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# **REDUCTION OF THE EFFECT OF SPOT JAMMING USING KALMAN** METHOD FOR SPECTRAL ESTIMATION

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## ABSTRACT

The paper proposes an adaptive system to reduce the spot jamming effects on radar operation. The proposed system is based on Kalman method for spectral estimation. The system has a short transient response. Consequently, it is suitable for radar waveforms that usually contain only a small number of samples. Simulation results have shown that the proposed system significantly reduces the spot jamming signal and enhances the desired signal. The effects of different factors, such as the filter order and the interference power are evaluated with the help of computer simulation.

#### **KEY WORDS**

Spectral Estimation, Adaptive Whitening filter, Kalman Algorithm

## INTRODUCTION

This paper presents a proposed system to reduce interference signals. Previous work in this area utilized the tapped delay line feedback loop approach [1]. This implementation has the disadvantage of long transient response. The proposed system avoids the long transient response problem by using a spectral estimator. which converges very quickly to the correct estimate of the interference spectrum. The proposed system involves implementing a whitening filter (WF) and a modified matched filter (MMF) as shown in Fig.1. The system parameters are adapted in real time based on the parametric estimate of the interference spectrum. The system description is represented in section II. The desired signal is a continuous wave (CW) signal, while the interference is a band limited signal which represents the case of spot noise jamming. The theoretical background of the Kalaman method of spectral estimation is presented in section III. The simulation results are shown in section IV.



## Fig.1 Optimum receiver for interference environments.

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## SYSTEM AND SIGNALS

The functional form description of the proposed system is shown in Fig.2. The spectral estimator will estimate the interference characteristics, and will adapt the WF parameters to be able to convert this interference into white noise. However the useful signal will be distorted by the whitening filter. Thus, the matched filter response should be adapted to match the distorted useful signal. This adaptation will be performed through convolution, time reversing, and conjugating processes between the useful signal and the WF parameters estimate. This process will produce a new frequency response for every new estimation of the interference signal, and then the interference signal will be reduced at the output of the MMF.



Fig.2 Functional diagram of the chosen system

The adaptive WF is implemented as a controlled FIR filter as shown in Fig.3. The MMF has the same implementation.



Fig.3 The structure of the adaptive WF

To analyze the system shown in Fig.2 consider the detection of a desired signal S(t) in the presence of an interference signal C(t) and a white thermal noise N(t). Generally, the interference signal is a non white random process, while the thermal

noise is a stationary process with a power spectral density  $N_0/2$ . The function of the WF in Fig.2 is to convert the interference signal plus the white noise at its input to a new white random process at its output. Thus the output of the WF is given by:

$$H_1: R_1(t) = S_1(t) + N_1(t)$$
(1)  

$$H_0: R_1(t) = N_1(t)$$
(2)

where  $H_1, H_0$  are the two hypothesis which denote the desired signal present or absent respectively, and  $N_1(t)$  is a white random process.

Over a short period of time the interference is modeled as a stationary autoregressive (AR) process[2]. The parameters of this AR process are estimated, and in turn, these estimates are used to design the whitening filter and the modified matched filter. Specifically, the interference samples are modeled as an N-parameter AR process which is given by [3]:

$$c(k) = \sum_{i=1}^{N} w_i c(k-i) + n(k)$$
(3)

where  $\mathcal{W}_i$  are the coefficient of the AR process, and c(k-i) is the interference sample lagging by "*i*" samples from the present sample.

In this case the power spectrum of the interference process is given by [4'5]:

 $\phi_{c}(f) = \frac{1}{1 - \sum_{i=1}^{N} w_{i} \exp(-j2\pi i fT)}$ (4)

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let  $w_i$  be the estimate of  $w_i$  based on the observation of the interference samples, then the optimum whitening filter frequency response, assuming a large interference-to-noise ratio is given by:

$$H_{w}(f) = 1 - \sum_{i=1}^{N} w_{i}^{\Lambda} \exp(-j2\pi i fT)$$
 (5)

The impulse response of the WF filter "  $h_{WF} = (1, -W)$  " is utilized to modify the matched filter, then the impulse response of the MMF is given by :

$$g_{MMF} = g_{MF} * h_{WF} \tag{6}$$

The" \* " denotes the convolution process.  $g_{MF}$  is the original MF impulse response.  $g_{MMF}$  is the impulse response of the modified MF.

(8)

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## KALMAN METHOD FOR SPECTRAL ESTIMATION

This algorithm adopts a least-square approach rather than the statistical approach used in the LMS algorithm. That is, the Kalman method deals directly with the received data in minimizing the quadratic performance index.

Assume that N samples of the interference signal are available at the input of the WF, then the estimate of the  $k_{th}$  sample is given by

 $\overset{\Lambda}{c}(k) = H_{k-1}^{T} W_{k-1}$ where  $H_{k-1} = (c(k-1), c(k-2), \dots, c(k-N)).$   $W_{k}$  is the weights vector. (7)

 $\phi = \sum_{k=0}^{N} g^{N-k} \varepsilon_k^2$ 

Now, we wish to determine the weight vector  $W_k$  that minimizes the time-average weighted squared error which is given by

Where  $\varepsilon_k$  is the estimation error given by

 $\varepsilon_k = c(k) - \overset{\Lambda}{c}(k)$  $= c(k) - H_{k-1}^T W_{k-1}$ (9)

and g is a weighting factor 0 < g < 1.

The minimization of  $\phi$  with respect to the weights vector  $W_k$  yields the set of linear equations [6]:

$$Rc_k \cdot W_k = D_k \tag{10}$$

where  $Rc_k$  is the signal correlation matrix defined as :

$$Rc_{k} = \sum_{k=1}^{N} g^{N-k} \cdot H_{k-1} \cdot H_{k-1}^{T}$$
(11)

and  $D_k$  is the cross-correlation vector defined as :

$$D_k = \sum_{k=1}^{N} g^{N-k} . c(k) . H_{k-1}$$
(12)

The solution of (9) is :

$$W_k = Rc_k^{-1} . D_k \tag{13}$$

So to compute  $W_k$  from (13), one must solve the set of "N" linear equations for each new sample that is received. This is avoided as follows. First,  $Rc_k$  is computed recursively as :

$$Rc_{k} = g \cdot Rc_{k-1} + H_{k-1} \cdot H_{k-1}^{T}$$
(14)

We call (14) the time-update for  $Rc_k$  Since the inverse of  $Rc_k$  is needed in (13), we use the matrix inversion lemma [4]:

$$Rc_{k}^{-1} = (1 / g) \left[ Rc_{k-1}^{-1} - \frac{Rc_{k-1}^{-1} \cdot H_{k-1} \cdot H_{k-1}^{T} \cdot Rc_{k-1}^{-1}}{g + H_{k-1}^{T} \cdot Rc_{k-1}^{-1} \cdot H_{k-1}} \right]$$
(15)

which may be used to compute  $Rc_k^{-1}$  recursively.

For convenience, define  $V_k = Rc_k^{-1}$ . It is also convenient to define an *N*-dimensional vector, called the Kalman gain vector, as:

$$K_k = \frac{1}{g+L} V_{k-1} \cdot H_{k-1}$$
(16)

Where  $L = H_{k-1}^{T} \cdot V_{k-1} \cdot H_{k-1}$ . With the definition of  $K_k$ , (15) becomes:  $V_k = (1 / g) [V_{k-1} - K_k \cdot H_{k-1}^{T} \cdot V_{k-1}]$ (17)

Also  $D_k$  may be computed recursively as:

$$D_k = g \cdot D_{k-1} + c(k) \cdot H_{k-1}$$
(18)

Substitution of (17) and (18) into (13), we get the weights update equation as:

$$W_{k} = (1 / g) \left[ V_{k-1} - K_{k} \cdot H_{k-1}^{T} \cdot V_{k-1} \right] \left[ g D_{k-1} + c(k) \cdot H_{k-1} \right]$$
$$= W_{k-1} + K_{k} \left[ c(k) - H_{k-1}^{T} \cdot W_{k-1} \right]$$
(19)

#### Simulation results

## **Reduction of band limited interference**

The interference signal is modeled as a band limited signal, centered at the frequency :  $f_c = 0.3(f_s/2)$ , where  $f_s$  is the sampling frequency. This signal is amplitude modulated by a first order Marcove process which is given by

$$a_1(n) = \theta_1 a_1(n-1) + e_1(n)$$
<sup>(20)</sup>

where  $e_1(n)$  is a white noise sequence, and  $\theta_1$  is an arbitrary constant ( $|\theta_1| < 1$ ). Thus the  $n^{th}$  sample of the interference signal is given by

$$c(n) = a_1(n) \exp(j2\pi f_c nT) + e(nT)$$
<sup>(21)</sup>

where T is the sampling interval, and e(nT) is the noise sample. The variance of e(nT) is taken such that the interference to noise ratio INR =20 dB.

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The PSD of the interference signal is shown in Fig.4. The simulated desired signal is assumed to be (CW) signal such that SNR=-5dB. The central frequency of the desired signal is given by:  $f_0 = 0.32 (f_s \ / \ 2)$  . The PSD of the composed received signal is shown in Fig.5. comparing the received signal in case of pure interference as in Fig.4, and that one when the desired signal is added as in Fig.5, one can see that the desired signal is completely hidden by the interference. The convergence curve of the Kalman estimator is shown in Fig.6. It takes only a few samples to reach the steady state. The estimated PSD of the interference signal and the frequency response of the corresponding WF are shown in Fig.7. The figure shows that the WF reduces the interference signal by about "7 dB". Finally, the PSD of the MMF is plotted in Fig.8. Clearly the band limited interference at the output signal is severely reduced, while the desired signal is enhanced by "22dB".



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#### The effect of the estimator order on the performance of the method

The improvement factor (IMF) of the WF is defined as the depth of the notch in the frequency response at the frequency band equipped by the interference. This IMF is utilized as a measure for the quality of the spectral estimation process. A simulation program is constructed to evaluate the dependence of the IMF of the WF on the filter order. The interference signal is assumed to be a narrow band signal as represented in equation (20), it has different values of INR. Fig.9 shows the improvement factor of the WF versus its order. From the figure we see that, the improvement factor reaches its maximum value for order "N=4", and decreases when the order of the filter increases. This result agrees with the fact that, a higher order of the estimator will generate round off noise that accumulates due to the recursive computations[4].

#### The effect of the interference power on the performance of the method

The effect of increasing the interference signal power on the improvement factor of the adaptive WF, controlled by the Kalman algorithm, is evaluated in the case of band limited interference signal. The results are shown in Fig.10 for a filter order of "N=6". The results indicate that the improvement factor increases as the INR increases. During that the convergence time "number of iterations" is plotted versus INR as shown in Fig.11, where one can see that it decreases as the INR increases.



Fig.10 Improvement factor versus the input INR





## CONCLUSION

A method for reducing the effect of spot noise jamming on radar system is introduced. The method is based on converting the spot jamming signal to white noise by an adaptive whitening filter (WF). The matched filter (MF) of the radar receiver is modified to compensate for the distortion caused by the inserted WF. The system has been examined by computer simulation. The results show that, the adaptive WF has a short convergence time. The higher the interference power, the deeper the notch provided by the WF at the interference frequency. The results have also shown that there is an optimum value for the filter order that yields the best performance.

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## REFERENCES

- [1.] "Optimum radar signal processing in clutter", IEEE Trans. Information theory, Vol .1T-14, 1968.
- [2.] Wen Rong Wo "Adaptive AR modeling in white Gaussian noise" IEEE Translation on signal processing. Vol 45, No-5 May 1997.
- [3.] D.E. Bowyer "Adaptive clutter filtering using auto regressive spectral estimation" IEEE Trans. AES. Vol: 15,no-4 July 1979.
- [4.] Gaspare Galati, "Advanced radar techniques and systems", PeterPeregrinus Ltd, 1993.
- [5.] Saeed V. Vaseghi "Advanced signal processing and digital noise reduction", John Willey & Sons LTD. And B. G Teubner, 1996.
- [6.] John G. Proakis "Digital communications", Mc Graw-Hill Book Company, 1983.