Interference Cencellation for Ds-Ss using Rectagular Window LTV Filter and Sliding Discrete Time Fourier Transforms (DSTFT) Techniques

N. Murugendrappa, A. G Ananth

Abstract: ***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. These methods often reduce interference substantially; however the minor changes in interference characteristics such as the center frequency may require changes in the mathematical modeling. The rectangular window methods for fractional Fourier transform with accompanying blind interference excision scheme appears very promising for mitigating time-frequency dominated interference. The present work includes simulations with narrowband interference and comparison of the performance and illustration with different methods. The performance of the rectangular window is evaluated with existing Discrete Short time Fourier Transform algorithm (DSTFT) for filtering with various Jamming-to-Signal Ratios (JSR) starting from 40 dB to 100 dB in steps of 10 dB. Model simulation results with proposed algorithm shows considerable improvement in Signal-to-Noise Ratio (SNR) for the DS-SS signal compared to that of STFT filtering. The results are presented and discussed in the paper.***

Keywords- (SDTFT), (JSR), (SNR), DS-SS.

I. INTRODUCTION

Direct sequence spread spectrum (DS-SS) is a improved technique of data communication. By superimposing a pseudo random (PN) sequence on each data bit, the data spread is over a larger bandwidth and less susceptible to interference while being more secure. At the receiver end, the signal is simply dispread back to its original bandwidth, demodulated and the data is recovered. If interference is removed at the receiver before dispreading, the performance of the DS-SS system can be greatly Improved. Different methods like rectangular window and adaptive methods exist for filtering of the interference and mean square–error (MSR) criteria [1]. The STFT (Fourier transform) based methods have been developed for narrowband interference suppression.

II. FRACTIONA FOURIER TRANSFORM (FRFT)

Various transforms are employed for signal processing to obtain useful information, which is not explicitly available when the signal is in the time domain.

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Most of the real time signals such as speech, biomedical signals, etc., are non-stationary signals.

The Fourier transform (FT), used for most of the signal processing applications, determines the frequency components present in the signal but with zero time resolution. The fractional transforms, such as Fractional Fourier Transform(FRFT) [1–3], Fractional Cosine Transform (FRCT) [4–6] or fractional Sine Transform (FRST) [4,5] describe the energy density or signal intensity simultaneously in the time and frequency domain and have non-zero time frequency resolution in the transform domain. These transforms are used for optical signal processing, time variant filtering, as swept frequency filters, for pattern recognition and signal compression [1–3]. In real time applications, samples of input signal arrive in a sequential manner.

The transform of the signal is obtained by processing sequentially the blocks of N samples. In this procedure, for the computation of transform of N input samples, the system has to wait till the arrival of all N new input samples. Instead, when a new sample arrives the transform can be computed by processing the new block of samples consisting of the newly arrived sample and the N-1 samples of the previous block. This technique is referred to as the sliding technique. Sliding technique reduces the computational complexity and improves the speed. This technique has been successfully employed in computing the discrete Fourier transform (DFT) of the real time signals and is called sliding discrete Fourier transform (SDFT) [7, 8]. The coefficients obtained using the SDFT provide only the frequency information.

Since the fractional transforms have non-zero time frequency resolution, they are better suited for processing the non-stationary real time signals. FRFT is a generalization of ordinary FT with an order parameter ¼ a p/2. The order parameter a represents the angle of rotation of the signal in time–frequency plane. The FRFT is identical to ordinary FT when a ¼ p/2 [1–3]. The discrete fractional Fourier transform (DFRFT) with its kernel expressed in the closed form has the lowest complexity and negligible error in computation [9] satisfy many of the properties. It suited for most of the real time applications due to the simpler and closed form of discrete fractional convolution and correlation [9]. The discrete fractional cosine transforms (DFRCT) and the discrete fractional sine transform (DFRST) are related to DFRFT [4, 5].

However, much attention has not been given to use the sliding technique for fractional transforms.

In the present work the methods of computing the three sliding discrete fractional transforms namely sliding discrete fractional Fourier transform (SDFRFT), sliding discrete fractional cosine transform (SDFRCT) and sliding discrete fractional sine transform (SDFRST) are investigated. Their performances are compared in terms of computational complexity, variance of quantization error and signal-to-noise ratio (SNR). The performances of each of the proposed sliding discrete fractional transforms are compared with that of the SDFT by calculating the SNR for one of the signals available in the sound quality assessment material (SQAM).
Brief mention of the DFRFT; DFRCT and DFRST techniques and the methods of implementation of sliding discrete fractional transforms are presented in the paper.

III. DIRECT SEQUENCE SPREAD SPECTRUM (DS-SS)

A typical DS-SS system is shown in Figure 1. In the system, transmitter generates an SS signal which in turn is transmitted over a communications channel as a binary phase shift keying (BPSK) modulated signal [13]. Additive channel noises as well as jamming signal are also transmitted with the signal. At the receiver, the noise and interference signal is first demodulated. The standard SS receiver correlates the base band SS signal with the synchronized PN sequence, and the resulting signal is processed and input into a threshold detector to estimate the transmitted binary data sequence.

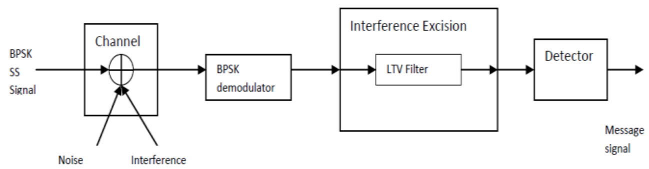

Figure 1: Block diagram of spread spectrum system

Let $b_k = \pm 1$ be the kth message symbol transmitted in a DSSS system such that

$$\mathbf{w}_k = b_k \mathbf{p}_k \qquad (1)$$

where $\mathbf{p}_k = [c_0, \ldots, c_{L-1}]^T$ for {k=1, 2...} is a PN sequence with a chip length L, $c_n = \pm 1$ is the nth chip of the PN sequence, and $\mathbf{w}_k$ is the SS signal. The received signal $\mathbf{r}_k$ at the output of the BPSK demodulator will consist of the SS signal $\mathbf{w}_k$, the additive noise term $\mathbf{n}_k$, and interference $\mathbf{i}_k$ such that

$$\mathbf{r}_k = \mathbf{w}_k + \mathbf{n}_k + \mathbf{i}_k . \qquad (2)$$

We use the notation **r** to refer to the received signal sequence:

zr = {r_1 (0)… r1(L-1),r_2(0),…} (3)

Similarly, we use the notations **w**, **n** , and **i** to refer respectively to the complete SS signal, noise and interference sequences before they are separated into L-element vectors in the form $\mathbf{w}_k$, $\mathbf{n}_k$ and $\mathbf{i}_k$, for k=1,2,…
At the receiver, the received signal **r** is first synchronized and correlated with the same spreading signal **p**. To estimate b_k, we use the PN sequences $\mathbf{p}_k$ to dispread $\mathbf{r}_k$, and integrate the result to generate the test statistics Λ:

$$\Lambda = \langle \mathbf{r}_k, \mathbf{p}_k \rangle = \mathbf{p}_k^T \mathbf{r}_k = \sum_{n=0}^{L-1} p(n) r_k(n) \qquad (4)$$

Using the test statistics Λ, we estimate the message symbols as in bellowed

$$\hat{b}_k = \begin{cases} +1, & if\ \Lambda \geq 0, \\ -1, & if\ \Lambda \geq 0. \end{cases} \qquad (5)$$

The performance of the DSSS model is measured with Bit Error Rate (BER) in this study; we process the received signal after BPSK demodulation. A comparison of estimated message symbol $\{\hat{b}_k\}$ with $\{b_k\}$, and expressing the number of erroneous estimates as a percentage of the total number of message symbols yields the BER.

IV. RESULTS AND DISCUSSIONS

Using Matlab simulations the Fractional Fourier Transform implementation for the best filters two types of transforms have been considered: First the fast approximate fractional Fourier transform algorithms and computation of the fractional Fourier transform is described (Ref H.M.Ozakatas, M.A.Kutay, and G.Bozdfagi.l and .IEEE Trans.Signal Process., 44:2141-2150, 1996).
There are two implementations: (Ref A.M.Kutay and J.O'Nell). For LTV filter, two chrips are considered as input signal. One of the chirp is taken as basis and time frequency graphs of input and output are plotted to show the performance and error variance of the filter with different windows *triangular, rectangular and DFRFT of the basis signal itself as a window.* To evaluate the performance of each window the Error variance of the filter output measured for different chirp rates and the simulation results are shown in Figure 3. The figure clearly demonstrates that the Error variance of all the three windows indicate similar behavior with increase of chirp rates. Further It is seen from the figure that the Rectangular window for higher chirp rates shows maximum error variance.

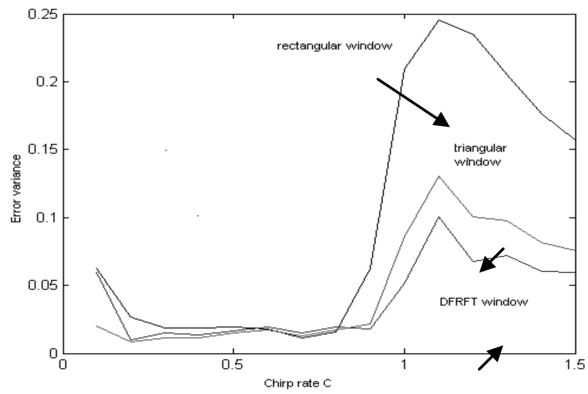

Figure 3: Error variance of the filter output with different windows V/s chirp rate.

Similarly by using simulation techniques for STFT and FRFT filters the BER is determined as a function of SNR. The figure clearly shows that the BER initially increases with SNR value and remains constant with further increase of SNR.

However the STFT filter shows that the BET remains steady initially and increases only for higher values of SNR. But the FRFT filters indicate the increase of BER as a function of SN. The results of the simulation studies clearly indicates that STFT filters are better suited for non stationary signals compared to that of FRFT filters.

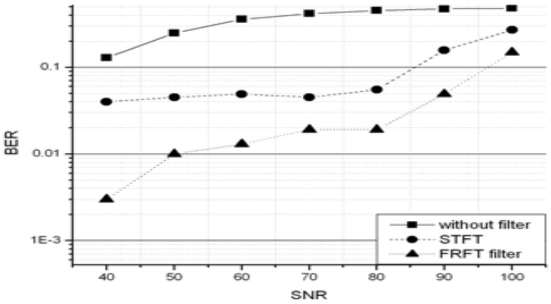

Figure 4: Error variance of with and without filter output V/s STFT and FRFT filter,

The Discrete Fractional Fourier Transform algorithm (DFRFT) described (Refs C.Candan.the discrete fractional Fourier transform , Bilkent Univ., 1998 , S.C.pei ,M.H.Yeh ,and C.C.Tseng) and Digital fractional Fourier transform based on orthogonal projection (Ref IEEE Trans.signal process.,47:1335-1348,1999) algorithms have been considered for further analysis:
The time frequency variation of Direct Sequence Spread Spectrum (DS-SS) signal with Jamming –Signal Ratio (JSR) varying between 40-100 dB has been determined. The signal is filtered using DFRFT filters and the output is shown in Figure 5.

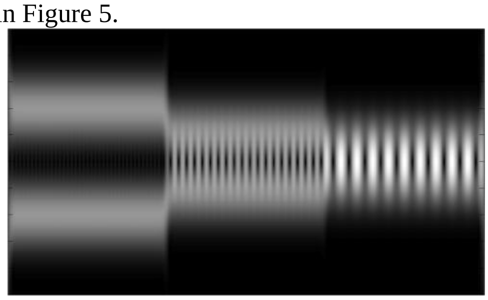

Figure 5.a

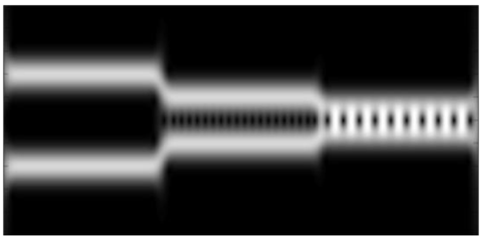

Figure 5.b

Figure 5: Time frequency image of DSSS signal a) with JSR 40dB (b) after using rectangular window LTV DFRFT filtering
The simulation results indicate that the DFRFT filters have significant advantages in designing and implementation compared to the traditional STFT and other LTV filters and shows better performance doe interference cancellation. The sharpness of the filter can be further improved by using the triangular and DFRFT windows The DFRFT window also show better performance in the filtering of overlapping chirp components. This advantage of the filters can be used in LTV filtering of Non-stationary signal application like speech, radar signal processing and in Digital communication .

V. CONCLUSIONS

From the results presented it may be concluded that

(a) The DFRFT filters show better performance compared to traditional STFT and other rectangular windows method
(b) They are better suited for filtering the overlapping chirp components.
(c) The present filtering techniques can be used for filtering Non- stationary signal processing applications.

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