLIND SOURCE SEPARATION**

Basic objective of BSS is separating or extracting original sound sources which are mutually independent from mixture of signals. Sometimes mixing structure is also unknown. Original source information is unknown so it is called as Blind source. Basic block diagram explains n number of sources is mixed through mixing matrix. Noise is added in mixed signals. Mixture of signals and noise is passed to un-mixing matrix to separate out signals. An Optimization algorithm is applied to separate these sources. Algorithms may be ICA, PCA, singular value decomposition, stationary subspace analysis. Thus BSS is estimation of  $S_in$  number of sources from *m* number of measurements.



#### Basic Block Diagram of BSS

Source component vector be  $s(k) = [s_1(k), s_2(k), \dots, s_n(k)]$ , Where  $k=1,2,\dots,N$ . and these source components are statistically independent. Vector of observed signals  $x(k) = [x_1(k), x_2(k), \dots, x_m(k)]$  at discreet time instant k. Linear BSS model would be expressed as x(k) = Hs(k) + d(k), where H is mixing matrix dimension  $(m \times n) d(k) = [d_1(k), d_2(k), \dots, d_m(k)]$  is vector of noise, independent on sources. Computation of un-mixing matrix W, is important whose output is y(k) = W.x(k), where y(k) estimate of source vector. In order to estimate output more near to source unmixing matrix W must be ideally inverse of H,  $W = H^{-1}$ .

## IV. ICA

BSS is an model in which optimization algorithm used is Independent component analysis (ICA). Term statistically independent sources is mandatory, it means ,if we consider two random source variables  $S_1$  and  $S_2$  then they are said to be independent if information on the value of any one variable does not give any information on value of other , and vice versa. These variables are considered as scalar. Term independency is not for mixture variables  $x_1$ ,  $x_2$ . Non gaussianity of components is maximized by ICA. It is necessary for source to be stationary. Sources must not allow having Gaussian distribution, if then only minimum one source is allowed. This is because whenever Gaussian signals linearly combine then resultant signal is also Gaussian. [3],If resultant signal is Gaussian then it is difficult or impossible to separate . In other words mixing matrix H is difficult to identify for Gaussian independent components.ICA categorized into methods like mutual information minimization, Maximum of non-gaussianity, maximum of likely-hood. ICA method is dependent on measure of non-gaussianity [3], and this measure is carried by method of Kurtosis, negentropy and neural network. Kurtosis is measure of peak of probability distribution function of



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# Vol. 4, Issue 5, May 2015

real valued random variable Source separation can be obtained by optimizing of suitable objective function which can be scalar measure of some distributional property of output y. Optimization of objective function depends on batchwise calculations. There are two ways by using estimated higher order statistics and second is adaptive separation [5] Adaptive separation is complicated. So its simple to use high order statistics such as kurtosis . Kurtosis is fourth order cummulant with zero time lags. The kurtosis of  $K^{th}$  source signal s(k) given by [5]

kurt[s(k)<sup>4</sup>] = E{ s(k)<sup>4</sup>} - 3[E{ s(k)<sup>2</sup>}]<sup>2</sup>

If source signal is Gaussian then kurtosis is zero, if source signal is sub-Gaussian then kurtosis is negative, if source signal is super Gaussian then kurtosis is positive.[5]. Another measure of non-gaussianity is given by negentropy. [3]

#### V. DISCREET WAVELET TRANSFORM

Fourier gives information of spectrum, for which signal considered is stationary, but speech signal is never stationary. Wavelet transform is a transform which provides time-frequency representations, of non-stationary signals, like audio. It is alternative of short time Fourier transform, Winger distribution. Important characteristics of Wavelets are scale and position. By using these characteristics it is helpful to resolve variations in signals as well as images with respect to scale and position. To process simultaneously time and frequency data signal, wavelet size can vary, which is beneficial than conventional transforms. . Low scale means detailed information at this scale, compressed wavelets are used. Low scale means high frequency which corresponds to fast changing details. High scale is overview not detailed, here stretched wavelets are used. These stretched wavelets correspond to slow changing features, so low frequency is concerned. Wavelet is categorized into continuous wavelet and discreet wavelet transform. Continuous wavelet transform gives redundant information, and in Discreet wavelet transform due to orthonormal properties there is no information redundancy.[7]. DWT consist filtering the input signal by two filters high pass filter and low pass filter Low pass signal is again further can be decomposed. Identity of signal is provided by low frequency, so low frequency content plays an important role. Consider the speech signal. This signal is decomposed into high frequency component and low frequency component Wavelet transformation in a pre-processing step is to improve the non-Gaussianity distribution of independent components that is important requirement for implementing ICA and to increase their independency. In this paper we have proposed Daubechies 32 wavelets .Decompose observed signal and then separate .In wavelet de-noising Wavelet transform operator is applied on signal contaminated of noise. Then de-noising operator is applied which consist of soft thresholding. Again inverse wavelet operator is applied to estimate original signal.

#### VI. ECHO CANCELLATION

Echo is a delayed and degraded version of original sound signal reflections. Echo may be either acoustic or electrical depending upon its nature. Multiple reflections undergoing due to objects or surface. This phenomenon is called reverberation. Causes of Reverberation are room dimensions, number of people, objects in room. Echo cancellation is used to reduce bandwidth consumption because of its silence suppression technique and remove undesired effect. When a some amount of speech signal from the speaker is captured by the microphone and is transmitted back, then an acoustic echo occurs. Main cause of acoustic echo is reverberation. Resultant degradation of quality of signal is caused because of acoustic echo. Acoustic Isolation Echo is generated due to the poor isolation between ear piece and microphone. Acoustic echo is found to be most common in today's wireless networks, because of the increasing use of headsets and Bluetooth headsets.

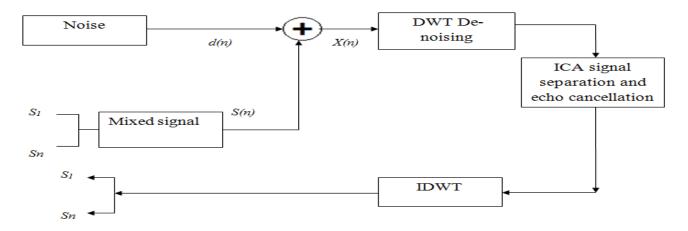
## VII. PROPOSED ALGORITHM

Here is the representation of block diagram where two or more speech signals or combination of speech and music signals, is mixed. Some additive noise is added in mixed signal. This contaminated signal is passed through discrete wavelet decomposition. Decomposed signal is subjected to independent component analysis which gives better results for source separation; also here ICA performs for echo cancellation. To regain original sources signal is passed through Inverse discrete transform for reconstruction.



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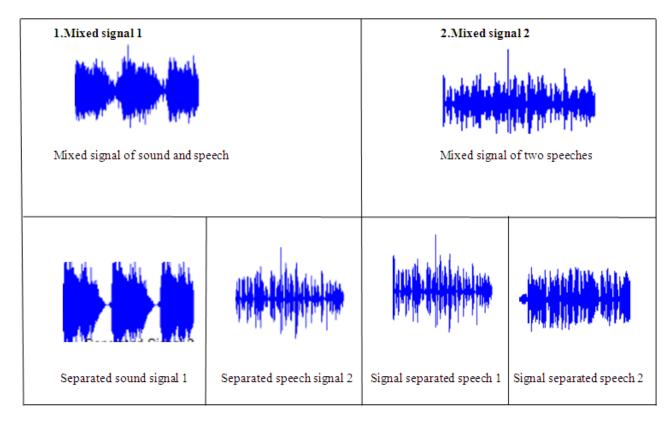
# Vol. 4, Issue 5, May 2015



Basic block diagram for source separation and echo cancellation.

## VIII. RESULTS

Implementation of combine ICA was performed on mixture of signals. Signals were individually observed and then mixed. Three types of signals were considered Music signal, Speech and Siren signal. Following diagram shows that mixture 1 consist of siren and speech signal. and Mixed signal 2 consist of two speeches, Appreciable separation using ICA and DWT de-noising.



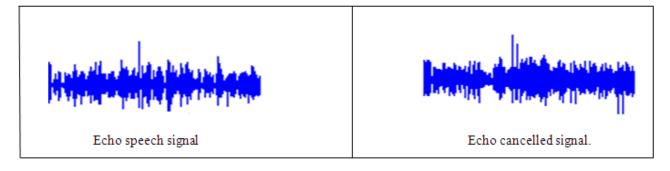
1. Source separation result



(An ISO 3297: 2007 Certified Organization)

# Vol. 4, Issue 5, May 2015

Delayed version of signal which resulted to echo was observed for speech signal and successfully Echo was removed using ICA algorithm. Following figure shows signal affected and contaminated with echo and Echo cancelled signal.



## 1. Echo cancellation result

## **IX. CONCLUSION**

This allowed us to optimize using Independent component analysis including Discreet wavelet transform for better denoising and better source separation. Combination of these both methods leads to efficient source separation technique for non-stationary signals and Echo cancellation also.

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