Adaptive FIR Design Using Levinson Algorithm for Radio Frequency Interference Reduction

Rashmi Bawankar, Dr. Rajesh Mehra, Preeti Singh,
ME Scholar, ECE, Associate Professor, ECE, ME Scholar, ECE,
NITTTR, Chandigarh.

Abstract-*The paper intends an adaptive filter design,based on Linear Prediction Technique. The Levinson-Durbinrecursiontechniquehas beenutilized here to obtain the filter coefficients while it also uses the autocorrelation method to estimate the linear prediction parameters for a segment of a random signal. The radio signals are of the frequency range between 10-100 MHz. The noise occurring in this frequency range is due to RFI or some other parameters like man made or machine made violations.The coefficients for the linear predictor (LP) are robustlycalculated in the MATLAB. The results shows that the Adaptive Levinson-Durbin algorithm used in Linear Prediction technique is more proficient as compared to theother methods appliedforreduction in radio frequency interference,as we have usedhere the FFT method for comparison which is non-adaptive in nature.*

Keywords—*Adaptive FIR, Levinson recursion, Linearprediction, RFI, MATLAB.*

I.INTRODUCTION

Here we will describe a method to abandonthe RFI signals from the original radio signal adulterated with RFI emitters. Also anassortment with other techniques that require a treatment of the signal with other window will be done. It will be shown that the method is both adaptive and efficient and that it can be used as an alternative to methods in the frequency domain[1].The Linear Prediction approach has been used to remove the RFI (Radio FrequencyInterference) lines from a signal violated withnarrow band emitters in the detection of induced radiopulses. The transmitted signals in band limited and frequency selective time dispersive channel distorts, causing interference and the radio frequency signal is contaminated or noised [2]. Most common problem in signal processing is the effect of interference noise in signals, Interference noise masks of the signal and reduces its Intelligibility [13].

Adaptive Filters: Adaptivelinear filters are lineardynamical system with variable or adaptive structure and parameters. They have the property to modify the values of their parameters, during the processing of the input signal,their transfer function, in order to generate signal at the output which is without undesired components, degradation, interference signals.The adaptive FIR filter is a mostwidely used tools in digital signal processing and image processing applications [14]. The goal of the adaptation is to adjust the characteristics of the filter through an interaction with the environment for anticipated values. The operation of adaptive filters is based on the estimation of the statistical properties of the signal, while modifying the value of its parameters in order to minimize a certain principle function [3].Mostly the transversal structure is used in the implementation of the adaptive filters defined bygiven equation [4].

$$e(n) = d(n) - y(n)(1)$$

$$y(n) = \sum_{i=0}^{N-1} wi(n)x(n-i)(2)$$

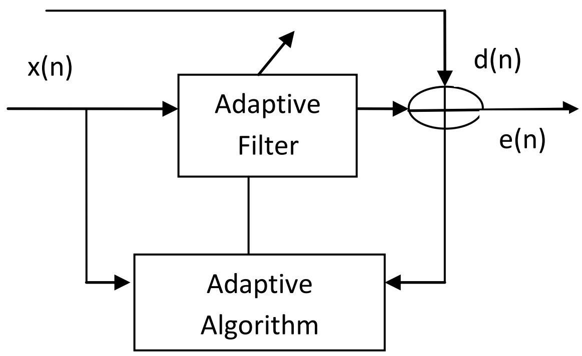

Fig1. Basic concept of adaptive filter [4]

A momentary process in such a system, which is initiated immediately after bringing excitation and which lasts until the output signal assumes a stationary value, i.e. the system enters equilibrium or steady state, has a finite duration. A good property of the FIR filters isthat their phase characteristic iscompletelylinear (the transfer functionG(z) for z = exp(jωt), $-\pi \le \omega T \le +\pi$; where j denotes imaginary zero, is denoted as amplitude-phasefrequency(spectral) characteristic; at that $|G(\exp(j\omega T))|$ is called the amplitude, and $\arg\{G(\exp(j\omega T))\}$isthe phase frequency (spectral) characteristic). Another good property is that they have unconditional stability, and because of that they represent the basis of the systems for adaptive signal processing [2].

II. LINEAR PREDICTION

Adaptive Linear Prediction:Linear prediction is the estimation of the value of the signal that wants to be known. A presumed radio signal denoted by the samples x(i) is adulterated by the RFI. The FIR filter is implemented here to moderate the RFI to get the consequential signal y(i). Thus the FIR filters with coefficients$a_n$are premeditated.
The filter equation can be given by-

$$y(i) = x(i) - \sum_{i=0}^{p} a(n)x(i-D-n) \quad (3)$$

here p is the number of coefficients and D is the delay-line. The delay-line D shows that there is gap between the samples that are used for the prediction and the sample that is to be predicted. This delay-line to allow transient signals to pass through the filter intactis necessary [6].

Adaptive linear prediction performs two operations. if the output is taken from the error signal e(n), is linear prediction. The adaptive filter coefficients predicts, from the statistics of the input signal x(n), what the next input signal will be. The next, if the output is taken from y(n), is a noise filter similar to the adaptive noise cancellation neither the linear prediction output nor the noise cancellation output will converge to Wiener solution. This is true forthe linear prediction output because if the error signal did converge to zero, this would mean that the input signal x(n) is entirely deterministic, in which case we would not need to transmit any information at all [7].

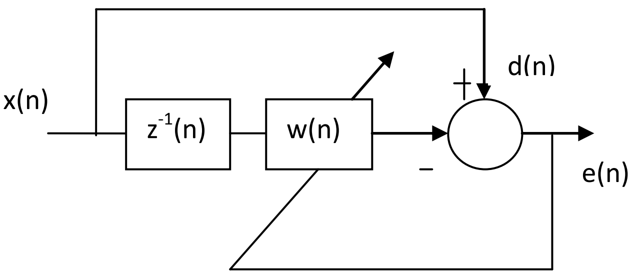

Fig 2.Linear Prediction Configuration[8]

In the case of the noise filtering output, y(n) will converge to the noiseless version of the input signal. [8]

Levinson-DurbinRecursion:The Levinson-Durbinalgorithm usesthe autocorrelation method to estimate the linear prediction parameters for a segment of a random signal.Assuming we have the autocorrelation functionsforthe input process x(n),

$$R_{x,x}(k) = \frac{1}{N-k}\sum_{l=1}^{N} x(l)\, x(l-k) \quad (4)$$

We need to solve the M × M system ofequations to get the desired coefficients (a_k). TheLevinson-Durbin Recursionis a direct recursive method for solving for the coefficients of the prediction filter. It makes particular use of the Toeplitz structure of the matrix R [9].The key to the Levinson-Durbin method of solution that exploits the Toeplitz property of matrix is to proceed recursively, beginning with a predictor of nth order and to increase the order recursively, using lower-order solutions to obtain the solution to the next higher order [10].The steps intricate in the Levinson algorithm are as follows, it uses the prediction filter coefficients of order n-1 to derive the coefficients of the filter of order n.
For n=1 to N

i) Firstly calculate nth order reflection coefficients

$$\Gamma n = -\frac{\Delta n-1}{Pn-1}$$

ii) Then we need to calculate the coefficients of the nth order prediction error filter which is given by

$$\hat{a}_{n,k} = \hat{a}_{n-1,k} + \Gamma_n \hat{a}^*_{n-1,n-k},$$

$$k = 0,1,\ldots\ldots,n$$

where, $\hat{a}_{N,k} = \{\ 1 \quad k=0$
$\quad -\hat{a}_{N,k} \quad k=1,2,\ldots\ldots\ldots,N$

iii) Afterwards the RMS error for the nth order filter as

$$Pn = Pn-1(1 - |\Gamma n|2)$$

iv) Lastly calculating $\Delta_n = r_n^{BT}\, a_{n-1}$.

Here r_n^{BT} = [$R_{XX}(n)$ $R_{xx}(n-1)$ ….. $R_{xx}(1)$]

The Algorithm is then initialized by setting $\hat{a}_0=1$, $P_0=R_{xx}(0)$, $\Delta_0=R_{xx}(1)$.

Performance measurements in Adaptive System:
Convergence rate, Minimum mean square error, Computational complexity, Stability, Robustness, Filter length are the factors for measurements.
i) Convergence rate: The convergence rate determines the rate at which the filter converges to it's resultant state. Usually a faster convergence rate is a desired characteristic of an adaptive system.
ii) Minimum mean square error: The minimum mean square error (MSE) is a metric indicating how well a system can adapt to a given solution. A small minimum MSE is an indication that the adaptive system has accurately modeled, predicted, adapted and/or converged to a solution for the system.

III.DESIGNED SIMULATIONS

From the simulated results obtained by the linear prediction method during application of Levison algorithm to reduce the RFI by several iterations is shown.Fig.3 shows the time varying original sinusoidal signal of radio frequency 10MHz and amplitude -1 to +1, which is taken as input for the filter. In fig.4 we have applied FFT on the original radio frequency signal and using hamming window technique to reduced the RFI from the original signal. Fig.5 is the graph of contaminated radio frequency with white noise showing radio frequency interference (RFI) which is to be reduced. In fig.6 the filtered output signal of Adaptive FIR filter during first iteration of linear prediction using the Levinson Recursion is shown.

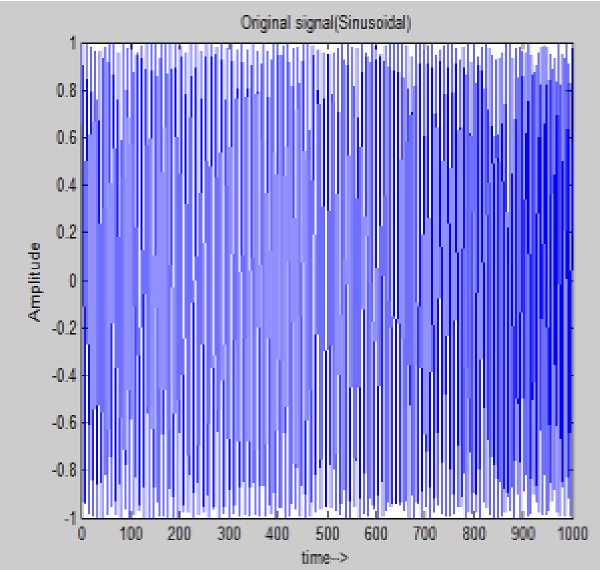

Fig.3 Time varying original sinusoidal signal

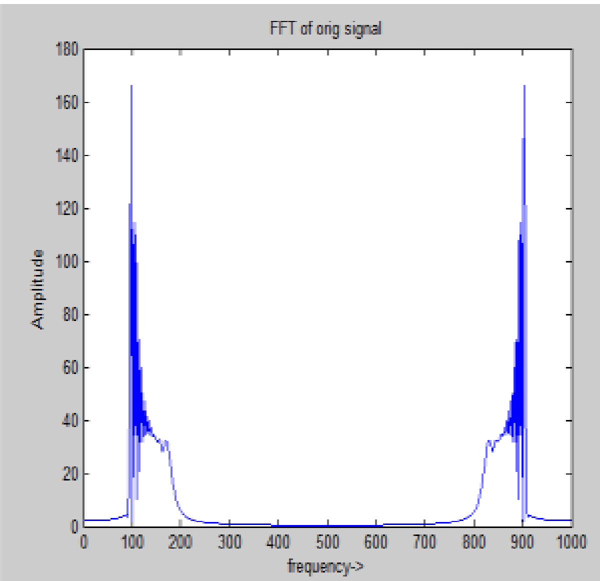

Fig.4 FFT of the original radio frequency signal

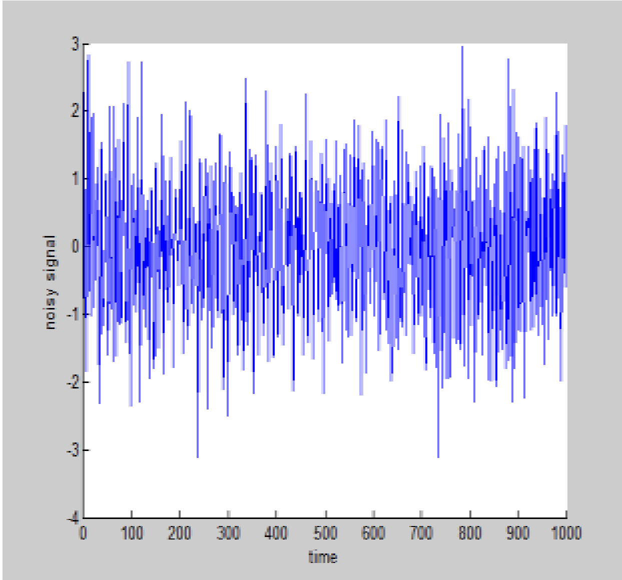

Fig.5 Contaminated radio frequency with white noise

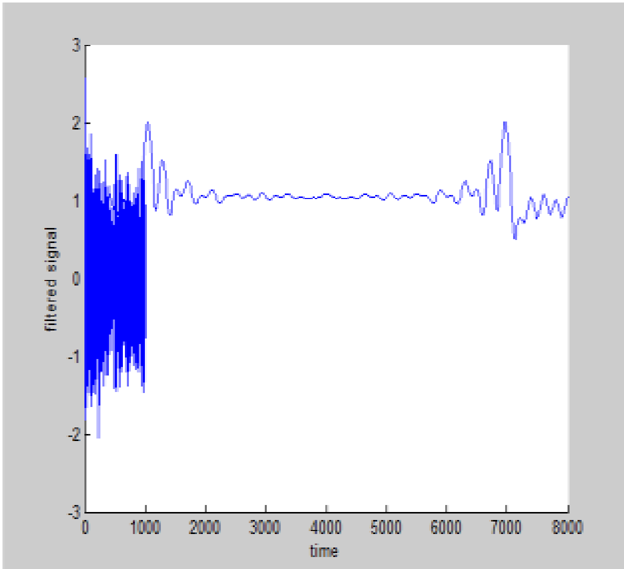

Fig.6 Filtered output signal

IV. RESULT ANALYSIS

Fig.7 shows the filtered response of the signal during increasing iterations of Linear Prediction Technique by Levinson Algorithm. Lastly the fig.8 is showing Minimum mean square error obtained after performing number of iterations using adaptive Levinson algorithm which reduces RFI from radio signal of frequency 10MHz.

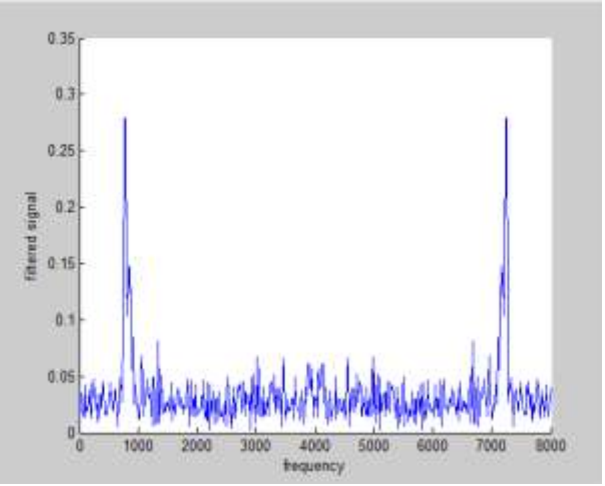

Fig.7 Filtered response of the signal during increasing iterations

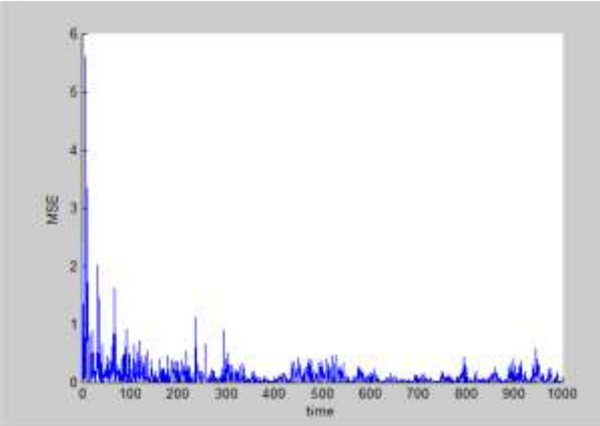

Fig.8 Minimum mean square error after performing number of iterations

No. of Iterations Performed	Values of Coefficients calculated
Iteration 1	0.8991
Iteration 2	0.0710
Iteration 3	0.1007
Iteration 4	0.0412
Iteration 5	-0.0551
Iteration 6	-0.1010
Iteration 7	-0.0411
Iteration 8	0.0469
Iteration 9	0.0716
Iteration 10	0.0050
Iteration 11	-0.0464
Iteration 12	-0.0398
Iteration 13	0.0246
Iteration 14	0.0497
Iteration 15	0.0115
Iteration 16	-0.0549
Iteration 17	-0.0570

V.CONCLUSION

The first method used here is FFT of original signal in time domain which reduces the noise or RFI while the result obtained is not satisfactory as there is no reduction in Radio Frequency Interference. The second techniqueappliedis adaptive Levison-Durbin Algorithm based onlinear prediction. The covariance's are obtained from 1000 iterations and finally the MSE is reduced leading to reduction in RFI from the original contaminated signal. The exploration is showing that the Levison algorithm used in linear prediction error filter is the adaptive algorithm that is more proficient ascompared to the other noise elimination techniques used. Thus it is concluded that linear prediction using adaptive Levison algorithm is more feasible than time to frequencydomain conversion using FFT which is one of the non-adaptive method.

REFERENCES

1) Z. Szadkowski, E.D. Fraenkel, D. Głas, R. Legumina, "An optimization of the FPGA/NIOS adaptive FIR filter using linear prediction to reduce narrow band RFI for the next generation ground-based ultra-high energy cosmic-ray experiment", Nucl. Instr.Meth.,ser. A, vol. 732, pp. 535-539, June 2013.
2) ChanpreetKaur, Rajesh Mehra, "An FPGA implementation efficient equalizer for ISI removal in wireless applications", IEEE Conference on Emerging trends in Robotics and Communication Technologies, pp. 96-99, 2010.
3) B. Kovacˇevic´ et al., "Adaptive Digital Filters," DOI: 10.1007/978-3-642-33561-7_2, Academic Mind Belgrade and Springer-Verlag Berlin Heidelberg, pp. 31-35, 2013.
4) Divya, Preeti Singh, Rajesh Mehra, "Performance Analysis of LMS & NLMS Algorithms for NoiseCancellation," IJSRET, Vol 2, Issue 6, pp.366-369, 2013.
5) A Amrita, R Mehra, "Embedded Design of an Efficient Noise Canceller for Digital Receivers" International Journal of Engineering - Engg Journals Publications, 2011.
6) Z. Szadkowski, E.D. Fraenkel, A. M. van den Berg, FPGA/NIOS implementation of an adaptive fir filter using linear prediction to reduce narrow-band RFI for radio detection of cosmic rays, in: IEEE Real Time Conference, 2012.
7) A. Schmidt, H. Gemmeke, A. Haungs, K-H.Kampert, C. Rühle, Z. Szadkowski, IEEE Transactions on Nuclear Science 58 August 4 2011.
8) Jenkins, W. Kenneth, Hull, Andrew W.,Strait, Jeffrey C., Schnaufer, Bernard A., Li, Xiaohui, "Advanced Concepts in AdaptiveSignal Processing," Kluwer Academic Publishers,1996.
9)MohitGarg, "Linear Prediction Algorithms," IITBombay,pp.4, April 2003.
10) John G. Proakis and Dimitris G. Manolakis, "Digital Signal Processing- Principles, Algorithms, and applications" 3rd edition, pp. 223-225.
11) L. Gupta, Rajesh Mehra, "Modified PSO based adaptive IIR filter design for system identification on FPGA", International Journal of Computer Application, vol.22, 2011.
12) Vikrant Vij, Rajesh Mehra, "FPGA based Kalman filter for wireless sensor networks", International Journal of computer technology, vol.2, 2011.
13) Daphni S, Bamini S, Judith Johnsi J, Thavasumony D, "FPGA Implementation of Adaptive Filtering Algorithm for Noise Cancellation in Speech Signal", International Journal of Engineering Trends and Technology (IJETT), vol.19,2015.

14)Kalyani R. Jambhulkar,"Various Reduction Techniques for Parallel FIR Digital Filter Using Parallel Architecture," International Journal of Engineering Trends and Technology (IJETT), vol.25,2015.

Ms. Rashmi Bawankar: Ms. Rashmi is currently pursuing her ME degree from National Institute of Technical Teachers' Training & Research, Chandigarh, India. She has received her bachelor degree of technology from T.G.P.C.E.T, Nagpur University, Nagpur, India in 2012. She has 2 years of industrial experience on PLC & SCADA Programming and maintenance.

Dr. Rajesh Mehra: Dr. Mehra is currently associated with Electronics and Communication Engineering Department of National Institute of Technical Teachers' Training & Research, Chandigarh, India since 1996. He has received his Doctor of Philosophy in Engineering and Technology from Panjab University, Chandigarh, India in 2015. Dr. Mehra received his Master of Engineering from Panjab University, Chandigarh, India in 2008 and Bachelor of Technology from NIT, Jalandhar, India in 1994. Dr. Mehra has 20 years of academic and industry experience. He has more than 250 papers in his credit which are published in refereed International Journals and Conferences. Dr. Mehra has 55 ME thesis in his credit. He has also authored one book on PLC & SCADA. His research areas are Advanced Digital Signal Processing, VLSI Design, FPGA System Design, Embedded System Design, and Wireless & Mobile Communication. Dr. Mehra is member of IEEE and ISTE.

Ms.Preeti Singh: Ms.Preeti is currently pursuing her ME degree from National Institute of Technical Teachers' Training &Research, Chandigarh, India. She has received her bachelor degree of technology from Harcourt Butler Technological Institute,Kanpur, India in 2007.Ms Preeti has 6 years of teaching and industrial experience.