ISSN 2394-3777 (Print)
ISSN 2394-3785 (Online)
Available online at www.ijartet.com
INTERNATIONAL JOURNAL OF ADVANCED RESEARCH TRENDS IN ENGINEERING AND TECHNOLOGY (IJARTET)
VOL. II, SPECIAL ISSUE VIII, FEBRUARY 2015 IN ASSOCIATION WITH
SRI KRISHNA COLLEGE OF ENGINEERING AND TECHNOLOGY, COIMBATORE
FOURTH NATIONAL CONFERENCE ON COMMUNICATION TECHNOLOGY INTERVENTIONS FOR RURAL AND SOCIAL DEVELOPMENT (NCC –TIRD 2015)-
DEPARTMENT OF ECE
7TH FEB 2015

FPGA Implementation of Denoising Speech Signal using Stationary Wavelet Transform

R.Sindhujaa[1], K.Kavin Kumar[2]

PG Scholar, ECE, Kongu Engineering College, Perundurai, India[1]

Assistant Professor, ECE, Kongu Engineering College, Perundurai [2]

Abstract: Speech signals are often contaminated with acoustic noise, which is present in a variety of listening environments. This problem is of critical importance because background noise is particularly damaging to speech intelligibility for people with hearing loss and hearing aids users. The wavelet transform plays an important role in signal analysis and widely used in many applications such as signal detection and Denoising. The basic idea behind the project is to estimate the uncorrupted speech from the distorted or noisy speech signal and is also referred to as speech "Denoising". There are various methods to help restore speech from noisy distortions. In this project by using Wavelet Packet Transform (WPT) and Stationary Wavelet Transform (SWT) the background noise in speech signal can be removed. Wavelet Packet Transform (WPT) is a generalization of wavelet decomposition to offer a richer frequency range for signal analysis. Here details and approximations (i.e.,) high frequency and low frequency components are split to give a wavelet packet decomposition tree. By further decomposing the signal into packets the noise can be removed at each stages of decomposition. By using Stationary Wavelet Transform (SWT) the signal can be denoised and on further sampling the speech signal can be enhanced and finally a noise free signal is obtained. The simulation has been done in MATLAB. Finally the design will be implemented in XILINX Virtex-5 FPGA Kit.

Keywords: Denoising, FPGA, Stationary Wavelet Transform (SWT), Wavelet Packet Transform (WPT), Xilinx.

I. INTRODUCTION

Digital hearing aids score over their analog counterparts because of using advanced digital signal processing (DSP) algorithms to compensate speech signal and improve intelligibility of the hearing impaired in noisy environment. The general problem of noise reduction has been addressed in great depth by researchers who have investigated several speech enhancement algorithms and techniques for various noisy environments to reduce the effects of background noise on speech intelligibility and overall sound quality[1]. Speech enhancement methods include: spectral subtraction, mean square estimation formant based methods, Discrete Fourier transform (DFT) and Discrete Wavelet Transform (DWT) [1].

Wavelet based approaches have become a popular alternative to the Fourier transform in the field of digital signal processing due to their multi-resolution capability[3]. The Wavelet based approach is applied for reducing noise by expanding the speech in a series of implicitly filtered, shift invariant wavelet packet basis vectors. The implicit filtering operation allows reducing correlated noise while retaining low-level high-frequency spectral components that are necessary for intelligible speech compensation for sensory neural impairments[1]. The toughest challenge for designers is the implementation of the devices which meet the requirements of hearing aid users, like portability, low power consumptions, noise reduction and intelligibility.

Conventional design use DSP devices as design platform which are based on general-purpose, fixed-architecture technology. On the positive, the improvement of FPGA devices offer flexibility in modification of the design architecture, high performance as well as low cost of prototyping.

In this work the features of the Wavelet Packet Transform (WPT) and the Stationary Wavelet Transform (SWT) are exploited for the design of signal Denoising [1]. The SWT for denoising shows the capability of SWT to offer better denoising performance than the ordinary orthogonal wavelet. The aim of the proposed Wavelet Packet

ISSN 2394-3777 (Print)
ISSN 2394-3785 (Online)
Available online at www.ijartet.com
INTERNATIONAL JOURNAL OF ADVANCED RESEARCH TRENDS IN ENGINEERING AND TECHNOLOGY (IJARTET)
VOL. II, SPECIAL ISSUE VIII, FEBRUARY 2015 IN ASSOCIATION WITH
SRI KRISHNA COLLEGE OF ENGINEERING AND TECHNOLOGY, COIMBATORE
FOURTH NATIONAL CONFERENCE ON COMMUNICATION TECHNOLOGY INTERVENTIONS FOR RURAL AND SOCIAL DEVELOPMENT (NCC –TIRD 2015)- DEPARTMENT OF ECE
7TH FEB 2015

Transform and Stationary Wavelet Packet Transform module is to perform speech enhancement and noise reduction.The decomposition and best tree selection of signals corrupted with additive Gaussian white noise [4] are performed in the Daubechies wavelet basis, then adaptive thresholding method based on local thresholding [5] is applied on the wavelet detail coefficients for denoising. Simulations are run on Matlab.

II. Noise

Noise means any unwanted sound. Noise is not necessarily random[6]. Sounds, particularly loud ones, that disturb people or make it difficult to hear wanted sounds, are noise. For example, conversations of other people may be called noise by people not involved in any of them; any unwanted sound such as domesticated dogs barking, neighbours playing loud music, portable mechanical saws, road traffic sounds, or a distant aircraft in quiet countryside, is called noise. Acoustic noise can be anything from quiet but annoying to loud and harmful.

A. *Different Noises affecting Speech Signal*

Noise is any unwanted signal that interferes with a speech signal in the band of interest[6]. Specialists have distinguished three types of noise that are particularly damaging to speech intelligibility:

1.Random noise with an intensity-frequency spectrum similar to that of speech.

2.Interfering voice: The interference produced by many other voices of roughly equal intensity (known as speech babble) has physical characteristics similar to that of random noise with a speech-shaped intensity-frequency spectrum.

3.Substantial room reverberation: Reverberation is produced by sound being reflected off walls, floors, ceilings and other hard surfaces.

B. *Sources of Noise*

Noise induced by the environment or the transmission channels can be linear in the power spectrum domain or linear in the log spectral domain or non-linear in both domains[6]. Environment noises are usually additive. Depending on the environment, such assumptions may or may not hold. Another assumption, often made is that the noise is stationary and uncorrelated with the speech signal. The long term stationarity of the noise[3] excludes distortions occurring frequently in office environments such as door slams and speaker induced noise. Noise sources can be classified in various types according to the task and the context in which the conversation occurs. Acoustics is a sound spoken in a room is prolonged, with a more or less logarithmic decay, so that it is present to mask subsequent sounds[2]. In telephone speech there is often reverberation when the microphone is placed too far from the talker. The type of noise induced is a convolution noise.

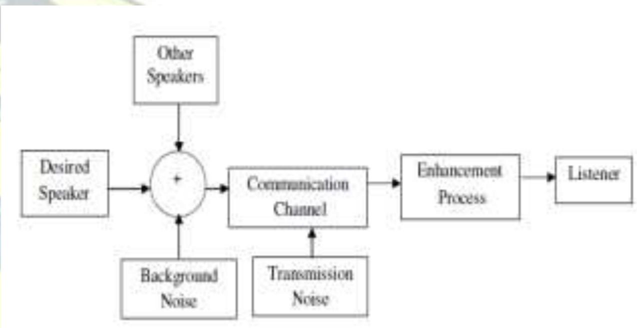
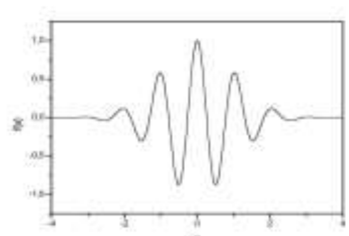


Fig.1 Common Sources of Noise

III. Wavelets

A. *Wavelet Definition*

A wavelet is a wave-like oscillation with an amplitude that begins at zero, increases, and then decreases back to zero. It can typically be visualized as a "brief oscillation" like one might see recorded by a seismograph or heart monitor. Generally, wavelets are purposefully crafted to have specific properties that make them useful for signal processing. Wavelets can be combined, using a "reverse, shift, multiply and integrate" technique called convolution, with portions of a known signal to extract information from the unknown signal.



ISSN 2394-3777 (Print)
ISSN 2394-3785 (Online)
Available online at www.ijartet.com

INTERNATIONAL JOURNAL OF ADVANCED RESEARCH TRENDS IN ENGINEERING AND TECHNOLOGY (IJARTET)
VOL. II, SPECIAL ISSUE VIII, FEBRUARY 2015 IN ASSOCIATION WITH
SRI KRISHNA COLLEGE OF ENGINEERING AND TECHNOLOGY, COIMBATORE
FOURTH NATIONAL CONFERENCE ON COMMUNICATION TECHNOLOGY INTERVENTIONS FOR RURAL AND SOCIAL DEVELOPMENT (NCC –TIRD 2015)-DEPARTMENT OF ECE
7TH FEB 2015

Fig.2 Wavelet

B. Wavelet Transform

The Wavelet transform is similar to the Fourier transform (or much more to the windowed Fourier transform) with a completely different merit function[5]. The main difference is this: Fourier transform decomposes the signal into sines and cosines, i.e. the functions localized in Fourier space; in contrary the wavelet transform uses functions that are localized in both the real and Fourier space. Generally, the wavelet transform can be expressed by the following equation:

$$F(a,b) = \int_{-\infty}^{\infty} f(x)\psi^{*}_{(a,b)}(x)dx$$

where the * is the complex conjugate symbol and function ψ is some function. This function can be chosen arbitrarily provided that obeys certain rules.

C. General Procedure for Wavelet Denoising

The general wavelet denoising procedure is as follows.

- Apply wavelet transform to the noisy signal to produce the noisy wavelet coefficients.
- Select appropriate threshold limit at each level and threshold method (hard or soft thresholding) to best remove the noises.
- Inverse wavelet transform of the thresholded wavelet coefficients to obtain a denoised signal.

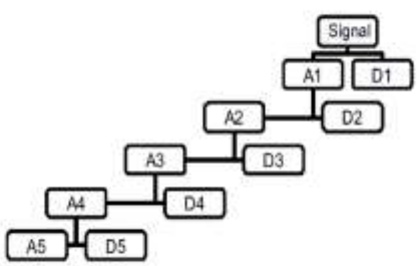


Fig.3 Wavelet tree decomposition

D. Wavelet Packet Transform

WPT is a generalization of wavelet decomposition that offers a wider range of possibilities for signal analysis. WPT is an expansion of classical wavelet decomposition that presents more complex and flexible analysis because in WPT[5]analysis the Details (D) as well as the Approximations (A) are separated like complete binary tree. For each level of decomposition, the signal is filtered into approximation information of the signal which contain lower frequency component and detail information which contain lower frequency component. Figure 4 shows the wavelet packet decomposition tree obtained up to 3 level of decomposition.

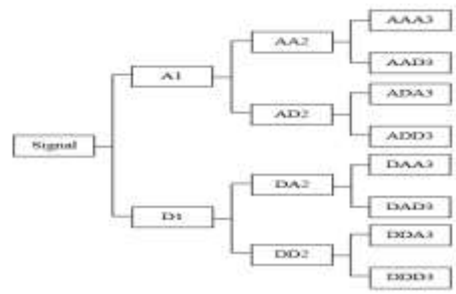


Fig.4 Tree structure of the wavelet packet decomposition at level 3

The top level of WPT is the time representation whereas the bottom level is the better frequency resolution. Hence with use of WPT[5] a better frequency resolution can be obtained for the decomposed signal. At first level, both algorithms are using low pass filter, g(n) and a high pass filter, h(n), to generate two sets of coefficients (approximation coefficients and detail coefficients). Here, the convolution process is applied between disturbance signal with a low pass filter, g(n) and a high pass filter, h(n), respectively, which followed by down sampling by 2. With MRA based on WPT[6] the reconstructed output signals will have equal frequency bands, because both outputs from the high pass and low pass filters are further decomposed. Both processes can be repeated multiple times to obtain more detailed signals at higher levels. Implementation of decomposition a signal into 2levels of MRA based on WPT is shown in Figure view of the fact that WPT generates large number of nodes it increases the computational burden. In DWT only approximations are further decomposed thus reducing the level of decomposition and thereby computational attempts. Because of the criterions, WPT is chosen since it may be efficiently searched for best basis as the events waveform is analyzed.

ISSN 2394-3777 (Print)
ISSN 2394-3785 (Online)
Available online at www.ijartet.com

INTERNATIONAL JOURNAL OF ADVANCED RESEARCH TRENDS IN ENGINEERING AND TECHNOLOGY (IJARTET)
VOL. II, SPECIAL ISSUE VIII, FEBRUARY 2015 IN ASSOCIATION WITH
SRI KRISHNA COLLEGE OF ENGINEERING AND TECHNOLOGY, COIMBATORE
FOURTH NATIONAL CONFERENCE ON COMMUNICATION TECHNOLOGY INTERVENTIONS FOR RURAL AND SOCIAL DEVELOPMENT (NCC –TIRD 2015)-DEPARTMENT OF ECE
7TH FEB 2015

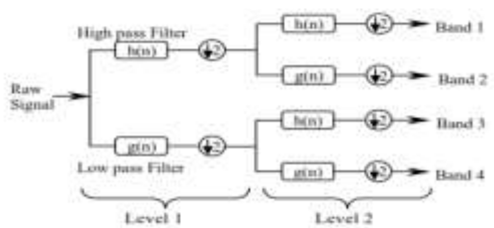


Fig.5 Decomposition of original signal into second levels of MRA based on WPT

E. Stationary Wavelet Transform

The main feature of SWT is that it is a time-invariant transform. Shift-invariance is important in many applications such as change detection, denoising and pattern recognition. The SWT[10] algorithm is close to the DWT, but the down sampling operation after filter convolution is suppressed. The decomposition obtained is then a redundant representation of the signal. The benefit of this redundant representation over the memory-efficient decimated DWT is the reduction of artifacts at discontinuities and irregularities in reconstructed signals. These artifacts are caused by unpredictable changes in coefficients with different time shifts. Figure.6 shows the general decomposition step and Figure.7 shows the stationary wavelet filter calculations. CAj and CDj represent the approximation and detail coefficients of level j, respectively and Hj and Gj represent the low- and high pass decomposition filters at the same level. There is no unique Inverse Stationary Wavelet Transform (ISWT) [10], therefore the inverses obtained for every non-decimated DWT are averaged. This can be done recursively, starting from level j down to level 1. In this way the main features of the analyzed signal is captured.

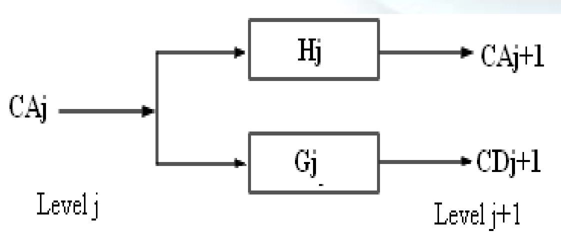


Fig.6 Stationary Wavelet Decomposition

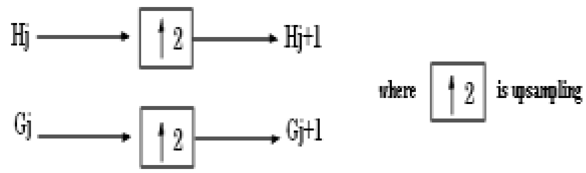


Fig.7 SWT Filter calculation

F. Steps in Stationary Wavelet Transform Denoising

- Load a signal
- Perform a stationary wavelet decomposition of a signal
- Construct approximations and details from the coefficients
- Display the approximation and detail at level 1
- Regenerate a signal by using inverse stationary wavelet transform
- Perform a multilevel stationary wavelet decomposition of a signal
- Reconstruct the level 3 approximation
- Reconstruct the level 1, 2, and 3 details
- Reconstruct the level 1 and 2 approximations
- Display the results of a decomposition
- Reconstruct the original signal from the level 3 decomposition
- Remove noise from a signal

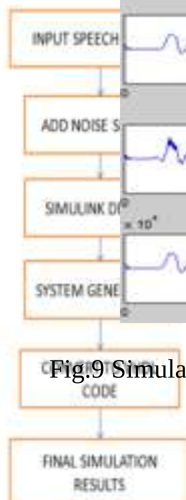
G. SWT De-Noising Procedure and Principles

The general de-noising procedure involves three steps. The basic version of the procedure follows the steps described below.

1. Decompose : Choose a wavelet, choose a level N. Compute the wavelet decomposition of the signal s at level N.
2. Threshold detail coefficients : For each level from 1 to N, select a threshold and apply soft thresholding to the detail coefficients.
3. Reconstruct : Compute wavelet reconstruction using the original approximation coefficients of level N and the modified detail coefficients of levels from 1 to N.

IV. Flow Diagram of The Proposed Work

ISSN 2394-3777 (Print)
ISSN 2394-3785 (Online)
Available online at www.ijartet.com
INTERNATIONAL JOURNAL OF ADVANCED RESEARCH TRENDS IN ENGINEERING AND TECHNOLOGY (IJARTET)
VOL. II, SPECIAL ISSUE VIII, FEBRUARY 2015 IN ASSOCIATION WITH
SRI KRISHNA COLLEGE OF ENGINEERING AND TECHNOLOGY, COIMBATORE
FOURTH NATIONAL CONFERENCE ON COMMUNICATION TECHNOLOGY INTERVENTIONS FOR RURAL AND SOCIAL DEVELOPMENT (NCC –TIRD 2015)-
DEPARTMENT OF ECE
7TH FEB 2015



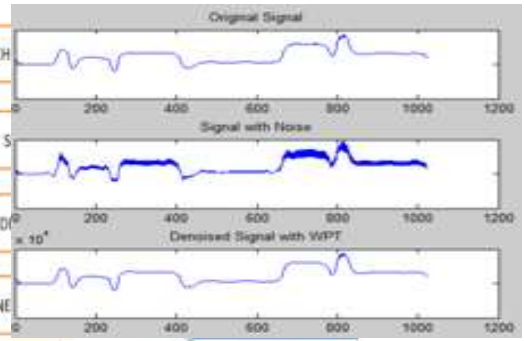


Fig.9 Simulation result using Wavelet Packet Transform

METHOD	MSE	SNR (dB)
WPT	2.7675	0.8272
SWT	2.0231	1.5789

V. RESULTS AND DISCUSSION

A. Simulation Results

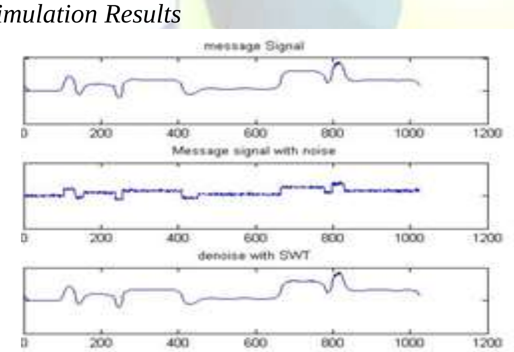


Fig.8 Simulation result using Stationary Wavelet Transform

The result obtained for a randomly generated signal using Stationary Wavelet Transform and Wavelet Packet Transform is shown in the figure. The first plot shows the randomly generated input signal, the second plot shows the input signal with white noise, since it has equal power at all frequencies and the third plot shows the denoised output signal with Stationary Wavelet Transform and Wavelet Packet Transform. And SNR and MSE for both the methods were tabulated.

B Based on the tabulation it is clearly shown that Stationary Wavelet transform is far better than Wavelet Packet Transform. So we can proceed with SWT for further implementation purposes.

5.2 Simulink and Xilinx based Design with its Output

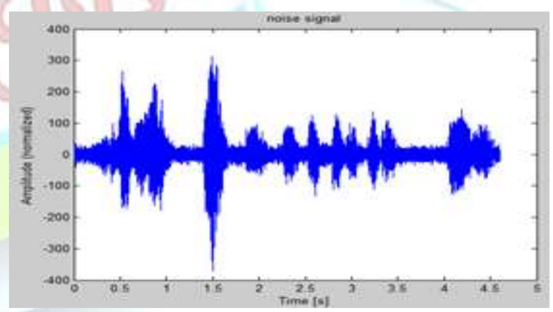


Fig.10 Input Noisy speech Signal

Fig.10 Shows the input noisy speech signal and Fig.11 shows the the selective noise signal which is fed as an output to simulink design .

ISSN 2394-3777 (Print)
ISSN 2394-3785 (Online)
Available online at www.ijartet.com
INTERNATIONAL JOURNAL OF ADVANCED RESEARCH TRENDS IN ENGINEERING AND TECHNOLOGY (IJARTET)
VOL. II, SPECIAL ISSUE VIII, FEBRUARY 2015 IN ASSOCIATION WITH
SRI KRISHNA COLLEGE OF ENGINEERING AND TECHNOLOGY, COIMBATORE
FOURTH NATIONAL CONFERENCE ON COMMUNICATION TECHNOLOGY INTERVENTIONS FOR RURAL AND SOCIAL DEVELOPMENT (NCC –TIRD 2015)-
DEPARTMENT OF ECE
7TH FEB 2015

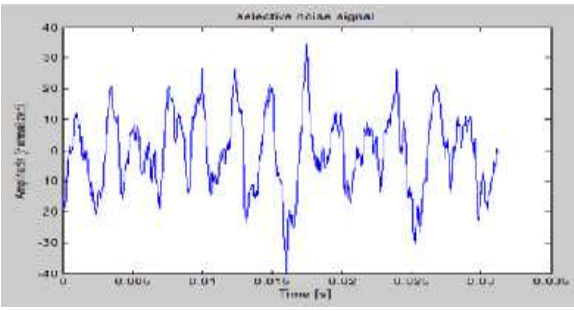


Fig.11 Selective Noise signal

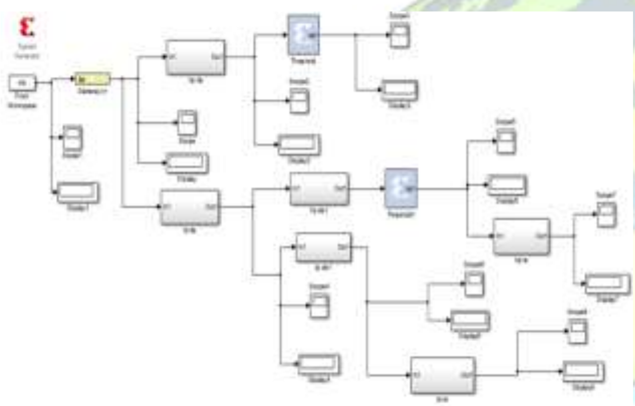


Fig.12 Simulink Design

VI. CONCLUSION

The denoised signal is obtained using Wavelet Packet Transform (WPT) and Stationary Wavelet Transform (SWT). The denoised signal obtained using Wavelet Packet Transform (WPT) and Stationary Wavelet Transform is implemented in XILINX VIRTEX-5 FPGA kit. The output obtained from Stationary Wavelet Transform method will be finally implemented in XILINX VIRTEX-5 FPGA kit and the parameters will be measured and compared. Using System Generator the Simulink design will be converted into Vhdl code.Vhdl Code will be configured with Xilinx and implemented using Xilinx Virtex-5 FPGA.Power, Area, Processing Speed, Processing time can be obtained from the synthesis report.

REFERENCES

[1] Nivin Ghamry,"FPGA Implementation of Hearing Aids based on Wavelet Packets "Journal of Computers,Vol7,No.3,March 2013.
[2] Galya Georgieva- Tsaneva, Krassimir Tcheshmedjiev,"Denoising of Electrocardiogram data with methods of Wavelet Transform", 2013-IEEE vol. 40, no. 5, pp. 1068–1077.
[3] Jaishankar and Duraisamy.K,"Audio Denoising using Wavelet Transform",2012-11th IEEE Int. Conf. Electron., Circuits Syst., pp.5–8.
[4] Rajeev Agarwal & Jai Karan Singh," Elimination of White noise from Speech Signal using Wavelet Transform by Soft and Hard Thresholding" VSRD-IJEECE, vol. 1 (2), 2011, 62-71(2011).
[5] Slavy G.Mihov, Ratcho M. Ivanov, Angel N.Popov,"Denoising Speech Signal by Wavelet Transform" Annual Journal Of Electronics, 2009, ISSN 1313-1842.
[6] H.Lewitt, "Noise reduction in hearing aids" ,a review,Journal of Rehabilitation of Research and Development (2006).
[7] Sara Grassi, Alexandre Heubi, "Study of a VLSI Implementation of a Noise Reduction Algorithm for Digital Hearing Aids",7th European Signal Processing Conference EUSIPCO, 1661-1664, (2005).
[8] B.C.J Moore, Basic auditory processes involved in the analysis of speech sounds, Philosophical. Transaction of the Royal Society 363 (2008), 947-963.
[9] D. L. Donoho, and I.M. Johnstone, Ideal spatial adaptation via wavelet shrinkage, Biometrika81 (1994), 425-455.
[10] P. Nason, B.W. Silverman, "The stationary wavelet transform and some statistical applications", Lecture Notes in Statistics 103 (1995), 281-299.

BIOGRAPHY

R.SINDHUJAA received the B.E degree in Electronics and Communication Engineering from Anna University, Chennai, in

ISSN 2394-3777 (Print)
ISSN 2394-3785 (Online)
Available online at www.ijartet.com
INTERNATIONAL JOURNAL OF ADVANCED RESEARCH TRENDS IN ENGINEERING AND TECHNOLOGY (IJARTET)
VOL. II, SPECIAL ISSUE VIII, FEBRUARY 2015 IN ASSOCIATION WITH
SRI KRISHNA COLLEGE OF ENGINEERING AND TECHNOLOGY, COIMBATORE
FOURTH NATIONAL CONFERENCE ON COMMUNICATION TECHNOLOGY INTERVENTIONS FOR RURAL AND SOCIAL DEVELOPMENT (NCC –TIRD 2015)-
DEPARTMENT OF ECE
7TH FEB 2015

2012. Currently, She is pursuing M.E degree in VLSI DESIGN from Kongu Engineering College affiliated to Anna University, Chennai. Her research areas of Interests include VLSI, Speech Signal Processing.

K.KAVIN KUMAR received the B.E degree in ECE from Vellalar Institute of Engineering and Technology, Erode in 2007 and M.E VLSI DESIGN in Shree Sastha Institute of Engineering and Technology, Chennai in 2011.Currently,he is pursuing Ph.D in Anna University, Chennai. He is now working as an Assistant Professor in Department of ECE for 3 years at Kongu Engineering College, Perundurai. His research areas of Interests include VLSI, Signal Processing.