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Development of Adaptive De-noising Algorithm using Wavelet Technique for a Linear FM Acoustic Signal

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Abstract

Background/Objectives: It is highly important to analyze on the challenges that influence the acoustic medium. The underwater signals are affected by variety of background noise (ambient noise) in the ocean which is both natural and man-made. **Method/Statistical Analysis:** The methodology involved in this research work is wavelet decomposition technique to reduce the underwater noise present in the acoustic signal to extract the details present in it. **Findings:** In this research work, a detailed knowledge on the ambient noise which was collected from the shallow water region of Bay of Bengal, was obtained by characterizing it and a suitable denoising algorithm was formulated using wavelet technique, in particular Gabor wavelet and improvement in SNR is verified using MAT lab simulink tool. **Applications/Improvements:** The applications of underwater acoustics includes exploration of the environment, monitoring and tracking the marine mammals, exploration of oil in the deep ocean, monitoring the submarines and underwater autonomous vehicles etc,. A comparison was done between Gabor wavelet and Symlet wavelet in denoising the noisy acoustic signal to improve the signal to noise ratio. It enhances the performance of Gabor wavelet. From the result, we can understand that, for an input SNR range of -15 db to 0 db, we obtained an output SNR in the order of 9dB, 15db and 14db at 20 KHz, 66 KHz and 86 KHZ respectively which enhances the performance of Gabor wavelet.

Keywords: Underwater Acoustic Signal Processing, Underwater Ambient Noise, Underwater De-Noising Techniques, Wavelet Decomposition

1. Introduction

The breath taking beauty of nature is primarily because of the water body i.e. ocean. Ocean is believed to be the lifeblood of our universe because it is the one which is regulating temperature, driving weather and eventually it reinforces all the living creatures. Yet we have explored only five percent of the ocean which means 95% of the underwater treasure is deep under the sea. We can unlock these secrets of the ocean only by having a good and deep understanding of the physical, biological, geological and chemical characteristics of this water body. Many expeditions are carried out all over the world to investigate and document on this unexposed area. In order to explore these secrets we have to deeply understand, analyze and evaluate the popular findings and theories concerning the ocean. Transmission and reception of signals at high speed is a problematic one, as it has its own drawbacks like conducting nature of the medium, temperature gradient, propagation of signal in multiple paths, restricted bandwidth and large Doppler spread.

To explore the wealth of ocean, we should have a good knowledge in underwater acoustics and sound transmission.

The applications of underwater acoustics includes exploration of the environment, monitoring and tracking

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the marine mammals, exploration of oil in the deep ocean, underwater sensor network , monitoring the submarines and underwater autonomous vehicles etc,.

It is highly important to analyze on the challenges that influence this acoustic medium. Some of the important factors are:

- Path loss because of attenuation and geometric spreading.
- Man made and ambient noise.
- Multipath propagation and multipath geometry.
- High delay and delay variance.
- Doppler spread.

In order to leave no stone regarding the underwater, we have to use the apt physical phenomena such as radio waves, optical waves and acoustics for our research. The radio waves in the microwave frequency get absorbed by the sea water immediately within feet of their transmission. Hence the antenna size for transmitting the radio waves should be very large and also its high attenuation makes itself unsuitable for underwater communication. The optical waves do not get suffered by attenuation but they undergo a severe scattering. Only lasers with extreme intensity can only travel in water.

The propagation of sound in the underwater, in particular shallow water region is a challenging one because of its sensitivity to depth of water, distance between the source and receiver, slope of the bottom layer which leads to variation in the impulse response of the acoustic signal.

From the literature review we understand that, sound waves travel about five times faster in water with the speed of 1520m/s when compared to air. As we go below the water surface, the speed of the sound decreases as the temperature of the water decreases. This continues till the thermo cline region. Under this zone, there is no change in temperature but there is an increase in the pressure, which causes the increase in sound speed. The wavelength of the sound is clearly defined as ratio between the speed of the sound and the frequency. Hence the wavelength of the sound in water (1500/20=75) is more than in the air (340/20=17). The underwater sounds reveal many information like global warming, earthquakes, flow of magma under the sea during the volcanic explosion etc. In this thesis, we shall have a detailed study of the ambient noises which is location specific, that is the noise present in Bay of Bengal will not be the same as in Arabian Sea. Hence we have chosen Bay of Bengal for our research work. This detailed study will include the original sea truth data taken from the Bay of Bengal of 30m depth, which will help us in designing the adaptive de-noising filter by characterizing the ambient noise. A complete study which includes frequency spectrum of the predominant wind driven noise with various parameters wind speed, depth of ocean; coherence length, directionality, etc. is needed. The results obtained clearly shows that the selection of NSL, frequency band, coherence length, directionality are the vital parameters in the design of de-noising algorithm[1]. Hence it is highly important to remove the noise present in the received acoustic signal so as to appreciate the important features of the signal using match filtering technique[2].

Analyzing the data has been successfully done by wavelets, which is a powerful tool in removing noise from the various type of source signal. In wavelet analysis, the noise levels are analyzed separately at each wavelet scale and the denoising algorithm are adapted accordingly. The main advantage of wavelet transform is that, it provides the local characterization of signals, which is not permitted in Fourier transform and STFT.

Wavelet packet transform in automatic transient signal classification is done through the development of a simple non-coherent feature extraction procedure for biologically generated acoustic signal in ambient noise[3,4]. Continuous wavelet transform has the capability of multi resolution analysis and hence is an attractive method for transient signal analysis[5].

Wavelet transform is suitable only to analyze the information in low frequency band and not suitable for high frequency band. But this limitation is overcome by wavelet packet transform as it is suitable to identify and appreciate the information in both the low and high frequency bands. This property makes it an ideal processing tool for the non-stationary transient signals. This property can also be explained in detail as; it has high frequency resolution and low time resolution in low frequency region of the signal and low frequency resolution and high time resolution in high frequency region[6].

Wavelet packet decomposition is a wavelet transform, in which the signal is passed through large number of filters when compared to that of discrete wavelet transform. It incorporates a linear combination of wavelets. These wavelets retain the properties like orthogonality, smoothness and localization of the mother wavelets. WPD decomposes both the detail and approximation co-efficient which overcomes the shortage of decomposing only the approximation co-efficient in DWT. WPD is best suited for signals which have oscillatory and periodic behavior, since it provides much freedom in deciding the basis function which is suitable for representing the given function.

In simple we can say that Wavelet Packet Decomposition (WPD)[7] is an extension of wavelet transform which allows best matched analysis to the signal. Hence it decomposes the received signal and thresholds are used to select the co-efficient, to remove the noisy part of the signal and a noise free signal can be synthesized.

Gabor wavelets have better time- frequency localization when compared to other wavelet packets and also they can be further translated over a finer time- frequency grid. Gabor wavelet is obtained with the Gaussian window. The noisy signal is spread over the spectral channels in which the co-efficient below the determined noise threshold is filtered out. The threshold is calculated by means of higher order statistics [8-10].

This part of research work deals with the development of denoising algorithm for the linear frequency modulated input signal which is added up with the original noise data collected from Chennai, the region of bay of Bengal. This paper also aims at discussing the performance of Gabor wavelet in denoising and improving the signal to noise ratio when compared to Symlet wavelet. The results are encouraging.

2. Underwater Ambient Noise

In the frequency range between 500-20 kHz, the wind noise is predominant over the ship and marine animal noises. And also it was well proved by Knudsen et al.[11], that there is a correlation between wind speed and under water ambient noise. Followed by the research work done by Wenz[12], gave the dependence of ambient noise on the wind speed and sea state condition. The ambient noise is characterized based on the frequency levels, which is given in Figure 1 (Composite of ambient noise spectra[12]).

The noise produced by the distant shipping becomes the main source of ambient noise in the frequency range of 20-500 Hz. After the removal of the noise created by the ships near by the receiver, we can identify some distant ships. Increase in the shipping traffic increases the noise in that region. The spray and bubble noise associated with the breaking waves, contribute to the ambient noise in the frequency range of 500 Hz-100,000 Hz. increase in

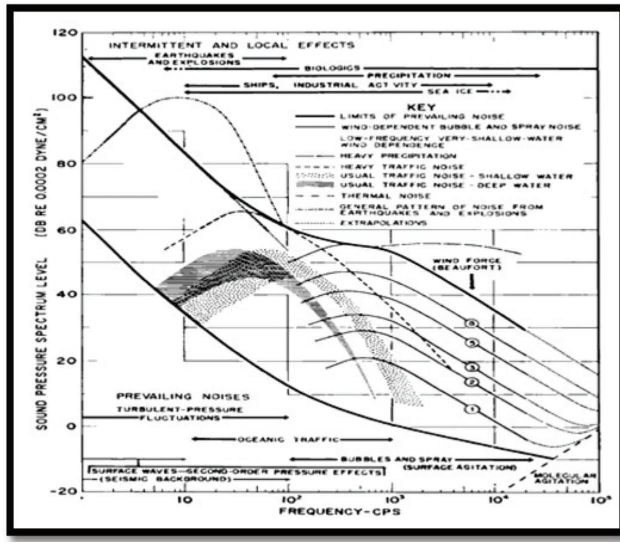


Figure 1. Composite of ambient noise spectra[12].

the wind speed, increases these noise. The thermal noise, which is caused by the random movement of the water molecules, dominates the other n noises in the frequency range greater than 100,000 Hz.

He ambient noises are caused by the intermittent sources of noise like biological noises due to whales, dolphins, porpoises, croakers, snapping shrimp and non biological sources like rain, earthquakes, explosions, volcanoes and surf etc.

3. Ambient Noise Measurement

It has been understood from the various literature that ambient noise collection from the ocean is a complicated process and hence selection of hydrophone, signal conditioning modules and data collection modules has to be given primary importance. During the measurement, we face the problem like separation of the self noise from the required ambient noise, which falls in the measurement band. The removal of this system noise is important because, these may introduce different spectral components.

Under water ambient noise is location specific, that is, the noise in particular region of Bay of Bengal is not the same as that in the region of Arabian Sea. Keeping this in mind, for this research work, the data has been collected from the location of Chennai, Bay of Bengal. The experimental set up includes a reasonably equipped boat to make the measurement at a depth of 30 m depth.

3.1 Experimental Description

The experimental set up given in Figure 2 (Experimental setup for the measurement in the sea) includes the boat which is equipped with two Omni directional hydrophone sensor with sensitivity of -170 db with re 1V/μPa, which covers the frequency range of 0.1 Hz to 25 KHz, Data acquisition system and power supply. The ambient noise where taken at a depth of 30 m which comes under the shallow water region.

The two hydrophones where placed in the "L" shaped PVC pipe with concrete filled in it. This arrangement is to ensure that the hydrophone has to sink because of its self weight in the water at 30 m depth and should not float or drift which may introduce some other irrelevant details.

The other specifications of the measurement systems are provided in Table.1

4. Noise Reduction Algorithm for Underwater Signals

The underwater acoustic signal is denoised in this research work by following three important procedures by means of which the noisy signal gets denoised automatically.

- Wavelet transformation of underwater acoustic noisy signal.
- Determination of threshold co-efficient.

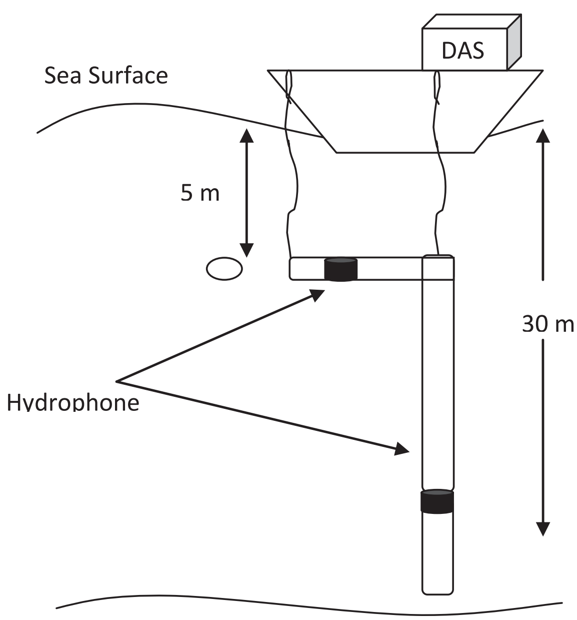


Figure 2. Experimental setup for the measurement in the sea.

Table 1. Hydrophones specifications

Parameter	Specifications
Frequency of operation	Up to 25kHz
Directivity	Omni directional
Sensitivity	−170dB
Type of material	Piezoelectric (PZT)
Operating depth	600 m
Survival depth	700 m
Operating Temperature range	−2 to + 55 C
Operating Voltage	12 to 24 VDC

- The de-noised or reconstructed signal is obtained by inverse wavelet transformation.

Underwater de-noising techniques are classically based on the projection of the ambient noisy signal on a new space, in which the signal and the noise do not overlap. The ambient noise in view of new space projection is eliminated by preserving the signal into subspace only. Applying into inverse projection technique, a de-noised form of the original signal is recovered.

In view of applying projection techniques, is the class of unitary transforms since they have the more useful properties for underwater signal processing. Among all, unitary transforms assure that the existence of an inverse transform technique and preserve the acoustic signal energy on the transformed space. Figure 3 shows the Wenz model[12] of the power spectral density of the underwater ambient noise.

The underwater de-noising techniques considered here are based on this framework: the acoustic signal is projected on a new space with a unitary transform technique, properly filtered in this new space and, with the inverse unitary transform technique, estimated back to the original signal space.

4.1 Wavelet Packet Transforms

Wavelet transforms and wavelet packet transforms are also known as time-frequency transforms are very much useful in studying the non-stationary phenomena and characterization of transient signals. Transient signals are the one which will exist only for a short duration of time, is of particular concern, as it will appear as a broadband energy on the frequency display. The standard spectral analysis method is not suitable for the frequency display of the transient signals. Fourier analysis is a supreme tool

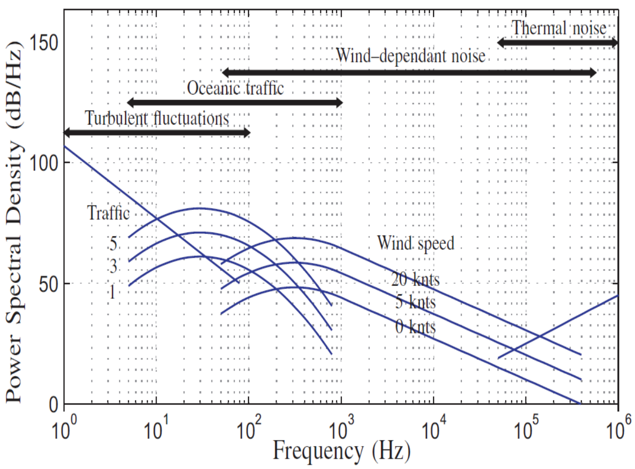


Figure 3. Wenz model[12] of the power spectral density of the underwater ambient noise.

in analyzing signals. But it is suitable for narrow band signals. Also, the STFT[13] is not suited for obtaining the signal characteristics of this type of transient signals[14], because of its non varying window.

The wavelet packet transforms offers a great freedom in analyzing these transient signals. Wavelet packet decomposition can be called as a natural extension of wavelet transforms which provide a level by level transformation from time domain to frequency domain. As the decomposition level increases, there is decrease in the time resolution and corresponding increase in frequency resolution. The basis functions in the WPD are produced from the mother wavelet by scaling and translation operation. This transform isolates signal variability both in time t, and also in "scale" s, by rescaling and shifting the analyzing wavelet.

The main difference between WT and WPD is the decomposition process. In WT, only the low frequency band is decomposed. But in WPD, both the low and high frequency bands are decomposed into uniform frequency bands.

4.1.1 Decomposition Process

The approximation and detail co-efficient in the WPD is obtained by the following algorithm

$$\mathrm{u}_{2m}^{i}(n) = \sum_{\mathrm{k}} \mathrm{h}(\mathrm{k}-2\mathrm{n})\mathrm{u}_{m}^{i-1}(k) \quad (1)$$

$$\mathrm{u}_{2m+1}^{i}(n) = \sum_{\mathrm{k}} \mathrm{g}(\mathrm{k}-2\mathrm{n})\mathrm{u}_{m}^{i-1}(k) \quad (2)$$

Here h (k) represents the detail co-efficient and g (k) represents the approximation co-efficients.in WPD both these co-efficient are decomposed.

4.1.2 Reconstruction Process

The process of reconstructing the original signal from its co-efficient is called synthesis process. It is done by performing the decomposition process in reverse order. The signals are interpolated by 2 at every level and then passed through the reconstruction filter and then added up.

$$\mathrm{u}_{m}^{j\text{-}1}(n) = \sum_{\mathrm{k}} \mathrm{H}(\mathrm{k}-2\mathrm{n})\mathrm{u}_{2m}^{i}(k) + \sum_{\mathrm{k}} \mathrm{G}(\mathrm{k}-2\mathrm{n})\mathrm{u}_{2m+1}^{i}(k) \quad (3)$$

4.1.3 Thresholding

It is a non linear technique, in which the wavelet coefficients are threshold on comparing with determined threshold at each level. Thresholding is a signal estimation technique, in which the noise is removed by properly determining the threshold, which determines the efficacy of denoising to a great extent.

Let us consider a decomposition of u into u_i elements and the resulting vector after thresholding as $D_T(u)$ with the threshold T. The wavelet threshold coefficients are obtained using either hard or soft thresholding functions.

Hard thresholding: $D_T(u_i) = 0\ ;\ |u_i| < T$

$$= u_i\ ;\ |u_i| > T$$

Soft thresholding: $D_T(u_i) \ = 0\ ;\ |u_i| < T$

$$= \mathrm{sign}\ (u_i) * (|u_i| - T) > T$$

The estimation of the noise threshold is based on the higher order statistics. The RMS value of the spectral channel is calculated by

$$S_{\sigma}(t) = \sqrt{\frac{1}{N}\sum_{i} x_{v_i}^{2}(t)},$$

Using this, the quantity of energy is calculated at a particular time over the spectral channel. The energy is low when the noise alone is present and there will be a considerable high energy when both the signal and noise are present. Based on this RMS value, the coefficients below the maximum RMS probability function are removed.

4.1.3.1 Window Functions

In this research work, two window functions are compared, namely Gabor and Symlet functions.

- **Gabor Transform**

In the Gabor transform, the signal is multiplied with the window w (t) function which is Gaussian and the Fourier transform is applied to the windowed signal.

It can be expressed in simple as

$$\Psi(t) = \frac{1}{(\sigma^2 \pi)^{1/4}} e^{-\frac{t^2}{2\sigma^2}} * e^{j\eta t}$$

The sample Gabor wavelet is shown in Figure 4 (Gabor wavelet).

- **Symlet Wavelet**

For comparative purpose, Symlet filter is also taken for the study. The Symlet are nearly symmetrical, orthogonal and biorthogonal wavelets proposed by Daubechies as modifications to the db family. The properties of the two wavelet families are similar. Wavelet is shown in Figure 5 (Symlet wavelet).

The spectral channel of a noisy signal exhibits different spectral characteristics. In the time - frequency plane, these components are characterized by the high and concentrated co-efficient whereas the unwanted noise is represented by the low and spread co-efficients.by removing these spread co-efficient, the noise gets removed.

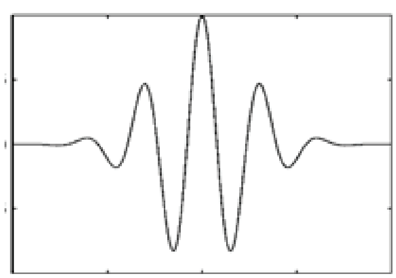


Figure 4. Gabor wavelet.

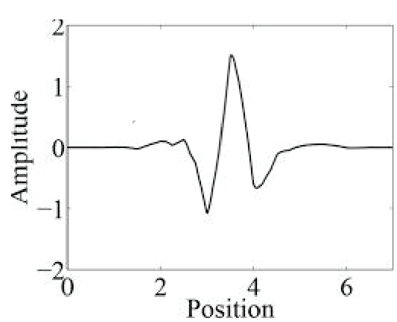


Figure 5. Symlet wavelet.

5. Simulation and Experimental Results

A gated linear Frequency modulated signal (acoustic) X (t) is considered to be the input signal through the water channel of mid frequency range. The simulated input signal is given in Figure 6 (Gated Linear FM signal). For this particular work, the simulated input signal is mixed with the original noise data, which is assumed to be stationary and Gaussian, with reference to Wenz-shaped power spectral density, collected from the region of Bay of Bengal, Chennai. Figure 7 (Time series of the raw data) shows the time series of noise data. The signal has got an improvement in the SNR output of 8 dB for the input range of -15 dB to 0 dB, when simulated at 1 kHz with Gabor wavelet[21]. The simulation of the input signal is done at 20 kHz, 66 kHz and 86 kHz. The results have been obtained with 50 different noise realizations and have to be taken in an average. By means of undergoing the proposed denoising algorithm, the unwanted noises have been removed from the useful signal. The proposed de-noising method or the simulation procedure shown in Figure 8, reduce the noise and recover the useful information in the signal.

The performance of a communication system is determined by the signal to noise ratio.

Figure 9 (*Improvement of SNR at 20 kHz, 66 kHz and 86 kHz*) shows input SNR versus the output SNR

$$\mathrm{SNR}_{IN} = 10 * \log_{10} \frac{P_i}{P_n} \text{ (dB)} \tag{10}$$

$$\mathrm{SNR}_{OUT} = 10 * \log_{10} \frac{P_o}{P_n} \text{ (dB)} \tag{11}$$

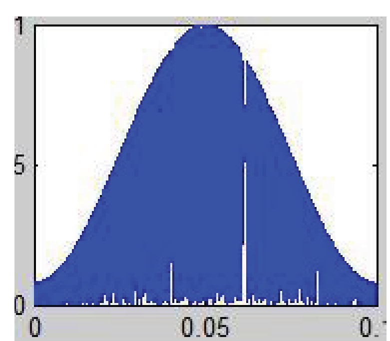


Figure 6. Gated Linear FM signal.

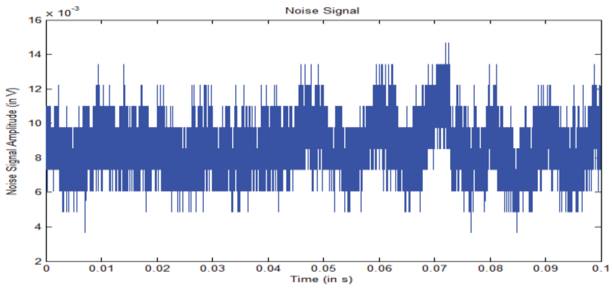


Figure 7. Time series of the raw signal.

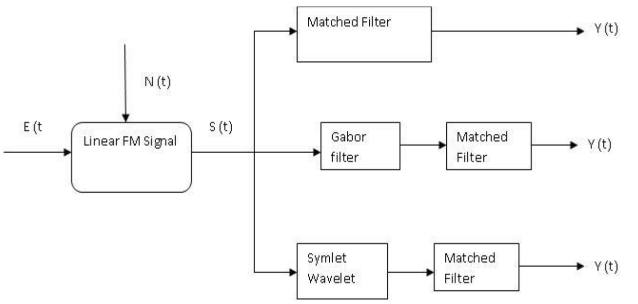
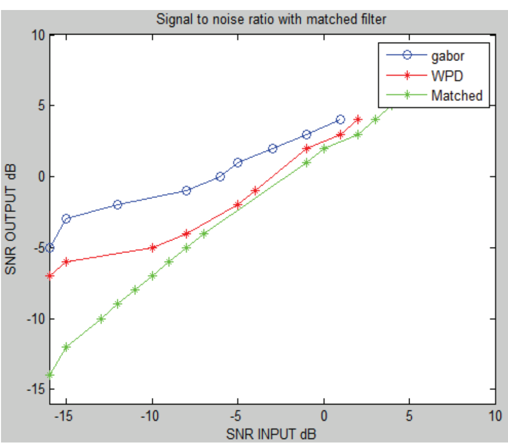
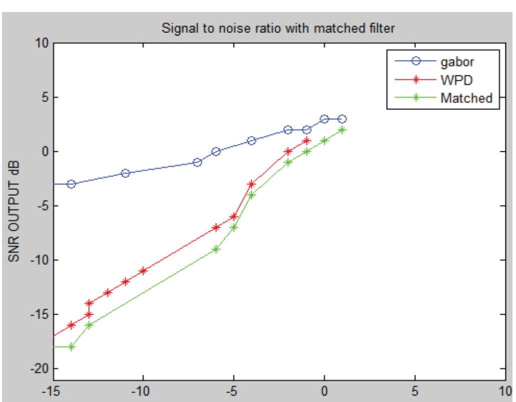


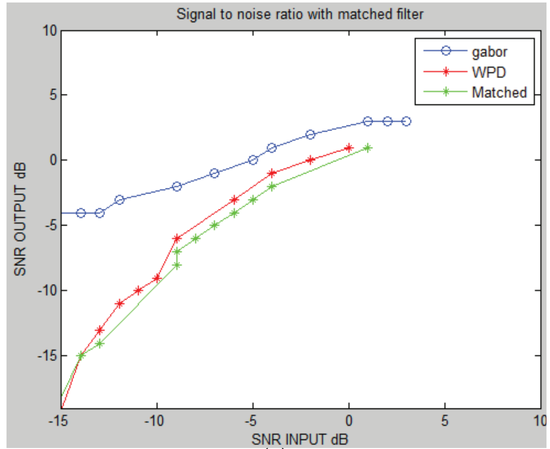
Figure 8. Schematic representation of the simulation procedure.



(a)



(b)



(c)

Figure 9. Comparative analysis of the de-noising procedures to improve SNR at 20kHz,66 kHz and 86 kHz.

Here P_i represents the signal power before denoising,
P_o represents the signal power after denoising.
P_n represents the noise power.

All the proposed methods show a considerable improvement in SNR by reducing noise. But the Gabor transform tool shows a higher performance when compared to other techniques. The signal has got an improvement in the SNR output of 8 dB for the input range of -15 dB to 0 dB, when simulated at 1 kHz with Gabor wavelet[15]. It gives an improved output SNR is order of 9dB, 15dB and 14 dB when simulated at 20 kHz, 66 kHz and 86 kHz respectively in the range of input SNR -15dB to 0dB.

6. Results and Discussion

In this research work, the essential of the denoising technique is well established, because of the acoustic ambient noise environment in the underwater. In this connection, the data collection was made in Chennai, Bay of Bengal, with the passive hydrophone having specifications mentioned earlier. Here, for our research analysis, we have taken the input signal which is standard in any acoustic based instruments and sensor network[22], in ocean sector, a linear FM signal of 10Hz with weighted function. The simulation was carried out by mixing the signal with the collected noise data which is used for denoising process.

The denoising procedure includes several wavelets to remove the noise and the performance was discussed with the help of SNR improvement as shown in Figure 9 (a), 9 (b) and 9 (c). The results obtained was very encouraging as that in the range of input SNR -15dB to 0dB, the improved output SNR is order of 9dB, 15 dB and 14 dB for the simulation done at 20kHz, 66kHz and 86 kHz respectively. This has proved beyond doubt by the reconstruction method that the time series is nicely improved with the Gabor filter.

Future works concern a firmware implementation of a new class of Time Frequency filters adapted to underwater environment signals whose frequency content varies strongly in time.

Underwater sensor network include.

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