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Study on the Nonstationary Jammer Suppression for DSSS Receiver

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Abstract

Using TVAR (Time Varying AR) model and adaptive notch filter is a new method for the non-stationary jammer suppression in DSSS(Direct Sequence Spread Spectrum). The performance of TVAR model for IF (Instantaneous Frequency) estimation will be affected by some factors such as basis functions. The DSSS (Direct Sequence Spread Spectrum) is sensitive to non-stationary jammer, especially the LFM (Frequency Modulation) jammer. Focusing on this problem, the optimal basis function of TVAR model for the IF estimation of the LFM signal is obtained in this paper. Besides the depth and width of notching, the phase properties of notch filter affect the SINR of correlation output to the narrow band jammer suppression in DSSS, in response to the problem the closed solution of correlation output SINR improvement has been derived when a single frequency jammer passes through direct IIR notch filter, and its performance has been compared with those of five coefficient FIR filters. Later, a novel method for LFM jammer suppression based on Fourier basis TVAR model and direct IIR notch filter is proposed. The simulation results show the effectiveness of the proposed method.

Keywords: *DSSS, TVAR model, nonstationary jammer, anti-Jamming*

1. Introduction

The Direct Sequence Spread Spectrum System (DSSS) is sensitive to non-stationary jammer. The problem about non-stationary jammer suppression, especially FM jammer suppression in DSSS communications has been attracting many researchers, many jammer suppression techniques are reported [1-7], and most of the techniques employ Instantaneous Frequency (IF) estimation and Time-Frequency Analysis (TFA). The method of Cohen class non-parameter time-frequency distribution is used to estimate IF [1], [4-7], the disadvantages of this method are large computation burden and slow convergence. Another method for IFestimation is Short Time Fourier Transform (STFT) [8], but the accuracy of IF estimation by using STFT is poor, especially for fast changing non-stationary jammer. High-resolution IF estimation can be got by Time-Varying AR (TVAR) model [3, 9, 10], and the TVAR model coefficients can be estimated by Recursive Least Square (RLS), then the convergence increases and the computation complexity decreases [10]. Shan proposed the FM jammer suppression in DSSS by using TVAR model and five coefficients FIR notch filters, and the time basis function of TVAR model is also used **Error! Reference source not found.**]. In addition to IF estimation, to the non-stationary interference suppression the other key techniques is filter design. The five coefficient FIR filters was used in Reference [1] and [10], for the linear phase properties, and the performance of five coefficient FIR filters was studied in Reference [1]. For IF estimation, besides the time basis function, there are Fourier basis function and Lagrange basis function TVAR model. For No-Linear Frequency Modulation (LFM) jammer suppression, the best basis function of TVAR model is selected through simulation in this paper. The direct IIR notch filter is used widely in engineering, besides the depth and width of notching, the phase properties of notch filter affect the SINR of correlation output to the narrow band jammer suppression, then the closed solution to output SINR improvement of correlation is derived after a narrow band jammer passes through direct IIR notch filter, and its narrow band jammer suppression performance is compared with five coefficients FIR filters. Based on the above analysis, a new FM interference suppression technique in DSSS based on Fourier basis function TVAR model and direct IIR notch filters is proposed in this paper.

The paper is organised as follows. In Section 2, the TVAR model of non-stationary jammer and the IF estimation based TVAR model are introduced. The performance of different

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TVAR model based basis function is compared in Section 3. The closed solution to output SINR improvement of correlation, after a narrow band jammer passes through direct IIR notch filter, is derived, and the narrow band jammer suppression performance is compared with five coefficients FIR filters in Section 4. The performance of the proposed method is simulated in Section 5. In Section 6, the conclusion is given.

2. Non-stationary Jammer Based on Time-frequency Analysis

Based on the above analysis, a new jammer suppression method using Fourier basis function TVAR model and direct IIR notch filters was proposed in this paper. Figure 1 shows the basic principles of jammer suppression time-frequency analysis.

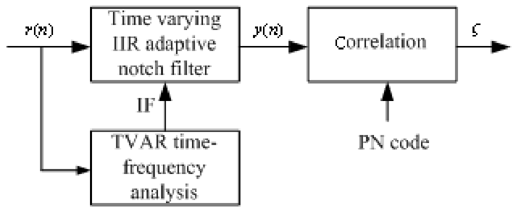


Figure 1. Basic Principles of Jammer Suppression Based on Time-frequency Analysis

Where $r(n)$ is the DSSS signal $y(n)$ is the output of notch filter, ς is the correlation output. Where, the keys are the IF estimation and how to select the notch filter.

3. The IF Estimation of Non-stationary Jammer Based TVAR Model

The TVAR modelling of non-stationary sequence $x(n)$ with order p is **Error! Reference source not found.**]:

$$x(n) = -\sum_{i=1}^{p} a_i(n)x(n-i) + e(n) \tag{1}$$

In which $e(n)$ is stationary white noise with zero mean and variance σ^2, and $\{a_i(n), i = 1, 2, \cdots, p\}$ are TVAR coefficients. The TVAR coefficients are modelled as linear combinations of a set of basis time function $\{u_k(n), k = 1, 2, \cdots, q\}$:

$$a_i(n) = \sum_{k=0}^{q} a_{ik} u_k(n) \tag{2}$$

In which $\{u_k(n), k = 1, 2, \cdots, q\}$ are any basis functions, q is the order of basis, and a_{ik} are polynomial functions of time n. The basis function used in Reference **Error! Reference source not found.**] is:

$$\{u_k(n) = n^k\}_{k=0}^{q} \tag{3}$$

The basis function is the power of time, and it can be called time basis. Besides the time basis function, the Fourier basis function and Lagrange basis function are generally used **Error! Reference source not found.**].

Combining (1) and (2) yields the prediction equation:

$$x(n) = -\sum_{i=1}^{p} \left(\sum_{k=0}^{q} a_{ik} u_k(n)\right) x(n-i) + e(n) \tag{4}$$

The estimation of a_{ik} aims to minimize the total squared prediction error:

$$J(n)=\sum_{n}\left(x(n)+\sum_{i=1}^{p}\sum_{k=0}^{q}a_{ik}u_{k}(n)x(n-i)\right)^{2} \tag{5}$$

Minimizing the above error related to each coefficient leads to the equations:

$$\sum_{i=1}^{p}\sum_{k=0}^{q}a_{ik}c_{kl}(i,j)=-c_{0l}(0,j)\quad 0\le j\le p,1\le l\le q \tag{6}$$

Where, $c_{kl}(i,j)=\sum_{n}u_{k}(n)u_{l}(n)x(n-i)x(n-j)\quad 0\le k,l\le q$

If basis vector is expressed as $[a_{10},\cdots;a_{1q},\cdots;a_{p1},\cdots;a_{pq}]=\theta$, θ can be computed by using RLS **Error! Reference source not found.**], then we can construct the TVAR coefficients $\hat{a}_i(n)$ with Equation (2), by rooting the polynomial formed by TVAR linear prediction filter $1+\sum_{i=1}^{p}\hat{a}_i(n)z^{-i}$ at each instant n, we can get the time-varying poles: $z_i(n), i=1,2,\cdots,p$. The instantaneous angles of certain poles proved estimation of the IF $f(n)$:

$$f(n)=\frac{ang[z_i(n)]}{2\pi}\cdot f_s\qquad |z_i(n)|\approx 1 \tag{7}$$

Where f_s is sampling rate.

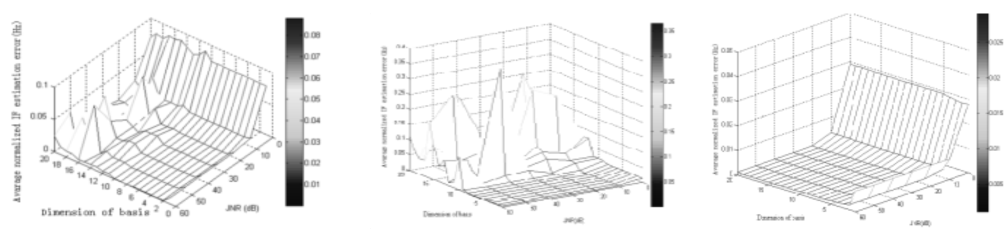
4. The Performance Analysis of Basis

The precision of IF estimation is determined by the order of TVAR, basis function, and the order of basis. For the FM jammer suppression in DSSS, Shan has given the principle on how to select the order of TVAR modelling **Error! Reference source not found.**]: for a signal consisting of M FM components in white noise with moderate Signal-to-Noise Ratio (SNR) or higher, a TVAR signal model can be used, with order p=M for complex exponential FM components and p=2M for real signals. There are still two problems: one is selecting the basis function, and the other is selecting the order of TVAR. The optimal basis function of TVAR model for the Instantaneous Frequency (IF) estimation of the LFM signal was obtained by comparing IF estimation precise and anti-noise performance of several types basis functions, including time basis function, Fourier basis function, and Lagrange basis function through simulation.

The LFM signal used is:

$$J(t)=\sqrt{2P_J}\cos(2\pi(f_0t+f_\Delta t^3)) \tag{8}$$

Where P_J is the power of signal, $f_0=45\mathrm{MHz}$ is the central frequency of LFM, and $f_\Delta=800\mathrm{GHz/ms}$. The sampling rate is 1KHz, and the order of TVAR is 2. Figure 2 shows the average IF estimation error (normalized to sampling rate) for different SNR (from 0 to 60dB), and dimension (from 1 to 20). The number of simulation is 100.



(b) Fourier basis (c) Lagrange basis

(a) Time basis

Figure 2. Average IF Estimation Error (normalized to sampling rate)

Based on the simulation result, the following conclusions were obtained:

(1) Time basis function: the IF estimation error increases because of the over fitting when the dimension of basis is larger than 10. The time basis function is constant when the dimension is 1, then the IF estimation error increases; IF estimation error increases significantly when SNR is below 20dB, it shows that the anti-noise ability is poor; (2) Lagrange basis function: over fitting does not occur, when the dimension is below 20. Lagrange basis function is constant when the dimension is 1, then the IF estimation error increases. IF estimation error increases significantly when SNR is below 10dB, it shows that the anti-noise ability is better than time basis function; (3) Fourier basis function: the IF estimation error increases because of the over fitting, when the dimension of basis is larger than 10 for IF estimation error increases significantly when SNR is below 10dB, this shows that the anti-noise ability of it is better than time basis function the IF estimation precision of it is better than that of others when the dimension is lower.

Such conclusion can be drawn that the Fourier basis function is superior to others for LFM signal IF estimation.

5. Adaptive Filter Design and Performance Analysis

The direct IIR notch filter is used widely in engineering. The transfer function of IIR notch filter is:

$$H(z)=\frac{(z-e^{j\omega_0})(z-e^{-j\omega_0})}{(z-\alpha e^{j\omega_0})(z-\alpha e^{-j\omega_0})}=\frac{1+2\beta z^{-1}+z^{-2}}{1+2\alpha\beta z^{-1}+\alpha^2 z^{-2}} \tag{9}$$

β determines the notch frequency, let ω_0 is notch frequency, then:

$$\omega_0 = arc\cos(-\beta) \tag{10}$$

α is structure factor of notch filter, and $\alpha < 1$.

Besides the depth and width of notching, the phase properties of notch filter affect the SINR of correlation output to the narrow band jammer suppression. It is necessary to derive the closed solution to output SINR improvement of correlation after the single frequency jammer passes through direct IIR notch filter. The correlation output SINR is derived as following.

The DSSS signal is:

$$r(n)=\sqrt{P_s}b(n)c(n)+j(n)+v(n) \tag{11}$$

Where P_s is the power of signal; $b(n)$ is sign , $b(n)\in\{+1,-1\}$; $c(n)\in\{+1,-1\}$ is PN code; $j(n)$ is jammer; $v(n)$ white noise process with zero mean and variance σ^2. $c(n)$, $j(n)$ and $v(n)$ are independent to each other.

The output of notch filter $y(n)$ can be expressed as:

$$y(n)=\sum_{k=0}^{\infty}h(k)r(n-k) \tag{12}$$

Where $\{h(n), n=0,1,\cdots,\infty\}$ is impulse response of notch filter. The SINR of output of correlate after jammer is be moved is **Error! Reference source not found.**]:

$$\mathrm{SINR}_0(\zeta)=\frac{LP_s h^2(0)}{P_s\sum_{k=1}^{\infty}h^2(k)+\sigma_n^2\sum_{k=0}^{\infty}h^2(k)+\frac{1}{2\pi}\int_{-\pi}^{\pi}S_j(\omega)\left|H(\omega)\right|^2 d\omega} \tag{13}$$

Where L is spreading gain, σ_n^2 is the power of noise, $S_j(\omega)$ is the spectrum of jammer. Substituted Equation (10) into Equation (9), then:

$$H(z)=\frac{1-2\cos(\omega)z^{-1}+z^{-2}}{1-2\alpha\cos(\omega)z^{-1}+\alpha^2z^{-2}} \tag{14}$$

According to initial value theorem, then:

$$h(0)=\lim_{z\to\infty}H(z)=\lim_{z\to\infty}\frac{1-2\cos(\omega)z^{-1}+z^{-2}}{1-2\alpha\cos(\omega)z^{-1}+\alpha^2z^{-2}}=1 \tag{15}$$

According to Parseval's Relation:

$$\sum_{k=0}^{\infty}h^2(k)=\frac{1}{2\pi j}\oint_c H(z)H(z^{-1})z^{-1}dz \tag{16}$$

Substituting the transform function Equation (14) into Equation (16), according to Residue theorem, we have:

$$\sum_{k=0}^{\infty}h^2(k)=\frac{1}{\alpha^2}+\frac{1-\alpha}{\alpha^2(\alpha+1)}\left(-2+\frac{(\alpha-1)^2(\alpha+1)^2}{1-2\alpha^2\cos 2\omega_0+\alpha^4}\right) \tag{17}$$

According to Equation (17), the jammer frequency have a certain power disturbances to the frequency of the notch filter impulse response, but disturbances is very small when α is close to 1, if disturbances is neglected, then:

$$\sum_{k=0}^{\infty}h^2(k)=\frac{1}{2\pi j}\oint_c H(z)H(z^{-1})z^{-1}dz=\frac{1}{\alpha^2}-\frac{2(1-\alpha)}{\alpha^2(1+\alpha)}=\frac{3\alpha-1}{\alpha^2(1+\alpha)} \tag{18}$$

According to Equation (15), (17) and (18), then:

$$\mathrm{SINR}_0(\zeta)=\frac{LP_s\alpha^2(1+\alpha)}{[\alpha(3-\alpha-\alpha^2)-1]P_s+(3\alpha-1)\sigma_n^2+\frac{1}{2\pi}\int_{-\pi}^{\pi}S_j(\omega)\left|H(\omega)\right|^2d\omega} \tag{19}$$

In the ideal condition:

$$\frac{1}{2\pi}\int_{-\pi}^{\pi}S_j(\omega)\left|H(\omega)\right|^2d\omega=0 \tag{20}$$

From Equation (19), we have:

$$\mathrm{SINR}_\mathrm{o}(\zeta)=\frac{LP_s\alpha^2(1+\alpha)}{[\alpha(3-\alpha-\alpha^2)-1]P_s+(3\alpha-1)\sigma_n^2} \tag{21}$$

If there is no notch filtering, the SINR is:

$$\mathrm{SINR}_\mathrm{no}(\zeta)=\frac{LP_s}{\rho_j(0)+\sigma_n^2} \tag{22}$$

We define the SINR improvement factor:

$$\gamma=\frac{\mathrm{SINR}_\mathrm{o}(\zeta)}{\mathrm{SINR}_\mathrm{no}(\zeta)}=\frac{[\rho_j(0)+\sigma_n^2]\alpha^2(1+\alpha)}{[\alpha(3-\alpha-\alpha^2)-1]P_s+(3\alpha-1)\sigma_n^2} \tag{23}$$

The numerator and denominator are divided by the power of DSSS signal, then:

$$\gamma = \frac{[SIR^{-1} + SNR^{-1}]\alpha^2(1+\alpha)}{[\alpha(3-\alpha-\alpha^2)-1] + (3\alpha-1)SNR^{-1}} \quad (24)$$

Equation (24) shows that γ is determined by SIR , SNR , and α .

The SINR improvement factor of correlation output of five coefficient FIR notch filter is [1]:

$$\gamma = \frac{2[1+\cos^2(\omega_0)]^2\left(\sigma_n^2 + \rho_j(0)\right)}{\left(2 - \frac{2}{L} + 32\cos^2(\omega_0) - \frac{16}{L}\cos^2(\omega_0)\right)P_s + \left(1 + 16\cos^2(\omega_0) + [1+2\cos^2(\omega_0)]^2\right)\sigma_n^2} \quad (25)$$

Where P_s is the power of DSSS signal; L is spreading gain; ρ_j autocorrelation function of jammer; ω_0 is normalized frequency; σ_n^2 is the power of noise.

The first part of denominator of Equation (25) can be defined as general system's noise. When $\omega_0 = 0$ and processing gain $L \to \infty$, the system's noise of FIR approaches to $17P_s/4$. From Equation (23), when $\alpha \to 1$, the system's noise of IIR approaches 0. We can see that distortion of IIR notch filter is far less than that of FIR notch. Based on comparing Equation (23) and (25) we got the conclusion that the correlation output SINR improvement factor of the five coefficients FIR notch filter to the single frequency jammer suppression is determined by the SNR, SIR, angular frequency and spreading gain. However, The SINR improvement factor of correlation output of the direct IIR notch filter determined only by α .

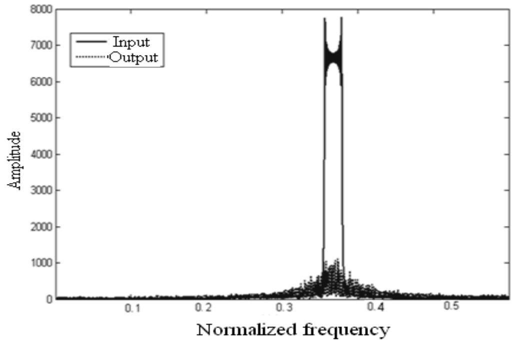
6. LFM Jammer Suppression

The basis vector θ can be computed by using RLS, then we can construct the TVAR coefficients $\hat{a}_i(n)$ with Equation (2), by rooting the polynomial formed by TVAR linear prediction filter $1+\sum_{i=1}^{p}\hat{a}_i(n)z^{-i}$ at each instant n. we can get the time-varying poles: $z_i(n), i = 1, 2, \cdots, p$, and IF $f(n)$ by using Equation (10). The coefficients β is adjusted according to $f(n)$.

To evaluate the performance of the method proposed in this paper, we compare it with the methods using time basis TVAR and five coefficients FIR filter [11], and using WVD and five coefficients FIR filter [1].

We consider a single-user code-on-pulse DSSS communication system with BPSK modulation and spreading gain is L=1023 chips per bit. To focus on the effect of jammer cancellation on correlation output SINR: $SNR = -20dB$ LFM jammer, which chirps linear form 0.34 to 0.37Hz (normalized to unit sampling rate) is:

$$f_j(t) = 0.34 + 0.03t/127, t = 0, 1, \cdots, 127$$



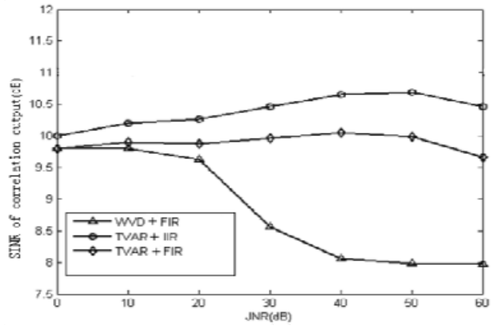


Figure 3. Frequency Spectrum of Input and

Figure 4. The Correlation Output SINR for

Output of IIR Notch Filter Different Method

The used TVAR model is a second-order autoregressive model (p=2), with the time-varying coefficient represented as a Fourier basis (q=3), α of direct IIR notch filter is 0.9.

Figure 3 shows the simulation result that frequency spectrum of the signal before and after the jammer passes through direct IIR notch filter with $JNR = 20dB$, the LFM jammer is suppressed significantly.

The Figure 4 shows the correlation output SINR for different method .In this case, it is observed from figure that the performance of TVAR method of this paper is better than that of the method of Reference [10], and significantly outperforms the method based WVD of Reference [1].

7. Conclusion

The simulation result shows that the Fourier basis function of TVAR model is superior to the time basis function and the Lagrange basis function for LFM signal IF estimation, then for LFM jammer suppression, the best basis functions of TVAR model is selected in this paper. The closed solution of output SINR improvement of correlation is derived after a narrow band jammer passes through direct IIR notch filter, and its narrow band jammer suppression performance is compared with five coefficients FIR filters, the result shows the performance of IIR notch filter is superior to the five coefficients FIR filters. Based on the above, a new FM interference suppression technique in DSSS based on Fourier basis function TVAR model and direct IIR notch filters is proposed in this paper. The performance of the method of this paper is superior to that of the method using STFFT and IIR filters, and time basis TVAR model and five coefficients FIR filters.

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