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W-STFRFT: Wiener Filtering in Short-Time Fractional Fourier Domain for Chirp Signal Enhancement

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Abstract: The major target of our research is to design and develop wiener filtering based on Short Time Fractional Fourier Transform for chirp signal enhancement. One of the efficient methods to analyze the chirp signal is fractional Fourier Transform (FRFT). Still, it fails in locating the Fractional Fourier Domain (FRFD) frequency contents which is required in some applications. For that reason in our research, we introduce the Short-Time Fractional Fourier Transform (STFRFT) for chirp signal enhancement. Using the advantage of STFRFT, we plan to design the wiener filtering based on STFRFT. In this study, at first the input signal is split in to two signals such as clean chirp signal and the noisy chirp signal based on the size of the hamming window. After that, multiplying the hamming window function with the frame of the signal is then multiplied with the Fractional Fourier Transform (FRFT). Finally, we apply the wiener filter to remove the noise from the signal. In experimental evaluation, we generate the mono signal model to compare the results of our proposed method.We compare our proposed technique with the wiener filter based on STFT, FFT and FRFT. Here, we obtain the maximum SNR of 30.79 db which is high compare to existing approach.

Key words: Short-time fractional fourier transform, chirp signal, wiener filter, SNR, MSE

INTRODUCTION

Typically, signal quality may considerably deteriorate in the presence of interference chiefly when the signal is also subject to reverberation. In many signal processing applications (Hlawatsch *et al*., 2000). The improvement or estimation of signals degraded by noise or interference is imperative. The objective of developing signal estimation algorithm requires diverse types of signals to validate the practical feasibility. Chirp signal is one of the potential signals used broadly for localization with time delay estimation. Because of its superior correlation property, chirp has a classic ability to decompose multipath into individual path. Generally, the estimation of chirp signal is done using Fourier transform which is significant to gain channel frequency response, imposes further complexity. Chirp assists us to minimize extra complexity by using de-chirping process because the complexity of de-chirping process is much smaller than the Fourier transform (Gannot *et al*., 2001; Jang *et al*., 2011). Chirp signal contains single tone signal whose immediate frequency differs linearly in time. These signals are habitually used in radar/sonar systems for ‘pulse compression’ where they are modulated by a short pulse to minimize the width of that pulse ambiguity function (Yeredor, 2004).

For chirp signal improvement, Wiener filtering is a classical model for the estimation of signal that has been implemented chiefly to one dimensional continuous signal with analysis and implementation based on continuous Fourier signal theory (Arcese *et al*., 2003). It is possible to carry out wiener filtering operations on time sampled signals and broaden the techniques (Pratt, 1972). To minimize the noise component, many chirp signal improvement algorithms are implemented in a transform domain. The motive for this is that it is normally easier to filter noise in the transform domain, as the signal is not present in equal power in all the transform coefficients. Hence, there are noise only coefficients which can be attenuated without overly distorting the underlying signals. Diverse transforms have been used for such intentions; some of the transforms are Discrete Fourier transform, Discrete Cosine transform, Fastest Fourier transform, Fractional Fourier transform, Wavelet transform, etc (Soon and Koh, 2003).

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The popular method for enhancing signal in the Fastest Fourier Transform (FFT) domain is Wiener filtering which reduces signal distortion (Venkateswarlu *et al*., 2011). The FFT is exploited in transform domain signal, audio, image and video compression. It has its own significance in the different fields. For such dynamic compute intensive and the applications that exploit huge data volume, the Graphics Processing Unit (GPU) based FFT algorithm can be the cost effective solution Ing (Soon and Koh, 2003; Izquierdo *et al*., 2002). In simple terms, the fractional Fourier Transform (FRFT) is a generalization of common Fourier transforms (Sejdic *et al*., 2011).

Using this, it allows the signals to be represented in intermediate domains amid time and frequency (Carlos and Margrave, 2003). In modern days the Fractional Fourier Transform (FRFT) become popular due to its examining and processing capability of non-stationary signals. The FRFT was constantly exploited in optics previously. De-noising may be carried out in time frequency domain (Ding and Yu, 2008). Due to non-stationary of signals and time-frequency identity of FRFT.

The motivation to exploit the wavelet as a possible choice for signal noise minimization is to find out new ways to minimize computational complexity and to achieve better noise reduction performance. The Fourier domain filters can be extended to the wavelet domain because they are derived based on the statistical properties of spectral components. Therefore, the filters are akin to the modern soft, hard or shrinking threshold models of wavelet de-noising that both operates on spectral magnitude and maintain the sign of wavelet transform coefficients Ningping. Varied from the tools mentioned above, it displays the time and FRFD frequency information together in time-FRFD frequency plane and provides the signal with 2-D support which is called Short Time Fractional Fourier Transform (STFRFT) support.

In this study researcher propose STFRFT based wiener filtering for mono and multi component chirp signal enhancement. The Short Time Fractional Fourier Transform (STFRFT) is multiplying the window function with the Fractional Fourier Transform (FRFT). Here, initially we set the size for the window function and split the noisy chirp signal and clean chirp signal based on the size of the window. The separated noisy chirp signal and the clean chirp signal are multiply with the hamming window and the solution is then multiply with the fractional Fourier transform. The solution we obtain from both clean chirp signal and noisy chirp signal are give as input to the wiener filter. The inverse short time fractional fourier transform is applied on the output of the wiener filter after processing both the input signals. We get the de-noised output after apply the inverse short time fractional fourier transform.

Literature review: Izquierdo *et al.* (2002) have enhanced noise of large grained materials of different algorithms. For the SNR improvement of ultrasonic signals to come from extremely scattering materials, an efficient technique that demonstrated is wiener filtering. These processing algorithms were derived from designed a filter that has large gain at frequencies where the SNR was high and low gain at frequencies where SNR was minute. However, these techniques do not considered two central ultrasonic effects: the finite-time duration of the flaw UT signal to come from a defect and the distortion of the frequency components of the traveling wave-front due to the dispersion. They have time-frequency Wiener filter takes into account these two characteristics. Experimental results showed that the time frequency algorithm has an stupendous performance on SNR enhancement.

Kutay *et al.* (1997) have presented time-invariant degradation and stationary signals and noise, the classical Fourier domain Wiener filter which could be applied in O(N logN) time, to provided the minimum mean-square-error to be estimated of the original undistorted signal. For time-varying degradations and non-stationary processes, the optimal linear approximation requires time for implementation. They considered filtering in fractional Fourier domains which enables significant reduction of the error compared with ordinary Fourier domain filtering for certain types of degradation and noise (especially of chirped nature), while required only O (N logN) implementation time. Thus, improved performance was achieved at no additional cost. Expressions for the optimal filter functions in fractional domains were derived and several illustrative examples were given in which significant reduction of the error (by a factor of 50) was obtained.

China have developed an implementation of employing Weiner filtering to signal processing. As has been previously mentioned, the purpose of these approaches was to reconstruct an output signal by making to be used for the accurate estimated ability of the Weiner filter. True enough, simulated results from the previous subdivision had proven that the Kalman filter indeed has the ability to be estimated accurately. Yao have presented an enhancement algorithm used temporal masking in the FFT domain. The

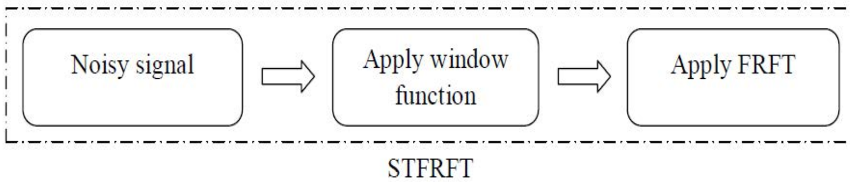

Fig. 1: Process of STFRFT

input signal was analyzed to be used FFT and then grouped into 22 critical bands. The noise power was estimated a minimum statistics noise tracking algorithm. A short-term temporal masking threshold was then calculated for each critical band and a gain factor for each band was then computed. The objective and subjective evaluations showed that the temporal masking based enhancement scheme outperforms the traditional Wiener filtering approach in the FFT domain.

Hadir *et al*. (2011) have presented a Wiener filter whose frequency response was optimized in the dimensionally reduced log-Mel domain. It was combination with Wiener filtering was motivated by the fact that signal reconstruction from log-Mel features sounds very unnatural. Hence, they correct only the spectral envelope and preserve the fine spectral structure of the noisy signal. Experiments on a Wall Street Journal corpus showed a relative improvement of up to 24% relative in PESQ and 45% relative in Log Spectral Distance (LSD), compared to Ephraim and Mallah's log spectral amplitude estimator.

Stankovic *et al*. (2003) have presented a signal-adaptive joint time-frequency distribution for the analysis of non-stationary signals. It was based on a fractional Fourier-domain realization of the weighted Wiener distribution has produced an auto-term close to the ones in the Wiener distribution itself but with reduced cross-terms. The computational cost of fractional domain realization was the same as the computational cost of the realizations in the time or the frequency domain, since the windowed fourier transform of the fractional fourier transform of a signal corresponds to the short-time fourier transform of the signal itself with the window being the fractional fourier transform of the initial one. The appropriate fractional domain was found from the knowledge of three second-order fractional fourier transform moments.

Benali have proposed the analysis of the signal and the identification of its parameters was an important step for diagnosis. In this study researcher present a new algorithm for ECG signal classification. Respiratory signal simultaneously recorded with the ECG signal will be used to classify each heart beat into two classes (abnormal and normal class) by the extraction of their parameters using various Multi-Layered Perceptron Neural Classifiers (MLPNNs). Principal Component Analysis (PCA) was used to reduce dimensions of input features and improve the performance of the neural classifiers. This algorithm was tested on Apnea-ECG database from the universal MIT Physio Net. As it will be shown later, the proposed algorithm allows to achieve high classification performances, describes both by sensitivity, specificity and the rate of correct classification parameters.

Problem statement and need for STFRFT: A powerful tool that can examine the chirp signal is Fractional Fourier Transform (FRFT). Because, the FRFD sampling theorem assures that the chirp signal can be sampled in lower rate contrast to the conventional Shannon sampling theorem (Xia, 1996; Tao *et al*., 2007, 2008). Higher concentration and lower sampling rate prepares the FRFT a more potential tool for analyzing chirp signal. It only discloses the overall FRFD frequency contents because it uses a global kernel. Sometimes we necessitate knowing not only the FRFD contents but also the changes happened in time to time. For instance, it is considered necessary in multiplexing in FRFD (Ozaktas *et al*., 1994) to distinguish and place the chirp pulses based on their FRFD frequencies. Therefore, the representation mixing both time and FRFD frequency information should be developed which is termed the Time-FRFD Frequency (TFFR).

The drawbacks using FRFT is that it fails in siting the fractional Fourier domain contents which are necessary for some applications. This issue can be Settled Using Short Time Fractional Fourier Transform (STFRFT). The STFRFT displays the time and fractional Fourier domain frequency data together in Short Time Fractional Fourier Domain (STFRFD). The STFRFT is formed by multiplying a window function with the original signal based on certain time interval and by applying the fractional Fourier transform on the solution of the product of window function and the original signal. Figure 1 shows the block diagram of the process of STFRFT.

In this Fig. 1, the window function is applied in the noisy signal. We set the time interval for the window function and based on the time interval it would multiply

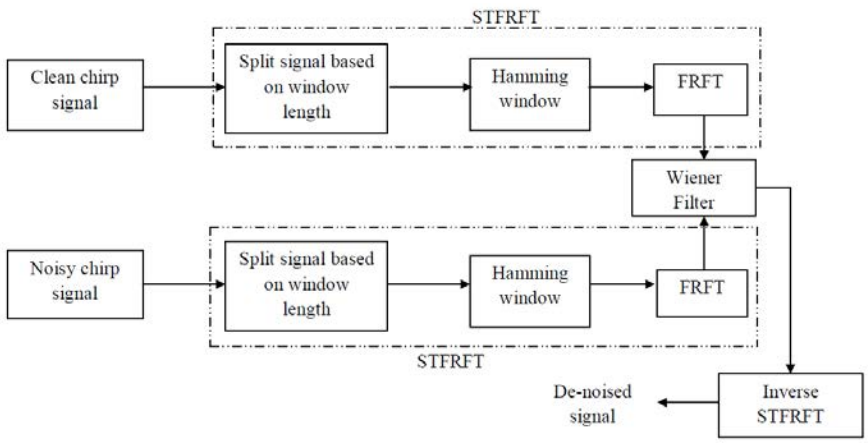

Fig. 2: Sample process of our proposed technique

the noisy signal and give a solution for the time interval we set. Thereafter, the fractional Fourier transform is applied on the solution we obtained after applying the window function.

Proposed W-STFRFT: This section explains our proposed wiener filtering based on short time fractional Fourier transform (STFRFT) for chirp signal enhancement. Figure 2 shows the block diagram of our proposed technique.

Figure 2 explains as follows: initially we set the size of the window we use and split the clean chirp signal and the noisy chirp signal based on the size of the window we set. Thereafter, we apply the hamming window on both the divided clean chirp signal and the noisy chirp signal separately. The hamming window is defined by an eq below:

$$w(n) = 0.54 - 0.46 \times \cos\left(\frac{2\pi n}{m}\right)$$

$$\text{where } 0 \le n \le m$$

Where:

$w(n)$ = Hamming window function
m = Size of the window
n = Varies from 1-m

The solution we obtain after multiplying the hamming window function with the frame of the signal is then multiplied with the Fractional Fourier Transform (FRFT). The process of splitting the signal based on the widow size and multiplying with the window function and with FRFT is called Short Time Fractional Fourier Transform (STFRFT). The solution after apply the STFRFT of both the clean chirp signal and the noisy chirp signal is give as input to the wiener filter and the output we obtain after the wiener filtering process is then apply in inverse short time fractional Fourier transform to get the de-noised signal.

MATERIALS AND METHODS

Signal enhancement process clean chirp signal and noise chirp signals are used to analyzing the potential of the signals. Signal splitting process based on the widow size and multiplying with the window function and with FRFT is called Short Time Fractional Fourier Transform (STFRFT) used with wiener filter. This entire process implemented in MATLAB Software.

Wiener filter: The motto of the wiener filter is to remove the noise from a corrupted signal. Generally, the filters are configured to obtain a desired frequency response. But the approach of the configuration of wiener filter is different. One is assumed to have awareness of the spectral properties of the original signal and the noise and one seeks the linear time invariant filter whose outcome would come similar to the original signal as possible. Here, the fundamental idea behind the wiener filter is to point out the frequencies where the chirp signal is dominant over the noise signal and to make the frequencies less effective where the chirp signal is weak compared to noise. Suppose the noise is additive, the noisy chirp signal would be:

$$y(t) = s(t) + n(t)$$

Where:

$y(t)$ = Noisy chirp signal
$s(t)$ = Noiseless chirp signal
$n(t)$ = Noise signal

A generalized wiener filter can be formulated as follows:

$$H(\varphi) = \left[\frac{P_s(\varphi)}{P_s(\varphi) + \alpha P_n(\varphi)}\right]^{\beta}$$

Where:
$p_s(\varphi)$ = Clean chirp power spectrum
$p_n(\varphi)$ = Noise power spectrum
α = Noise suppression factor
β = Power of the filter

This filter can regulate the amplitude at eachfrequency and perpetuate the original phase. Therefore, the estimation of short time fractional Fourier transform of clean chirp signal is:

$$S(\phi) = H(\phi)\,Y(\phi)$$

$$s(t) = F^{-1}\left[S(\phi)\right]$$

Where:
$S(\varphi)$ = Estimate of STFRFT of clean chirp signal
$y(\varphi)$ = STFRFT of noisy chirp signal

The STFRFT is formed from the fractional Fourier transform (FRFT) and the FRFT is shown by Equation below:

$$X_\alpha(v) = F^\alpha\left[x(u)\right] == \int_{-\infty}^{\infty} x(u) K_\alpha(u,v)\,du$$

Where:
$x(u)$ = Input signal
$ka(u,v)$ = Kernel function

The kernel function is described as follows:

$$K_\alpha(u,v) = \begin{cases} \sqrt{\frac{1-j\cot\alpha}{2\pi}}\, e^{j\left(\frac{u^2+v^2}{2}\cot\alpha - uv\csc\alpha\right)} \\ \text{where } \alpha \neq n\pi \\ \delta(u-v), \text{ where } \alpha = 2n\pi \\ \delta(u+v), \text{ where } \alpha = (2n \pm 1)\pi \end{cases}$$

In the above Equation the term $\delta(u)$ denotes the unit impulse function. The STFRFT is an algorithm that combines the privileges of both Short Time Fourier Transform (STFT) and fractional Fourier transform (FRFT). Initially the signal is multiplied with a window function and the length of the window function is variable and its center is instant of time t.

Subsequently, apply the FRFT on the slice of signal and the frequency spectrum can be obtained in the local fractional domain. The frequency resolution can be enhanced by varying the length of the window. The STFRFT is explained by an equation as:

$$STFRFT(t,f) = P = \sqrt{\frac{1-j\cot\alpha}{2\pi}} \int_{-\infty}^{\infty} x(u) g(u-t)$$

$$e^{j\left(\frac{u^2+v^2}{2}\cot\alpha - uv\csc\alpha\right)} \quad \text{where } \alpha \neq n\pi$$

In this formulation it is considered that we have an estimate of clean chirp power spectrum $p_s(\varphi)$.This estimateis computed from AR smoothed spectrum of noisy chirp signal by $p_y(\varphi)$ only a DC gain modification. The estimation of clean chirp power spectrum uses the original shape of AR smoothed spectrum of noisy chirp signal as:

$$P_s(\phi) = \frac{g_s^2}{g_y^2} P_y(\phi)$$

Where:
g_s = DC gain of noiseless chirp signal
G_Y = DC gain of noisy chirp signal

If we consider that noise and chirp signal are uncorrelated, then the AR smoothed spectrum of noisy chirp signal is:

$$P_y(\phi) = P_s(\phi) + P_n(\phi)$$

$$P_y(\phi) = \frac{g_s^2}{g_y^2} P_y(\phi) + P_n(\phi)$$

If we integrate both sides of the equation over , the equation would be:

$$\int_{-\pi}^{+\pi} P_y(\phi)\,d\phi = \int_{-\pi}^{+\pi} \frac{g_s^2}{g_y^2} P_y(\phi)\,d\phi + \int_{-\pi}^{+\pi} P_n(\phi)\,d\phi$$

Applying Parseval’s relation, the above equation can be simplified as:

$$\frac{g_s^2}{g_y^2} = \begin{cases} \frac{E_y - E_n}{E_y}, & \text{if } E_y > E_n \\ 0, & \text{else} \end{cases}$$

Where:
E_n = Noiseless chirp energy
E_y = Noisy chirp energy

By applying the solution we obtained after the Parseval's relation in the estimate of clean chirp spectrum, the clean chirp spectrum estimate would become:

$$P_s(\phi) = \frac{E_y - E_n}{E_y} P_y(\phi)$$

Using the above eq and by introducing the time dependent noise suppression factor, the wiener filter would be:

$$H(\phi) = \left[\frac{\left(\frac{E_y - E_n}{E_y} \right) P_y(\phi)}{\left(\frac{E_y - E_n}{E_y} \right) P_y(\phi) + \alpha_t P_n(\phi)} \right]^{\beta}$$

After simplification, the above wiener filter can be changed as follows:

$$H(\phi) = \left[\frac{P_y(\phi)}{P_y(\phi) + \left(\frac{E_y}{E_y - E_n} \right) \alpha_t P_n(\phi)} \right]^{\beta}$$

One advantageous property is that to make α_t inversely dependent on SNR and allow it to alter from frame to frame. This will guarantee stronger suppression of noise only frames and weaker suppression during chirp segments which are not corrupted as much to begin with. The desired SNR dependence is achieved by assigning α_t value as. The $\alpha 1 = E_n/E_y \alpha$. Then, the $H(\phi)$ expression of becomes:

$$H(\phi) = \left[\frac{P_y(\phi)}{P_y(\phi) + \left(\frac{E_n}{E_y - E_n} \right) \alpha' P_n(\phi)} \right]^{\beta}$$

To gain higher noise suppression, the value of α can be increased and it does not result in fluctuations in the chirp as much as it does in spectral subtraction. The cause for this outcome is that is $H(\varphi)$ always non negative whereas in spectral subtraction the equivalent filter may fluctuate amid negative values and positive values.

Step by step algorithm description: This study shows the step by step procedure of our proposed technique. This section explains as follows: initially we take the clean chirp signal and calculate the amount of noise to be added and after adding the noise with the clean chirp signal, it would become noisy chirp signal. We set the size for the window function and based on the size of the window function we divide the clean chirp signal and the noisy chirp signal. Each part of the divided clean chirp signal and the noisy chirp signal are multiplied with the hamming window separately and we product the solution with fractional Fourier transform. The short time fractional Fourier transform is the process of multiplying the segmented signal with the window function and with the fractional Fourier transform. The solution we obtained after the short time fractional Fourier transform of both clean chirp signal and noisy chirp signal are given as input to the wiener filter and the inverse short time fractional Fourier transform is applied on the signal we obtained from wiener filter to get the de-noised signal. The algorithm of our proposed W-STFRFT is shown in Algorithm. Algorithm of the proposed W-STFRFT.

Algorithm of W-STFRFT:
Take the clean chirp signal s(t)
Calculate the amount of noise to be addedn(t)
Input: noisy chirp signal y(t)
Set the Hamming window length m
Split the clean chirp signals(t) and noisy chirp signaly(t) based on
For each value of n, where n=1to m
Multiply separated clean chirp signal$S_{sp}(t)$ with hamming window
Apply FRFT to get $P_y(\varphi)$
End for
For each value of , where
Multiply separated noisy chirp signal$Y_{sp}(t)$ with hamming window
Apply FRFT to get $P_n(\varphi)$
End for
Apply the values of $P_y(\varphi)$and$P_n(\varphi)$ in wiener filter$H(\varphi)$ Apply inverse STFRFT
Output: de-noised signal

RESULTS AND DISCUSSION

This study expounds the performance of our proposed technique. Here, we compare our proposed Short Time Fractional Fourier Transform (STFRFT) based wiener filter with Short Time Fourier Transform (STFT) based wiener filter, Fast Fourier Transform (FFT) based wiener filter and Fractional Fourier Transform (FRFT) based wiener filter. We used two evaluation metrics which are Signal to Noise Ratio (SNR) and mean Squared Error (MSE) to compare the performance. We analyzed the performance based on single component chirp signal, double component chirp signal and multi component chirp signal.

Evaluation metrics: The study delineates the evaluation metrics we used to compare the performance of our proposed technique with other techniques. The evaluation metrics we used are Signal to Noise Ratio (SNR) and Mean Squared Error (MSE). The delineation is as follows. The SNR is an estimation used in engineering that contrast the level of desired signal to the level of background noise. The signal to noise ratio is defined as the power ratio amid the clean chirp signal and the de-noised signal. It is explained by an equation in decibels:

$$\text{SNR} = 10\log_{10}\left(\frac{s}{dn}\right)$$

Where:

s = Clean chirp signal
dn = De-noised chirp signal

The mean squared error is an evaluator that expresses the difference amid values implied by an estimator and the true values of the quantity being estimated. MSE estimates the average of the squares of the errors. The mean squared error is defined by an equation below:

$$\text{MSE} = \frac{1}{k}\sum_{i=1}^{k}\left(dn_i - s_i\right)$$

Experimental result of chirp signal model 1: The basic idea of our research is to chirp signal enhancement wiener filtering in short-time fractional Fourier domain for chirp signal enhancement. The system displays the time and FRFD frequency information jointly in the short-time fractional Fourier domain (STFRED).

This study shows the experimental outcome of our proposed technique with the techniques we used for comparison based on interference signal, noise signal and noise with interference signal. In our research, we generate the model of a chirp signal which is contaminated with noise and interference (Tao and Wang, 2010). The signal model is represented given as:

$$X(t) = S(t) + W_i(t) + W_n(t)$$

Where:

$s(t)$ = Chirp signal
$w_i(t)$ = Interference
$w_n(t)$ = White Gaussian noise

Here, the FRFD filtering is a pretty good choice because the chirp signal is highly concentrated in its matched order FRFD. But its performance degrades if interference cannot be separated from the signal in any order FRFD. Considering that interference may be separated from the signal in a 2-D domain, we propose a filter in the STFRFD and term it the STFRFD filter. The statistical amplitude distribution of the background noise in the STFRFD can be well approximated with a Rayleigh distribution:

$$f(X) = \frac{X}{\sigma^2}\exp\left(\frac{-\pi^2}{2\sigma^2}\right)$$

$$\text{Mean} = \sqrt{\frac{\pi}{2}}\sigma$$

An example is given to show the performance of the STFRFD filtering. Suppose that the received signal has been dimensionally normalized whose model is given in X(t):

$$X(t) = \sum_{k=1}^{3} W(t - t_k)\exp\left(j\frac{\mu_k}{2}t^2 + j\Omega_k\right)$$

where, $W(t\text{-}t_k)\rightarrow$ Trapezoidal envelope.

$$W(t) = \begin{cases} k_r(t - t_a) + \frac{A_0}{2} \\ t_a - \frac{t_r}{2} \le t \le t_a + \frac{t_r}{2} \\ A_0,\ t_a + \frac{t_r}{2} \le t \le t_a + t_{PW} - \frac{t_f}{2} \\ k_f\ t(t - t_a - t_{PW}) + \frac{A_0}{2}, \\ t_a + t_{PW} - \frac{t_f}{2} \le t_a + t_{PW} + \frac{t_f}{2} \\ 0 \quad \text{otherwise} \end{cases}$$

Where:

A_0 = Amplitude
k_r = Slope of the rising edge
k_f = Slop of the falling edge
t_r = Duration of the rising edge
t_f = Duration of the falling edge
t_a = Time of arrival
t_{pw} = Pulse width of chirp signal

Based on the model X(t) we have generated the signal and that model experimental results is described below. Figure 3-5 describes the result based on the creating model. When we adding the input signal to interference we obtain the experimental results is shown in Fig. 4. Here, STFT based wiener filtering technique, FFT based wiener filtering technique, FRFT based wiener filtering technique are the best among the existing techniques for chirp signal enhancement. Therefore, we have chosen to compare the performance of our proposed algorithm against that of these ones and Fig. 5 shows the experimental output we obtained for STFT based wiener filtering technique, FFT based wiener filtering technique, FRFT based wiener filtering technique and our proposed STFRFT based wiener filtering technique based on double component chirp signal. We add the noise with the input signal we obtain the result is shown in Fig. 5.

Figure 6 and 7 shows the performance of our proposed approach using various measures. The methods proposed STFT, FET and FRFT are the best known among existing schemes for signal enhancement. Furthermore, they characterize local details of the signal based on frequency, length and information

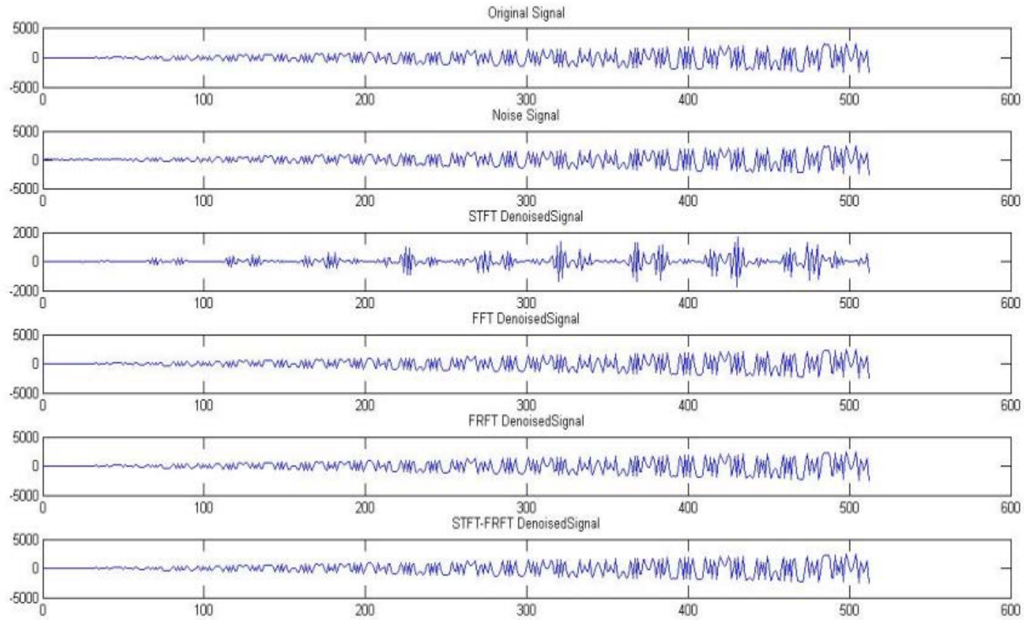

Fig. 3: Experimental output based on adding signal interference

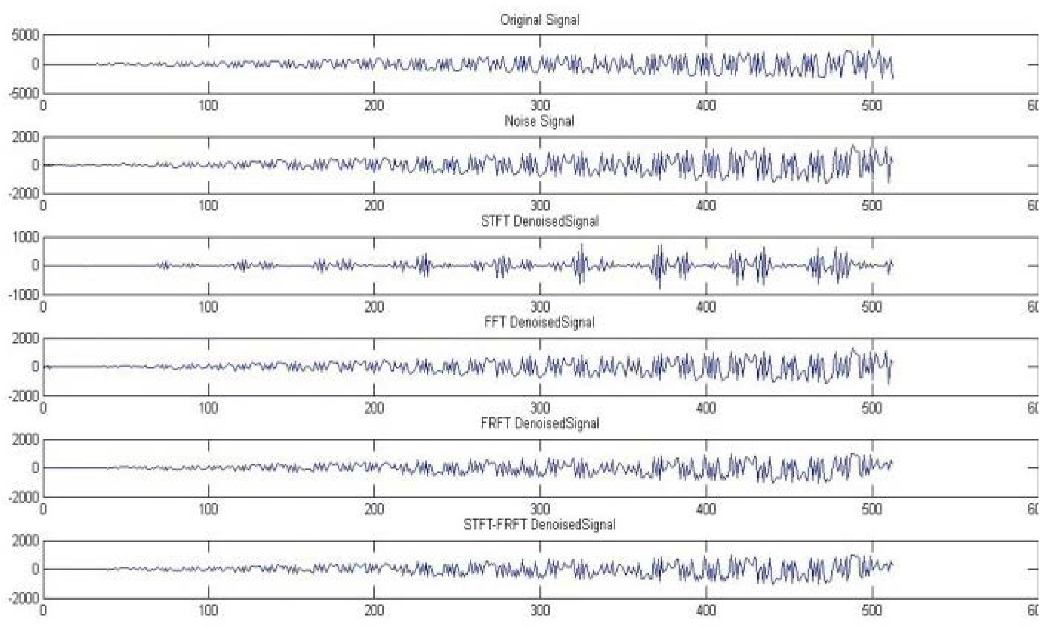

Fig. 4: Experimental output based on signal contaminated by interference

representation. Therefore, we have chosen to compare the performance of our proposed algorithm against that of these ones. From the results shown in Fig. 6 and 7, one can observe that STFT and our proposed one yield the best performances, followed by FFT based approach and FRFT. This is because that these methods very well describe the features of the images. Our method is slightly better than STFT based signal enhancement. In Fig. 6, we obtain the maximum SNR of 30.79 db which is high compare to the other approaches. When adding noise with the signal also we obtain the maximum SNR value. In Fig. 7, shows the performance evaluation of proposed

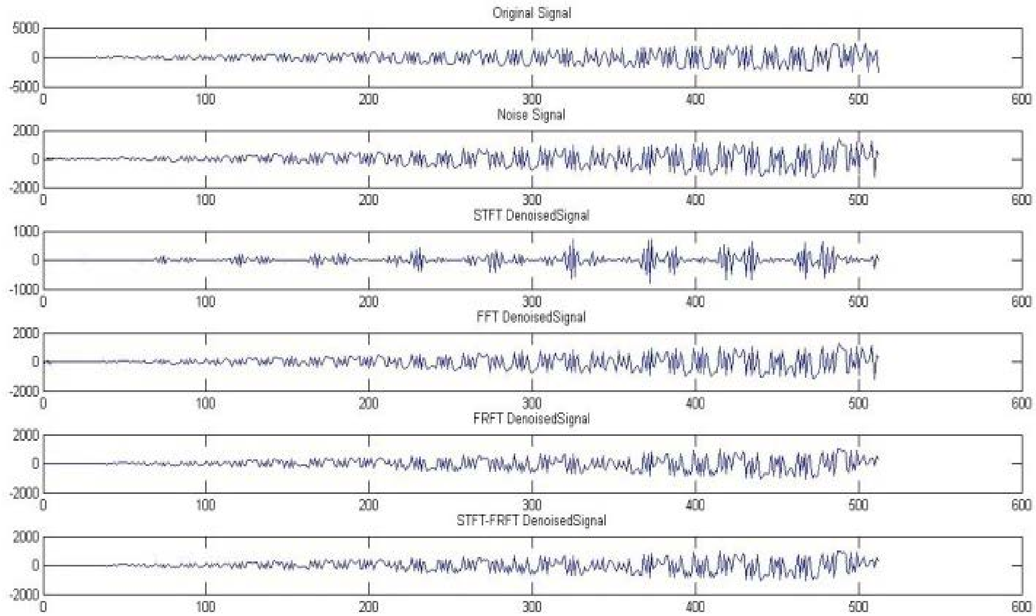

Fig. 5: Experimental output based on signal contaminated by i noise

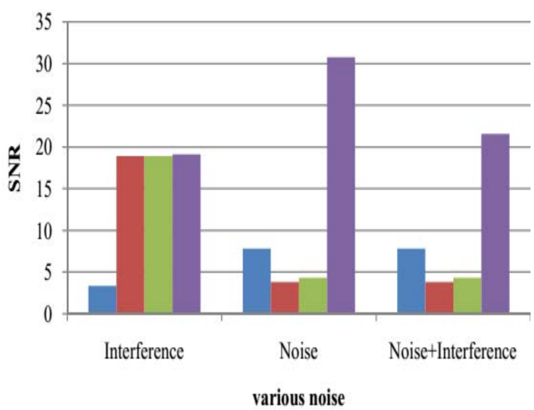

Fig. 6: Performance evaluation of proposed approach using SNR

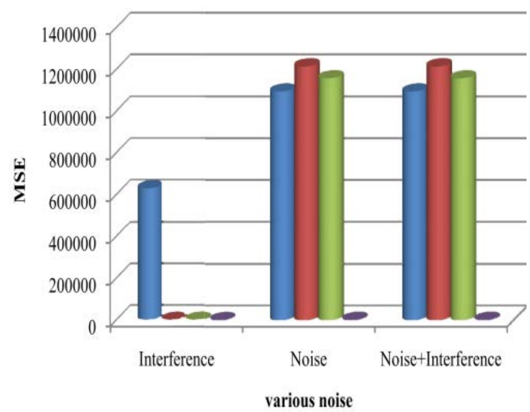

Fig. 7: Performance evaluation of proposed approach using MSE

approach using MSE. Here, in our proposed approach we obtain the mini minimum mean square error of 0.205994. From the above two graph we clearly understand our proposed approach is better compare to all the other existing approach.

Experimental result of chirp signal model 2: In this section we explain the experimental result based on chirp signal. Here we used three types of chirp signal such as Single component chirp signal, double component chirp signal and Multi component chirp signal. The experimental output following Fig. 8.

The experimental output we obtained for STFT based wiener filtering technique, FFT based wiener filtering technique, FRFT based wiener filtering technique and our proposed STFRFT based wiener filtering technique based on single component chirp signal is shown in Fig. 8 and 9 shows the experimental output we obtained for STFT based wiener filtering technique, FFT based wiener filtering technique, FRFT based wiener filtering technique and our proposed STFRFT based wiener filtering technique based on double component chirp signal. The experimental output we obtained for wiener filtering based on STFT, FFT, FRFT and STFRFT is shown in Fig. 10 based on multi component chirp signal.

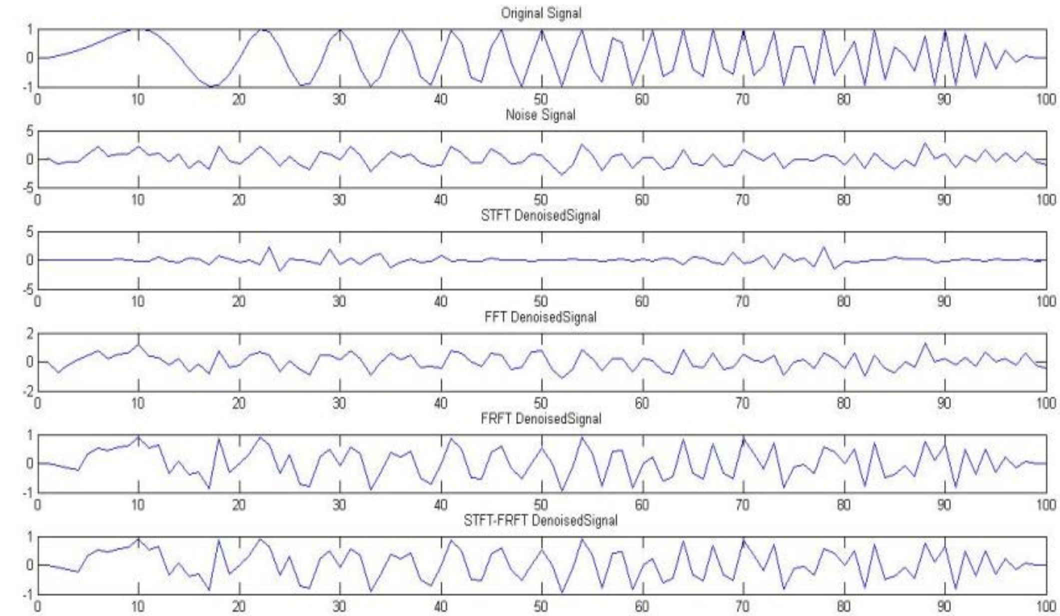

Fig. 8: Experimental output based on single contaminated by interference with noise

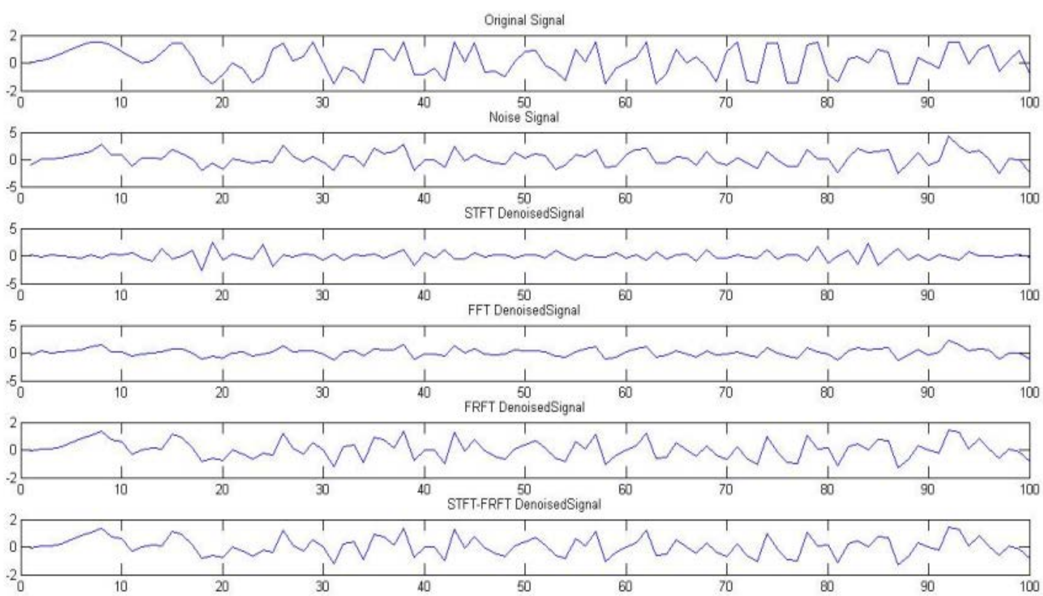

Fig. 9: Experimental output based on double component chirp signal

Single component chirp signal: This part contains the analysis based on the single component chirp signal in terms of signal to noise ratio and mean squared error. Figure 11 shows the signal to noise ratio for our proposed technique and the techniques we used for comparison using single component chirp signal.

In this Fig. 11, we showed the signal to noise ratio for STFRFT based wiener filter, STFT based wiener filter, FFT based wiener filter and FRFT based wiener filter. Here, the SNR is evaluated by varying the noise level. We set the noise levels as 0.6, 0.7, 0.8, 0.9 and 1. The average signal to noise ratio we obtained for STFT based wiener filter is approximately 1.7 db and the average signal to noise ratio

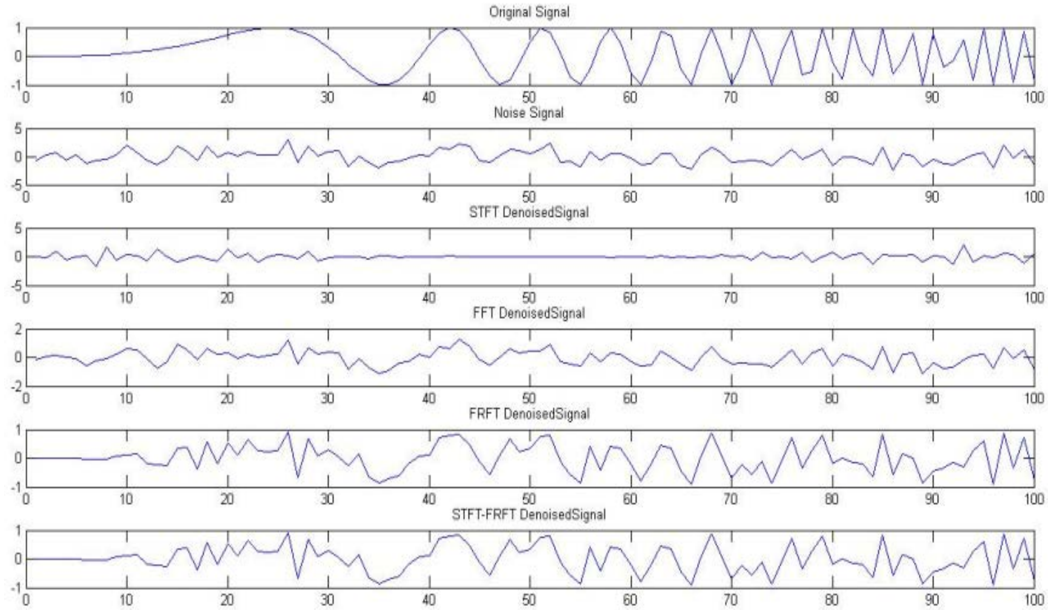

Fig. 10: Experimental output based on multi component chirp signal

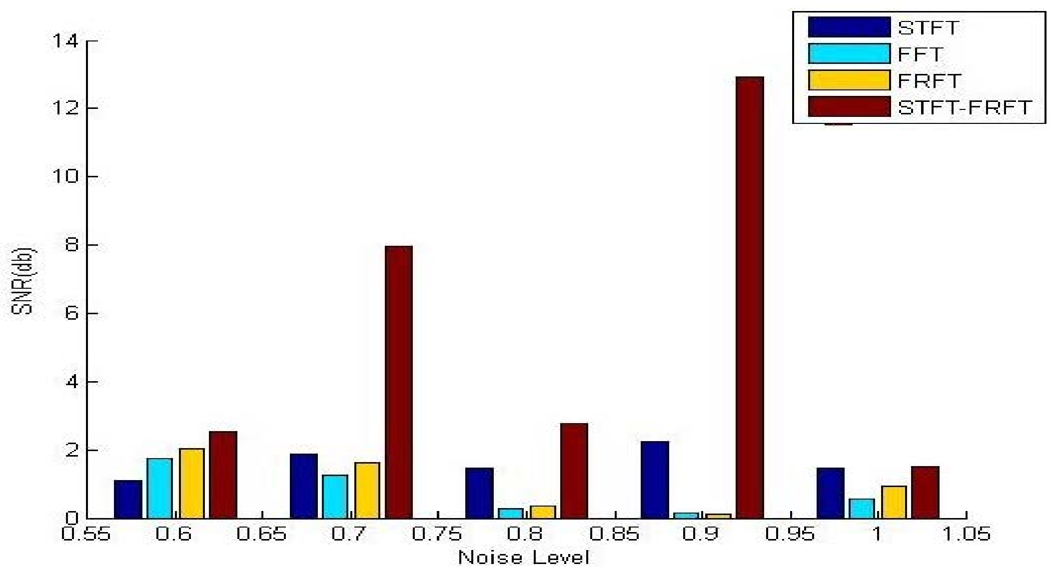

Fig. 11: SNR using single component chirp signal

we obtained for FFT based wiener filter is approximately 0.76 db and the average signal to noise ratio for FRFT based wiener filter is approximately 1.06 db and the average signal to noise ratio for our proposed STFRFT based wiener filter is approximately 5.6 db. Comparing these four techniques, our proposed STFRFT based wiener filter obtained high signal to noise ratio.

Figure 12 shows the mean squared error for our proposed technique and the techniques we used for comparison using single component chirp signal. The MSE is calculated for different noise levels and the noise levels we set are same as that we set for calculating SNR. The average mean squared error we obtained for STFT based wiener filter is approximately 0.812 and the average

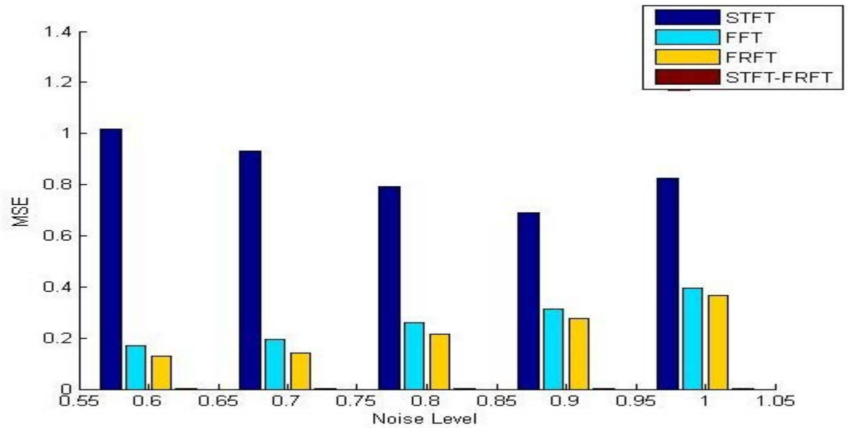

Fig. 12: MSE using single component chirp signal

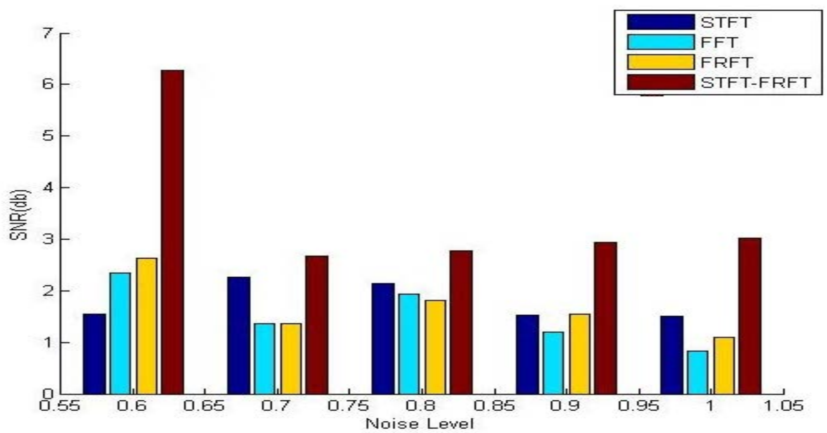

Fig. 13: SNR using double component chirp signal

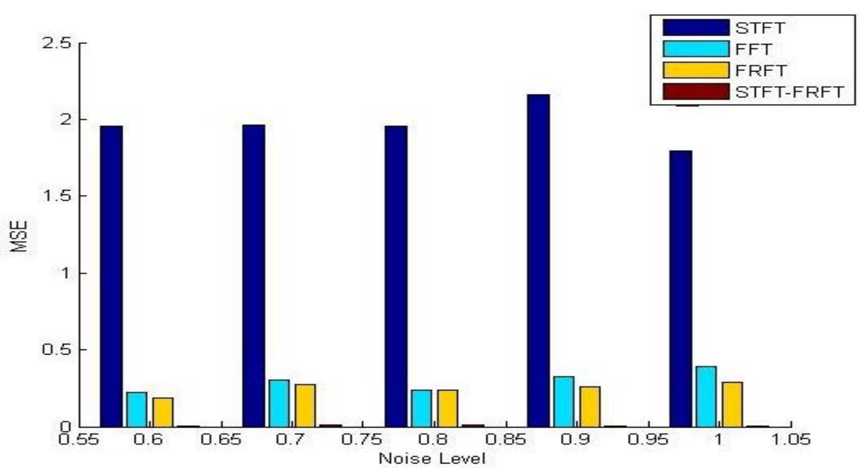

Fig. 14: MSE using double component chirp signal

Multi component chirp signal: This study shows the analysis based on multi component chirp signal in terms of signal to noise ratio and mean squared error for our proposed technique and the techniques we used for

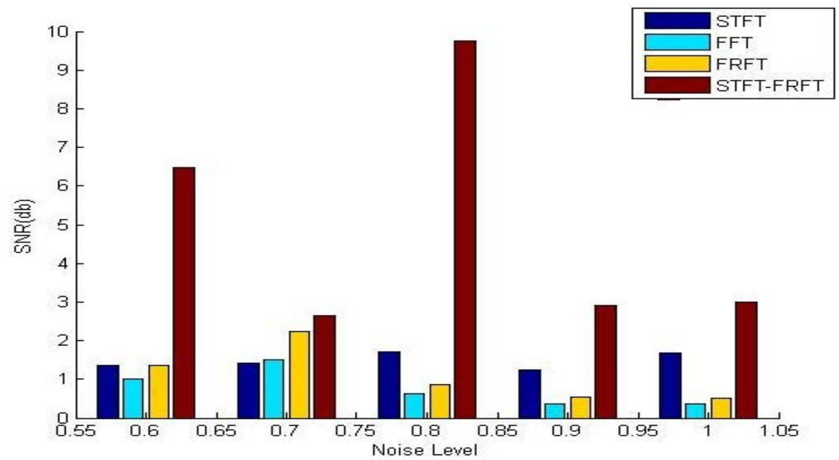

Fig. 15: SNR using multi component chirp signal

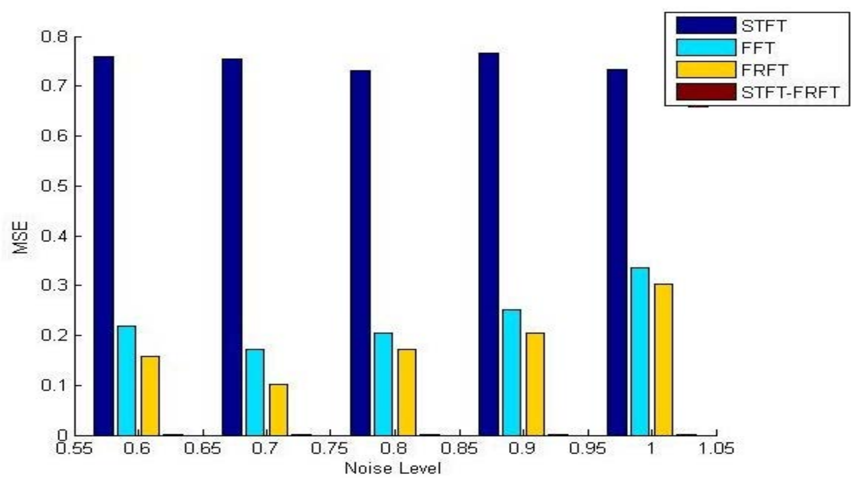

Fig. 16: MSE using multi component chirp signal

comparison. Figure 15 shows the signal to noise ratio for multi component chirp signal.

The signal to noise ratio values for all the four techniques is shown in Fig. 16. In this Fig. 16, the average signal to noise ratio obtained for wiener filtering based on STFT, FFT, FRFT and STFRFT are approximately 1.55, 0.78, 1.08 and 4.98, respectively. Among those four techniques, our proposed STFRFT based wiener filtering technique obtained high signal to noise ratio. Figure 8 shows the mean squared error using multi component chirp signal.

In this Fig. 16 the average mean squared error obtained for STFT based wiener filtering technique is approximately 0.75 and the average mean squared error obtained for FFT based wiener filtering technique is approximately 0.236 and the average mean squared error obtained for FRFT based wiener filtering technique is approximately 0.188 and the mean squared error for our proposed STFRFT based wiener filtering is approximately 0.01. Among those four techniques, our proposed STFRFT based wiener filtering technique obtained less mean squared error.

CONCLUSION

In this study researcher have proposed a technique named wiener filtering based on Short Time Fractional Fourier Transform (STFRFT). The STFRFT is based on window function based Fractional Fourier Transform. The size of the window is set by us and based on the window size we divided the clean chirp signal and the noisy chirp signal and applied the window function and thereafter we applied the Fractional Fourier Transform and given to the wiener filter and the outcome of the wiener filter is applied by the inverse STFRFT to get the de-noised signal. We implemented our proposed technique in MATLAB and we compared our proposed STFRFT based wiener filtering technique with STFT based wiener filtering technique, FFT based wiener filtering technique and FRFT based wiener filtering technique in terms of SNR and MSE and we demonstrated that our technique is better than other techniques we used for comparison.

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