INTERFERENCE CONCELLATION FOR ADAPTIVE LINER TIME VARIANT FILTER BY USING FRFT

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ABSTRACT

The time-frequency (TF) filter design of non-stationary signals such as chirp signals are studied in the fractional domain. The designed TF/ Adaptive linear time variant (ALTV) filter is implemented as a non-overlapping Fractional Fourier transform (FRFT). Using FRFT the adaptive filter is converted into mathematical form to be included in different methods for detecting the presence of a multi component linear FM interference. The present method involved in correctly estimating the orientation of the time-frequency plane using portions of spectral estimation of signal parameter via rotational invariant technique. The paper introduces a novel algorithm to excise single and multi-component chirp-like interferences for CDMA communication. The proposed method implies the interference with high interference power condition could be extended to any kind of chirp interference with some modification can be extended to examples with different signal to noise ratio (SNR) to achieve same result.

KEYWORD:-*Fractional Fourier transform (FRFT), Adaptive linear time variant (ALTV), Minimum Mean-Square-Error (MMSE), Signal To Noise Ratio(SNR), Linear Frequency Modulated (LFM)*

I.   INTERDICTION

Interference suppression in spread spectrum communication systems is very essential for achieving maximum system performance. Existing interference suppression methods do not perform well for most types of non-stationary signals. First the interference suppression schemes based on orthogonal time-frequency decomposition, wavelets and arbitrary time-frequency signals are considered.

It is very import method of data communication. By superimposing a pseudo random (PN) sequence on each data bit, the data is spread over a larger bandwidth and is less susceptible to interferers while being more secure. At the receiver, the signal is dispread back to its original bandwidth and data is demodulated.

The fractional Fourier transform is a time – frequency distribution and an extension of the classical Fourier transform in the areas of signal processing, especially in signal restoration and noise removal. It is hoped that this implementation and fixed-point error analysis will lead to a better understanding of the issues involved in finite register length implementation of the discrete fractional Fourier transform and will help the signal processing community make better use of the transform.

Time frequency representations [1, 2] are essential when working with non- stationary signals such as ultrasonic or acoustic signals. The most straightforward technique for representing a signal in the time-frequency plane is the short time Fourier transforms (STFT). This technique consists of pre windowing the signal under test x(t) around a particular time t and calculating its Fourier transform./the spectrogram, the plot of short-time spectra over a time-frequency plane, is commonly used for the analysis of non-stationary signals in the time frequency domain [3]. Nevertheless, as a result of the Heisenberg-Gabor principle [4]. The spectrogram has trade-off between time and frequency resolutions. When using the spectrogram, the results are dependent on the choice of the window length, which limits the number of applications where this can be applied.

If an interferer is removed at the receiver before dispreading, the performance of the DS-SS system can be greatly improved. A number of different methods exit which work well for a certain types of interferers. These include techniques based on adaptive filtering and minimum mean-square-error (MMSE) criteria [5]. Over determined FFT-based method have also been developed for narrowband interference suppression [6].

In case of linear frequency modulated (LFM) or chirp- type signals, as their frequency varies linearly by time, performance rates are limited for both time-domain and most of the transform domains. LFM signals are among the frequently used signals in real life, i.e., in noise control applications they are good models of the noise of mechanical systems with accelerating internal components. Moreover, a sin spired from the biological signals of bats and whales, they are employed in a range of radar applications. They may also be used as jammers to interfere communication systems. A Gaussian enveloped, single-component LFM signal is expressed as

$$x(t) = Ae^{\pi\gamma(t-t_0)^2}e^{j\pi[\alpha(t-t_0)^2+2\beta(t-t_0)]} \qquad (1)$$

Where is the chirp rate, t_0 and b are the time and frequency shift switch respect to the time–frequency origin, A and g are the parameters of the envelope. One of the most convenient analysis tools for LFM signals is the fractional Fourier transform (FRFT), which employs chirps as basic functions. FRFT is a generalization of the ordinary Fourier transform with a fractional order Para-meter; unitise mathematically powerful and efficiently computable linear transform. It has been employee din various application areas including time–frequency signal we propose a robust adaptive fractional Fourier domain filtering scheme in the presence of LFM- type noise. As the instantaneous frequency (IF) of LFM signals may show rapid variations in time, adaptation to a chirp signal is much more difficult compared to a sinusoidal signal in an active noise control systems. Moreover, the algorithm essentially handles chirp-type signals at all chirp rates. If the signal components present quadratic LFM or other type of characteristics, then the observation period of the signals can be made narrower so that the signal sections are approximated as chirps. if the noise has dominant low-frequency components, passive acoustic noise control techniques are either inefficient or expensive. In contrast, active noise control (ANC) systems are much more effective to cancel low- frequency noise and various active noise cancelation techniques have been proposed [7–9].

Many methods are proposed for IF estimation in the literature, including polynomial phase-based estimators, LMS or RLS-based adaptive filters, and time–frequency distribution-based estimators with inherent disadvantages [10, 11]. In most of the ANC systems, either adaptive filters or neural network based structures are employed to increase the system performance and robustness [7–10]. Adaptive filters mostly employ least mean squares (LMS) based algorithms and the adaptation is usually realized in time domain [12,13]. On the other hand, Fourier trans- form domain [14] In case of multi-component chirp-type signals, [15] shows that the fractional Fourier domain order corresponding to the transformed signal of minimum bandwidth gives IF estimates in sufficiently long observation periods. In [15], two different IF estimation algorithms are proposed in that optimization scheme. a least mean square (LMS) adaptive filter to remove the unwanted noise. An all LMS algorithm starts with an assumption of weight vector as zero initially and iteration continues till the error is minimized till its optimum level. This takes much more time to compute the optimized coefficients [18].

The computational complexity of adaptive filters is Block processing, a block of samples of the filter and desired output are collected and then processed together to obtain a block of output samples and filter weight is changed for each block. And when we do it in frequency domain   The topic Noise cancellation and many speech enhancements is widely researched and many speech algorithms make use of FFT to make it easier to remove noise embedded in the noisy speech signal. In transform domain it is easy to separate the speech energy and noise energy for example energy of white noise is uniformly spread through the entire spectrum concentrated in certain frequencies

Fraction Fourier Transform (Frft)

This is a generalization of classical Fourier transform. Was introduced a number of years ago in the mathematics literature but appears to have remained largely unknown to the signal processing community. To which it may, however, be potentially useful. The FRFT depends on a parameter α

and can be interpreted as a rotation by an angle α in the time-frequency plane. An FRFT with

$\alpha = \dfrac{\pi}{2}$ Corresponds to the classical Fourier transform, and an FRFT with α =0 corresponds to the identity operator. On the other hand. The angles of successive performed FRFT's simply add up, as do the angles of successive rotations.

II. METHODOLOGY

 We briefly introduce the FRFT and a number of its properties and then present some new result : the interpretation as a rotation in the time-frequency plane. And the FRFT's relationships with time-frequency representations such as the winger distribution .the ambiguity function. Short time Fourier transform and the spectrogram, these relationships have a very simply and natural form and support the FRFT's interpretation as a rotation operator. Examples of FRFT's some simple signals are given .An example of the application of the FRFT is also given. As a result, by fractionally Fourier transforming a chirp signal at the appropriate FRFT order, the minimum time- bandwidth product of the signal among all   fractional Fourier domains, i.e., the minimum rectangular area encapsulating the signal support on the time–frequency plane, is achieved. Using this idea, instead of dealing with a rapidly changing chirp signal sweeping a large frequency band in an ANC system. It is possible to operate on the signals with a narrower band. The projections of the WD domain is related to the FRFT and thus one way of determining the IF of an LFM signal component is to search for the peaks of the FRFT magnitudes computed at various fractional orders. This method makes use of the relationship between the RWT of a signal and its corresponding FRFT [16].
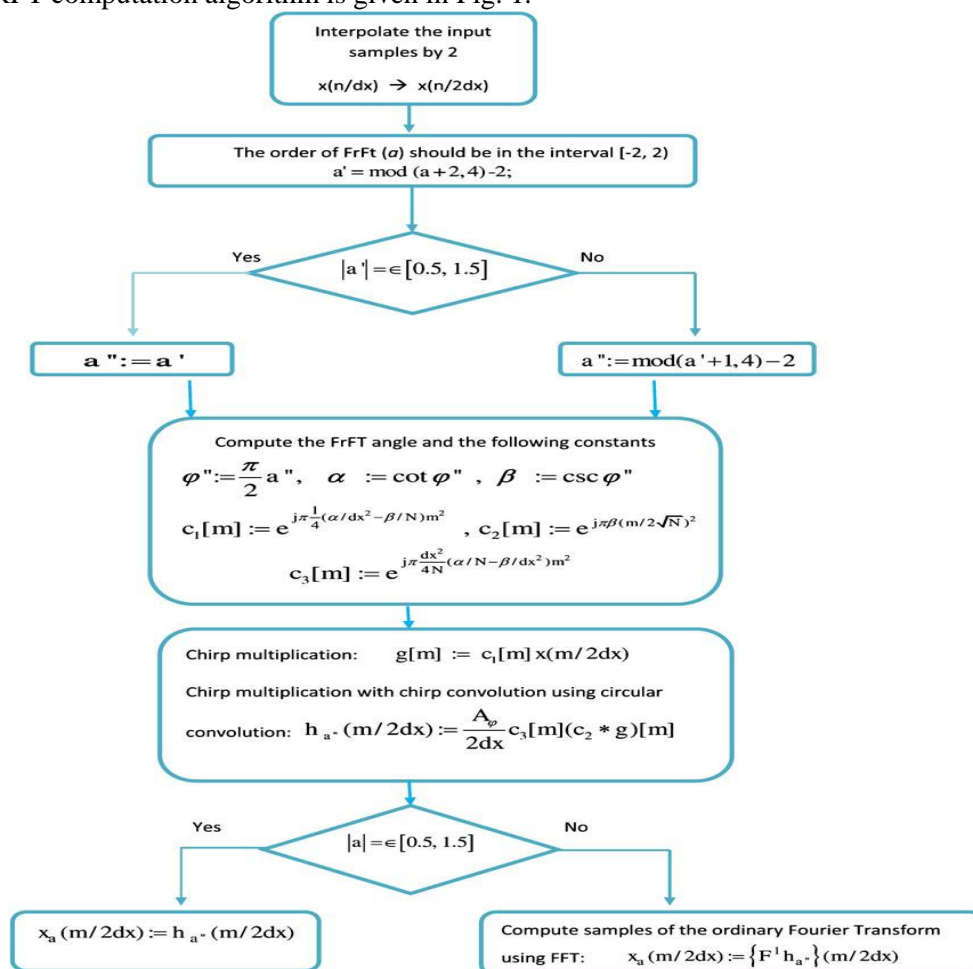The fast FRFT computation algorithm is given in Fig. 1.

Fig. 1. The FRFT computation algorithm

An amplitude - modulated chirp signal and its   time– frequency representation, The appropriate order of the FRFT, which is (1/3) for this case, direction so that the LFM signal is converted

to   an amplitude modulated sinusoidal signal. As a result, by fractionally Fourier transforming a chirp signal at the appropriate FRFT order, the minimum time- bandwidth   product of   the signal among all   fractional Fourier   domains. i.e., the minimum rectangular area encapsulating the signal support on the time–frequency plane is achieved. This definition is first stated as the generalized time-bandwidth product in [17]

The basic scheme of the FRFT- domain in adaptive filter an ANC application. Fig 2

Fig .2 Adaptive fractional Fourier domain filtering

III.   RESULTS AND DISCUSSIONS

The proposed fractional Fourier domain   adaptive filtering scheme of an ANC system is given in Fig. 2. The effect of the environment is summarized by an unknown from equation

Searching the optimum filter tap-weights to minimize the sums of squares of the cumulative error, LMS-based algorithms achieve satisfactory performance with low computational complexity. Error is minimized by decreasing   the filter tap-weights   in the direction of the gradient with a step-size m, recursively

Denote the input vector and error signal at time n. To reduce the effect of the power of the input signal on the system performance m   may be normalized by the power of the signal as in the normalized-LMS (NLMS) algorithm. In ANC systems, filter-x LMS (FX-LMS) algorithm is used to reduce the secondary path effects. In the FX-LMS algorithm, tap-weights w(n) are recursively adapted in the direction of the gradient with a step-size m by using the filtered reference signal   Through the secondary path model S (n),

Emphasize on   the system identification problem of ANC, thus the secondary path is not considered. In this paper, we employ a secondary acoustic path model of an FIR filter and assume its proper estimation as S(n) and we focus on the fractional Fourier domain adaptive filtering scheme.

During the adaptation process, the FRFT order a should be estimated and kept updated. The observation period also depends on the computational power of the processor that operates the algorithm. IF estimation requires   several times of FRFT algorithm giving rise to extra computational cost to the FRFT-domain adaptation. The system set-up and simulations are given.

The adaptation algorithm works best if the noise is formed as consecutive chirps where each of them may have   different chirp rates and different domain it represents in fig 3 . However, if the noise source generates other types of signals, i.e., signals with parabolic or higher order polynomial-type IFs, then the algorithm approximates the phase behaviour as if the ambient noise has linear-piecewise IF and the algorithm operates on the most fitting fractional Fourier domain. Except the FRFT- order estimation at certain time intervals,   adaptation scheme is similar to the Fourier-domain adaptation in practical circuits and system applications
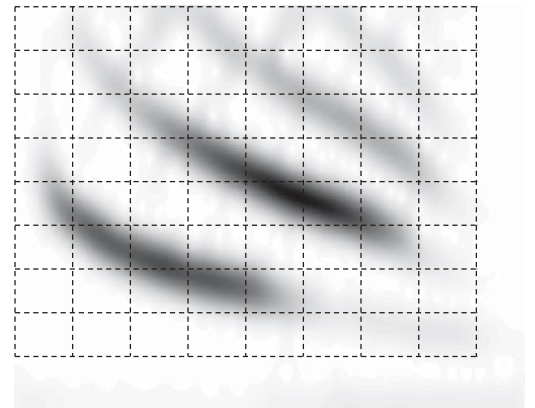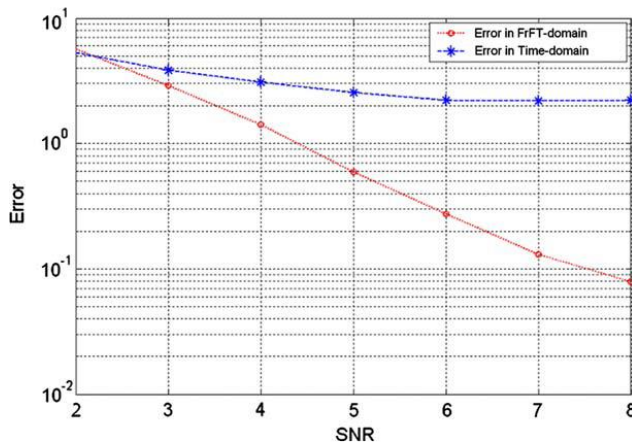
Fig. 3 Error energy in fractional Fourier and time-domain adaptation Schemes with respect to SNR in AWGN.

An LFM noise signal to be modelled by the fractional Fourier domain adaptive filter and its appropriately ordered FRFT and time domain are shown in Fig 3. The real error signals are represented in different windows method also shows in fig 3. The chirp signal is transformed to a sinusoidal signal at the estimated transformation order of a ¼ 1. The input and output signals of the LMS-based adaptive Filter in the fractional Fourier domain in solve in FRFT demine it present in time demine it shows fig 3 with different error variance and real error energy signal represented in fig 4.and the signal apply in FRFT method for estimating error presence than it be solve in two different method like LTV Filter and adaptive modal the result shows graphical representation in fig 5 and fig 6 in different SNR. LTV filter in fig 5 after the mathematical modelling adaptive liner time variant filter it representation of result in fig 6. the figure shows its having different signal to noise ratio by viewing the different suppuration of noise in dB the real error energy signal having the B(bandwidth) is approximately 0.5 %, than it con applying liner time variant filter after suppression of bandwidth goes high nearby 1% , the preference bandwidth adaptive liner variant filter
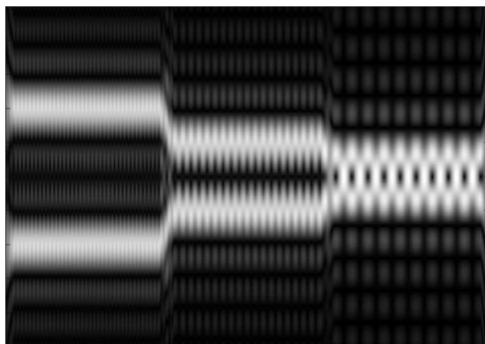

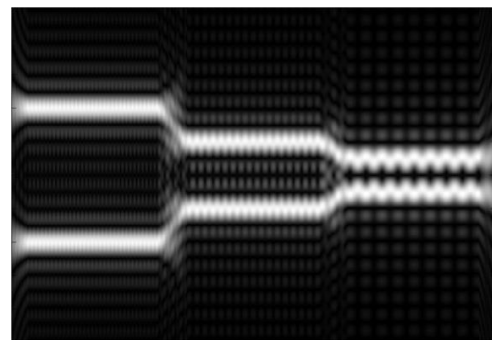**Fig.4** Error energy for real signal B=0.5

Fig.5 Ltv filter using real signal B=1
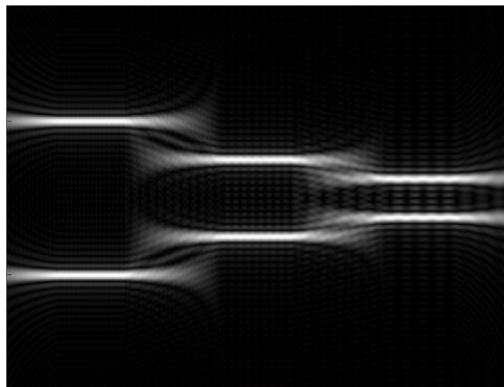

Fig.6 Adaptive ltv filter real signal B=1,5

IV.   CONCLUSIONS

From the results discussed in the paper the following conclusions are can be drawn.

(1) A novel fractional Fourier domain adaptive filtering scheme has been proposed to analyze its performance in an ANC application.

(2) As a system parameter, the estimated IF of the input LFM signal by searching the peak values of the modulus square of the FRFT, is equal to the RWT of the WD.

(3) The fractional Fourier domain adaptive filtering approach avoids the difficulties of adaptation in a rapidly time-varying signal environment by transforming these signals to fractional Fourier domains where the signals become slowly time-varying.

(4) A Comparison of the simulated error signals for synthetic single-component LFM and a real signal with multi-components for both time and fractional Fourier domain adaptive filtering schemes indicates that the total error energy of adaptive filtering in fractional Fourier domain is significantly less than that of the time-domain adaptive filters.

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