International Journal of scientific research and management (IJSRM)
\||Volume||3||Issue||6||Pages|| 3062-3064||2015||
Website: www.ijsrm.in ISSN (e): 2321-3418

Blind Source separation and Echo cancellation using Discreet Wavelet transform and ICA

Shweta Yadav[1], Prof. Meena Chavan[2]

[1]Dept. of Electronics, BVDUCOEP, Pune, India
shwetaa.pune@gmail.com

[2] Dept. of Electronics, BVDUCOEP, Pune, India
mschavan@bvucoep.edu.in

Abstract: Advances in telecommunications have lead to modern technologies like conference calls, video calls, audio calls. This widely communicate over *hands- free devices, where this type of communication is affected by reverberations, mixing of sources, acoustic echo etc. Goal of this paper is to estimate noise free original sources from mixture of multiple sources, and hence phenomenon known as Blind Source Separation (BSS).This implementation of source separation and estimation is performed by Independent component analysis (ICA). Aim of ICA is representing data linearly such that components must be statistically independent. De-noising was performed using discreet wavelet transform (DWT) by decomposing and thresholding. Effective results were obtained when ICA-DWT combinely worked than only ICA. Peak signal to noise ratio for ICA-DWT was more than only ICA.*

Keywords: Blind source separation, independent component analysis, Discrete wavelet transform.

1. Introduction

Many advances in communication have lead to hands- free communication. Hands-free communication included video conference calls and audio conference calls. Corporate meetings are many conducted on conference calling. Often it happens that two or more persons speak at same time and microphone receives both signals in a form of mixtures. Mixtures of these signals further also get contaminated with background noise. Background noise may contain noise of fan, telephone ring, door bell, computers, traffic etc. Background noise consist higher level of low frequency. Individual speech recognition is very difficult in this case. Classical methods for noise removing do not effectively work in suppressing noise. It becomes very difficult for the listener at another end to predict original speeches of speaker. This paper proposes attempts in separating two original sources from mixture of signals. Here we are not having any knowledge of original signal and there mixing structure, so only it is abbreviated as “Blind source separation BSS”. Consider a room in which there are two or more speakers whose source vector denoted as S(K). Vector of Mixture of signal observed at microphone X(K) , and noise vector be d(K) . Estimated output vector be Y(K). There are several methods for BSS, Principal component analysis (PCA), Singular vector decomposition (SVD), and Independent component analysis (ICA). Proposed optimization algorithm used for blind source separation is ICA

2. Blind source separation

High order dependencies can be vanished by ICA but PCA can only vanish Second order dependencies. It becomes difficult for PCA to estimate original sources if they are non-Gaussian [5]. Block Diagram shows Two or many sources are mixed in mixing matrix H, further noise is added. For estimating original sources correctly, un-mixing matrix W must be ideally equal to mixing matrix. Consider n number of sources $S(K)= [s_1(k), s_2(k), \ldots s_n(k)]$, where K = k=1,2,…N be discreet time instant, are mixed together. $d(k)= [d_1(k), d_2(k) \ldots d_m(k)]$ be noise added to mixed vectors. And m number of Observed signals be $X(k)= [x_1(k), x_2(k), \ldots x_m(k)]$. Observed signals $x_1(k) = a_{11}\, s_1(k) + a_{12}\, s_2(k) + \ldots a_{1n}\, s_n(k)$

$$x_2(k) = a_{21}\, s_1(k) + a_{22}\, s_2(k) + \ldots a_{2n}\, s_n(k)$$

$$\vdots$$

$$x_m(k) = a_{m1}\, s_1(k) + a_{m2}\, s_2(k) + \ldots a_{mn}\, s_n$$

$$H=\begin{bmatrix} a11 & \cdots & a1n \\ \vdots & \ddots & \vdots \\ am1 & \cdots & amn \end{bmatrix}$$, H is mixing matrix.

From all these components BSS model can be stated as $x(k) = Hs(k) + d(k)$,. Output of system is given by $y(k) = Wx(k)$, Here goal is to compute un-mixing matrix W . So $W=H^{-1}$. Thus source estimated can be $S=H^{-1}X$.

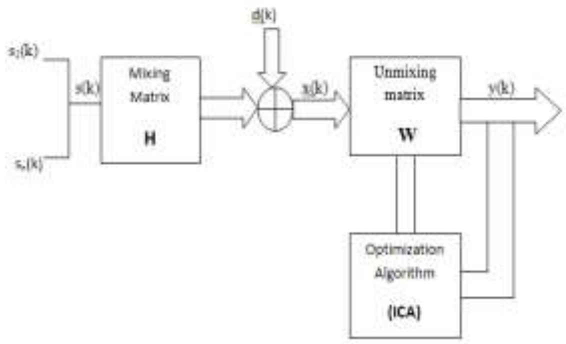


Figure.1.Block Diagram of BSS

3. Discreet wavelet transform

De-noising is performed using DWT. Audio signals are non-stationary, which means there frequency response varies with time .Fourier transform here fails for non-stationary signal.DWT efficiently provide information in both time and frequency domain. Low frequency that is high scales provide overall knowledge lasting for long time over a signal, while high frequencies that is low scale provides detail knowledge. In Continues Wavelet transform there is more redundant information, while in discreet wavelet transform there is less redundant information [7].Following formula explains DWT

$$\mathrm{DWT}_{\psi}f(a, b) = \int_{-\infty}^{\infty} f(t)\psi(t)dt$$

Where $\psi_{m,n}(t) = 2^{-m}\psi(2^{m}t-n)$ gives translated form of mother wavelet a and b provide information of frequency and time respectively. Decomposition procedure consists of passing observed signal to high pass and low pass filter. Role of high pass filter is to filter components of high frequency and, low pass filter is to filter component of low frequency. Filtered High frequency component provide details, so they are called detailed information coefficients. Filtered low frequency components provide approximation information about signal, so they are called approximation coefficients. Decomposition was performed using Coeffliet wavelets for 8 levels. Following figure explains 1 level decomposition. For de-noising of signal, these coefficients were passed further for thresholding.

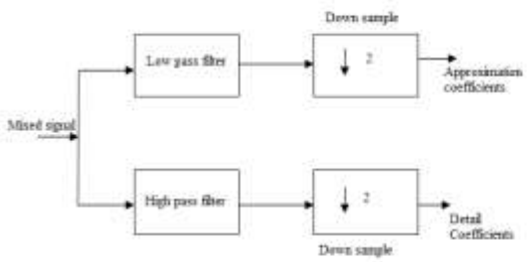


Figure .2. One Level decomposition diagram

Thresholding are of two types hard and soft. We implemented soft thresholding. Hard thresholding removes major noise but along with noise it also removes original signal also.

4. ICA

ICA is statistical linear transform method, where original random variables must be independent and scalar also [8]. Mixing matrix must have square dimensionality. Number of source variables must be equal to number of observed variables, it becomes difficult to estimate more sources from less number of observations. For solving contrast function is needed to be minimised or maximised. Random variables participating in mixture must be non-Gaussian. If these variables are Gaussian then after mixing them with orthonormal matrix they are independent but it is not possible to recover original sources. Only one Gaussian variable is allowed. According to centre limit theorem more sources when mixed then resultant mixture becomes more Gaussian [8]. Gaussian variables have large entropies. Our goal is to make non-Gaussian data , so ultimately entropies must be minimised. In this paper we have implemented pre-processing of ICA which consists centring and whitening. In process of centring sample means of X are subtracted from observed signal, this makes ultimately sources to zero mean. Whitening transforms mixing matrix into new matrix in which components are separated but dependant. Estimating proper rotation matrix is needed when applied to new matrix, will produces original independent sources. Optimization of objective function depends on batch-wise calculations. There are two ways by using estimated higher order statistics and second is adaptive separation [5]

5. Echo cancellation

Observed signals which are mixture of two or more sources undergo delayed version also. This is called echo. It may degrade signal. Depending on nature echo is subdivided into Electrical and acoustic echo. Wireless communication systems are affected by acoustic echo, because of acoustic coupling is produced between speaker microphones. So considerable attention must be given to acoustic echo's. Main source of acoustic echo is reverberations. Reverberations means signal emitting from speaker undergo multi-path propagation and reach microphone at different time intervals. Objects in room, wall, ceiling, roof all these causes to reflect sound signal and thus cause reflection. Microphone receives original signal and attenuated signal both. Without affecting original information of signal and to eliminate echo is challenging task. Independent component analysis is profound method to eliminate Effects of echo compared to any other conventional methods.

6. Proposed algorithm

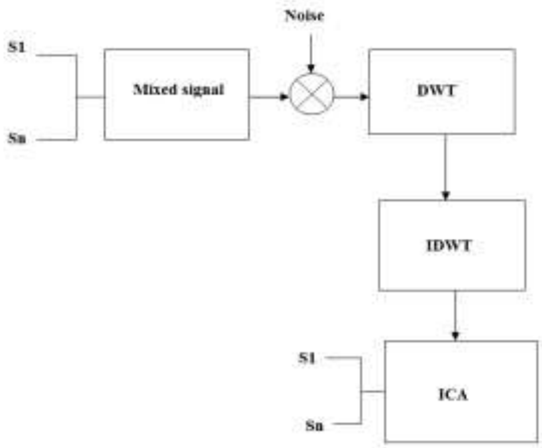
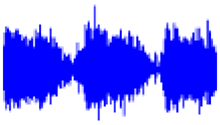


Figure .3.Basic block diagram for source separation and echo cancellation.

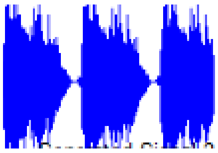
Audio signals of different types music, speeches and siren were mixed together in mixing matrix. This mixed signal may be contaminated with noise. Priory these mixed signals were allowed to undergo discreet wavelet de-noising. Noise less signal was then passed for ICA. Pre processing was first done on signal which consists of centering and whitening of signal. Whitening made easier to estimate original sources. Reconstruction of Decompose signal was necessary, which was performed by IDWT.

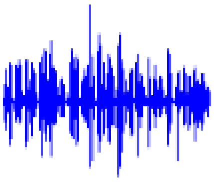
7. Results

Following figure shows mixed signal of siren and speeches and effectively seperation was performed using ICA.



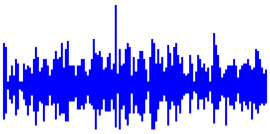
Mixed signal of sound and speech



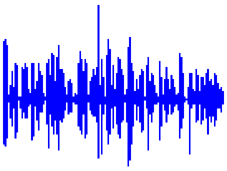


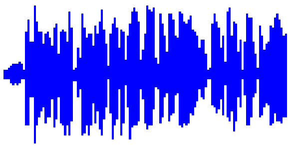
Separated sound signal 1   Separated speech signal 2

Following signal shows mixed signal of two speeches and there separated original speeches by using ICA.



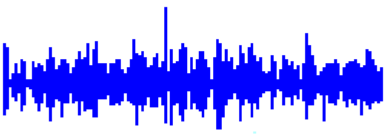
Mixed signal of two speeches



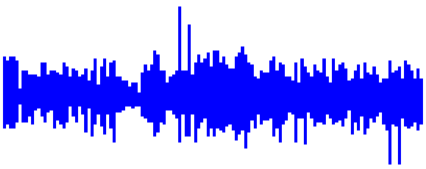


Signal separated speech 1   Signal separated speech.

Also following figure shows Echo containing speech signal. Using ICA echo was removed.



Echo speech signal



Echo cancelled signal.

Performance of source separation using solely ICA and ICA with DWT were measured using Peak signal to noise ratio. PSNR for ICA was up to 78 and PSNR for ICA-DWT was up to 84.

8. Conclusion

In Proposed paper separation of signal was successfully carried. Performance was analyzed by using Peak signal to Noise Ratio (PSNR).PSNR of system having only ICA was less i.e. 78 than PSNR of system having ICA with DWT technique i.e. 84. So it has been concluded that ICA-DWT together performed better than only ICA.

References:

[1] A.Wims Magdalene Mary1, Anto Prem Kumar2, Anish Abraham Chacko31 Karunya University, Coimbatore, India," *Blind source separation using wavelets*", IEEE International Conference on Computational Intelligence and Computing Research,2010.

[2] *Ted S. Wada, Biing-Hwang (Fred) Juang,* **"**Acoustic echo cancellation based on independent component analysis and integrated residual echo enhancement**" ,** IEEE Workshop on Applications of Signal Processing to Audio and Acoustics, 2009

[3] Aapo Hyvärinen and Erkki Oja, *, "* Independent Component Analysis: Algorithms and Applications", Neural Networks Research Centre Helsinki University of Technology P.O. Box 5400, FIN-02015 HUT, Finland *Neural Networks, 13(4-5):411-430, 2000*

[4] *Shoji Makino ,Shoko Araki , Ryo Mukai ,Hiroshi Sawada ,*" Audio source separation based on independent component analysis", NTT Communication Science Laboratories, NTT Corporation 2-4 Hikaridai, Seika-cho, Soraku-gun, Kyoto 619-0237, Japan,

[5] Niva Das, Aurobinda Routray, Pradipta Kishore Dash," ICA Methods for Blind Source Separation of Instantaneous Mixtures**:** A Case Study", Neural Information Processing – Letters and Reviews, Vol. 11, No. 11, November 2007.

[6] Dr. Alka Mahajan, Gajanan Birajdar," Blind Source Separation Using Modified Contrast
Function in Fast ICA Algorithm", *International Journal of Computer Applications (0975 – 8887)*
Volume 6– No.4, September 2010.

[7] Slavy G. Mihov, Ratcho M. Ivanov, Angel N. Popov," Denoising Speech Signals by Wavelet Transform", ANNUAL JOURNAL OF ELECTRONICS, ISSN 1313-1842,2009

[8] Vivek Anand. Dr. R. S. Anand.Dr. M. L. Dewal," Implementation of blind source separation of speech signals using independent component analysis", International Journal of Computer Science and Information Technologies, Vol. 2 (5) , 2011, 2147-21