INTERFERENCE CANCELLATION USING ADAPTIVE FILTER FOR FRACTIONAL DOMINE METHODS

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ABSTRACT

A new method forAdaptive Interference Cancellation (AIC) is proposed. The fractional domine concepts are adopted with Fractional Fourier Transformation (FFT) techniques. The Blocked LMS algorithm (BLMS) are used to show that the BER (Bit Error Rate) achieved is better than the direct adaptive filter method in FFT domine and also for other filtering methods. The performance of adoptive filtering method in fractional domine using blocked LMS algorithms shows asignificant improvement in the SNR ~ 0.6 dB for the suppression of interference noise and improvement in BER ~ 0.0428dB for Distributed Spread Spectrum (DSSS) received signal. The BLMS filtering adoptive algorithm developed in this paper for interference cancellation is found to be computationally simple and easy to implement.

KEYWORDS: *Adaptive Interference Cancellation (AIC), Fractional Fourier transformation (FRFT), Signal to Noise Ratio (SNR)*

I. INTRODUCTION

In modern communication systems the radio receiver demands wider frequency range response and fast signal processing capabilities. In such a complex signal environment an adaptivedigital filter can be an alternative to improve the receiver sensitivity when the signal to noise ratio (SNR) is low. Adaptive filters are very effective in processing in the absence of any information about signal environment and signals received. The application of adaptive digital filters with high-speed data processing is currently an important aspect in signal processing field for interference cancellation. The current signal processing involves development of better adaptive filtering methods for Adaptive Interference Cancellation.(AIC). The LMS algorithm is very useful for studying chirp-like interferences in direct sequence spreadspectrum communication. The algorithm consists of two stages namely adaptive signal decomposition and directionalelement detection stage. Based on the Fractional domine the received DSSS signal can be decomposed into its time-frequency (TF) functions using an adaptive signal decomposition algorithm.

Recently Fractional domine based FFT has gained attention for improving methods for AIC tools. These concepts have been forwarded by V.Namias[1] and applied for adaptive signal processing. The idea of FFT representing single variable and the cross domine problems solved by FFT Filtering algorithm was proposed and further improved [2] for data communication techniques. By spreading the data on a pseudo random (PN) sequence, the data will be spread over a larger bandwidth is less susceptible to interference and more secure for direct-Sequence Code Division Multiple Access (DS-CDMA) communication. [3, 4].The DS-spread spectrum increases the bandwidth of the transmitted message making it difficult to track the message. Given the ease in tracking jammers in the time–frequency (TF) domain, TF-based excusers are applied before de-spreading to enhance the robustness to interference [5]. A chirpsignal subjected to FRFT procedures represents rotation in the time-frequency plane. The FRFT's relationship with time-frequency representation is discussed [6].The adaptive filtering techniques using FFT methods have been extended for Narrowband Interference Suppression [7].

For the adaptive filters which employ least mean squares (LMS) based algorithms the adaptation is carried out in time domain [8, 9] and frequency domain using Fourier transforms [10]. In case of multi- component chirp-type signals, [12] the IF estimates for longer periods is provided by the fractional Fourier domain order corresponding to the transformed signal of minimum bandwidth Aleast mean square (LMS) adaptive filter can be designed to remove the unwanted noise and interference present in the signal[11]. The LMS algorithm makes use of the assumption that the weight vector is zero initially. The iteration is continued till the error is minimized to optimum level. This involves more computational time to compute the optimized coefficients.[13]The noise cancellation and speech enhancement techniques have been are widely researched. Normally the speech algorithms make use of FFT so that it easier to remove noise embedded in the speech signal. It is easy to separate the speech energy and noise energy in transform domain. For example energy of white noise is uniformly spread over the entire spectrum concentrated in certain frequencies [14]. In the present paper an attempt has been made to show that the adoptive filters with LMS algorithms can be used for suppressing the interference noise and the computational complexity of adaptive filters can be reduced by Block processing. A block of samples of the desired and filter output are collected and then processed together to obtain a block of output samples. The filter weights are changed for each block in frequency domain.

II. LEAST MEAN SQUARE (LMS) ALGORITHM

The application of adaptive filter isessentially cancellation of interference at receiver end. This adoptive filter can be used to cancel the unknown interference in the primary signal with the help of estimated error and detect error signal to the desired output as shown in Figure 1.

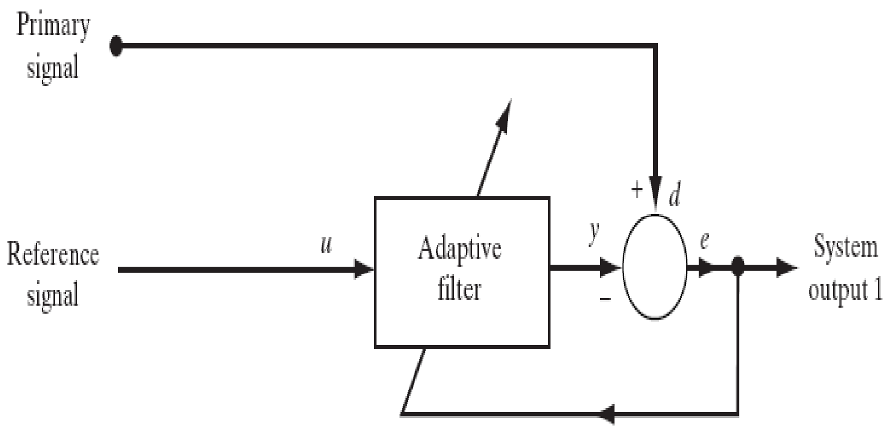

Fig 1 basic block diagram LMS algorithm,

The LMS Algorithm in filtering process will have two basic strategies
- Estimation of error from the difference of output to desired signal
-Estimation of thefilter output by convolution inputsignal and taps

The LMS algorithm will divide the entire input sequences $x(n)$ into the data blocks at with the size of L factor.The weight coefficients are updated block by block by the adoptive filter. The LMS algorithm conducts liner convolution between the weight coefficients and input signals, and linear correlation between the input and error signals.TheFast FourierTransformation isconducted in frequency domain of LMS to achieve fast convolution.

The Figure 2 shows the flow chart of the Blocked LMS (BLMS) algorithm modified for adoptive filtering.

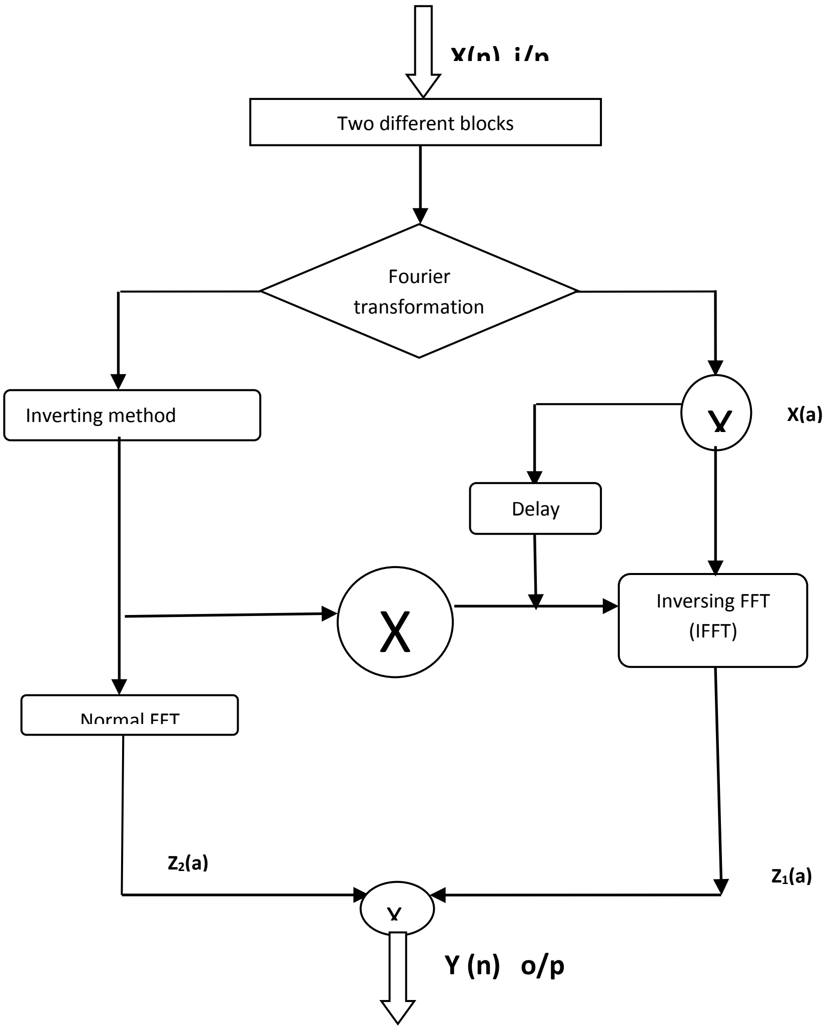

Fig 2 flow chart of BLMS algorithm

III. THE BLMS ALGORITHM

Step 1:- To convert noise into interference signal the input signal must verify into estimated method, the signal denoted as D signals

Step 2:- Every D input signal *x*(*n*) in the time domain get transformed into*X*(*a*) in the frequency domain.make up two cascaded data blocks for conducting *N* point fast Fourier transformation, *X* (*a*) is the input of the adaptive filter. *N* is taken twice as much as *D*– the number of the filter tap.

Sept 3:-*Obtain* the estimated value of interference *Y*(*k*) by Updating the weight coefficientZ_1 (*a*) according to the input signal *X* (*a*) and the error signal $Z_2(a)$

Sept 4:-To get the time domain signals and interference estimated value *Y*(*n*) conduct inverse fast Fourier transformation (IFFT) for the output signal of adaptive filter $Z_2(a)$. The signal*e*(*n*) in which the interference is removed is calculated by the differences between the interfered signal *d*(*n*) and *y*(*n*). The signal *e*(*n*) is transformed by the by Fourier transformation to get frequencydomain value $Z_1(a)$which can serve as a generating filter tap coefficient of the next iteration.

IV. RESULTS AND DISCUSSION

Consider a wide band communication signal as shown in figure 3 (a) .The interference signal is complex in nature. The corresponding estimated input signal with noise and interference is shown in figure 3 (b).The frequency parameter for input signal F_1=0.185 and for reference signal is F_2=0.193.

The N_1 step size is made equal to N_2 step size is set to 0.01 for the simulation. The detector can be used to estimate symbols from one to another close to it.

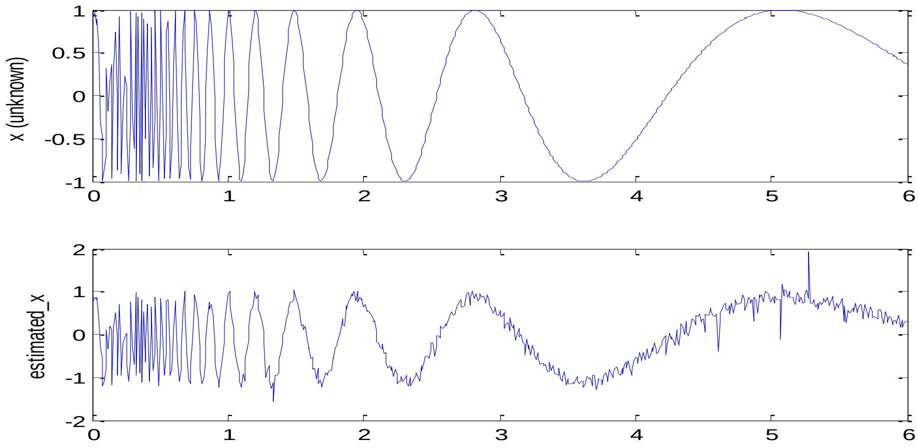

Fig 3. Estimated method of Interference signal

Using identification methods the noise and the interference signals are separated from the original input signal and the noise signals shown in figure 4(a) and interference signals in fig 4 (b) in different dB levels and symbol rate.

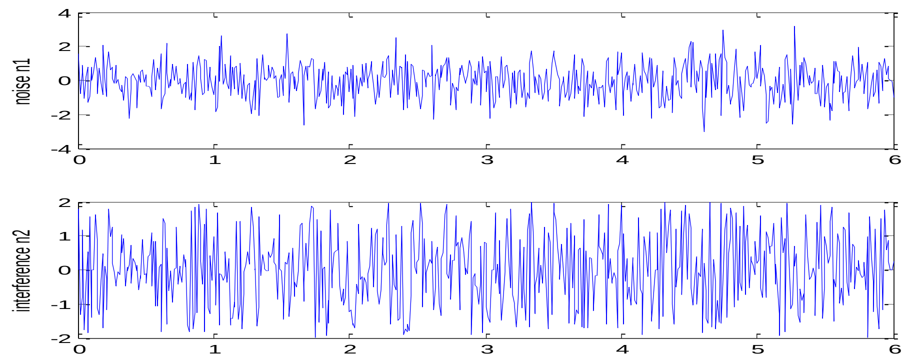

Fig 4. Saperated in noise and interference signal

The BLMS algorithms proposed are applied to the noise signals shown in figure 4 (a) and interference signals shown in figure 4 (b) to derive the interference signal and the interference cancellation from the original signal .The estimated interference signal present in the original signal is shown in figure 5 (a). The interference cancellation achieved by adoptive filtering using blocked LMS algorithms is shown in figure 5 (b).

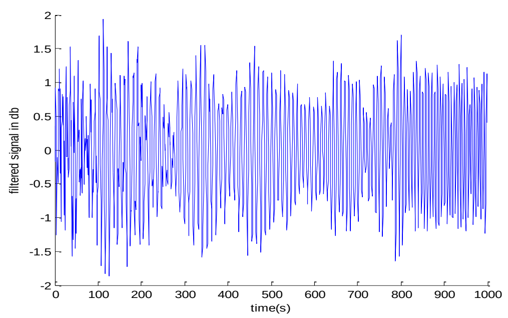

Fig.5. (a) Real time interference signal

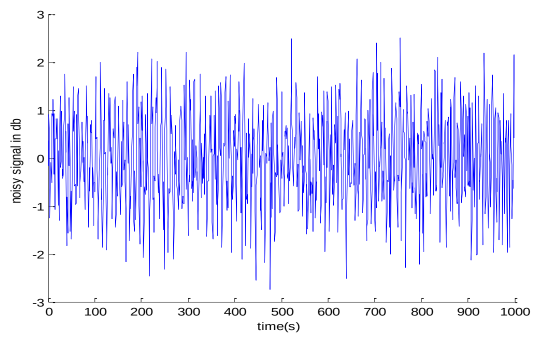

Fig.5.(b) filter output after interference cancellation in 100dB steps

The input signal level is considered at -15.56dB. The corresponding noise at the input level is -45.38 dB and the input SNR is 29.82 dB. The adoptive filter output after applying BLMS algorithm to the input signal remains at same level -15.56 dB and the interference cancelled signal output is at -45.98 dB with the output SNR is 30.42dB. The details of the signal parameters are given in table 1. It is clearly evident from the table that the adoptive filtering techniques used for interference cancellationusing fractional domain methods adopted in BLMS algorithms demonstrates a significant improvement in SNR~ 0.60 dB contributed by suppression of interference signal .

Table 1, SNR improvement in varies conduction in dB

	INPUT(dB)			OUTPUT (dB)			Improvement in SNR (dB)
TR	Signal	Noise	SNR	Signal	Noise	SNR	
1	-15.56	-45.38	29.82	-15.56	-45.98	30.42	0.60

It is also seen from table 1 that the steady-state behavior is reached very quickly for fewer than 100 symbols, the calculated steady-state BER is found to be 0.0426 for SNR~ 0.60 dB

V. CONCLUSIONS

The following conclusions can be drawn from the above presentation

(1) The adoptive filter techniques using fractional domain methods are applicable for the suppression of interference noise in DSSS signals at the receiver end
(2) The Blocked LMS algorithms for the adoptive filters in fractional domain developed in this paper are found to be efficient for the suppression of interference Noise
(3) The improvement in the suppression of interference noise achieved in SNR ~ 0.6 dB and BER~ 0.04 dB forthe received DSSS signals.

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