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A Novel Narrow-Band Interference Rejection Based on Wavelet-Packet Transform

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ABSTRACT: *A novel adaptive wavelet-packet transform for suppressing the interference in Direct Sequence Spread Spectrum (DSSS) communication systems is put forward in this paper. The optimal NBI (narrowband interference) rejection method is proposed using the adaptive wavelet-packet transform instead of sub-band elimination, which can suppress NBI efficiently and reduce the loss of the avail signal. Analysis and simulation results show that the interference suppression method of adaptive wavelet packet decomposition based on the ATF algorithm will obtain better performance than that of conventional sub-band elimination, especially when there are multi tone interferences. And compared with the traditional LMS adaptive algorithm, it can effectively improve the convergence speed of the algorithm.*

Keywords: *DSSS, convergence rate, NBI, adaptive wavelet-packet transform*

I. INTRODUCTION

The narrowband interference suppression technology in DSSS is usually divided into two categories [1]: transversal filter processing technology in time domain and transform domain processing (TDP) structure. Transversal filter processing technology in time domain has thoroughly narrowband interference suppression ability, but its convergence speed is not fast, can only handle stationary interference. TDP narrowband interference suppression based on fast Fourier transform (FFT) is real-time, but the time window is introduced in the transformation, which makes the side lobe effect in frequency domain. So, it is very difficult to remove the interference completely. With the increasing maturity of wavelet transform, some authors use the wavelet transform and the corresponding filter banks to realize the transform domain filtering. Its side lobe is less than the discrete Fourier transform, which can reduce spectrum leakage. Since the wavelet packet transform (WPT) has the ability of multi-resolution and good time-frequency localization, Landry proposed the use of wavelet packet decomposition to locate time varying interference, because the wavelet packet decomposition can maintain the characteristics of the noise energy which can be used to suppress the narrow band interference. The main method is that the input signal with narrowband interference is decomposed into irregular sub-tree, and then put the disturbed sub component to zero in order to remove the most serious sub band. After that, restore the signal, which can achieve better narrowband interference suppression effect. Tazebay introduces a new method of anti-jamming in the domain--adaptive time- frequency interference rejection (ATF) [2]. The proposed algorithm can generate a sub-band decomposition tree with different levels due to different input signal. In the process of the sub-band decomposition, only when the intensity of the transform domain energy exceeds the time domain or a given threshold, the node will continue to decompose. Therefore, the algorithm avoids unnecessary decomposition, which can greatly reduce the calculation of addition and multiplication. Compared with a fixed structure filter, one of the biggest advantages of ATF is that it can adaptively change the hierarchical structure of the sub band filter banks, reduce the segmentation of the transform domain, and locate the interference signal of the frequency domain more accurately. Thus it can reduce the sensitivity of interference, and it is a robust technique for suppressing interference.

II. ADAPTIVE WAVELET PACKET TRANSFORM TECHNOLOGY

Fundamentally speaking, in the interference suppression techniques of changing domain, the system based on the trade-off is adaptive, but the typical method is to determine the position of the interference, then the coefficient of the transform domain is multiplied by 0 or 1.That is to remove the interference of the transform domain coefficient, and save the transform domain coefficients without interference. So in general sense, the decision is not based on a processing parameter, for example, the optimization of output SNR or BER. Therefore, the weight of continuous transformation is used and some optimization algorithm such as LMS algorithm is used to adjust the weights, which can improve the performance of the system to the best [3].

In ATF algorithm, after the position of the narrow band interference is determined, the next step is to remove the narrow band interference. The original algorithm is in the determination of the disturbed zone, the wavelet packet decomposition coefficient of the sub band is set to zero, so as to achieve the purpose of eliminating the interference. This will remove both of the frequency bands or the sub bands—the useless signal and the useful signal "polluted" by the interference signal, which will inevitably cause damage to the useful signal, so that the performance of the receiver will be reduced. Especially when the layers number of the decomposition is not large or when the power of the interference is high, the removed useful signal will be more [4]. This paper presents an improved method for the suppression of interference, in which the hybrid tree structure in ATF is combined with the adaptive filter. The "polluted" signal is determined by means of the mixed tree structure algorithm, and the least mean square (LMS) error algorithm is used to track the interference frequency adaptively and remove it, which can effectively reduce the loss of useful signal. The new algorithm can improve the performance of the whole system to a certain extent.

In order to improve the convergence rate of the adaptive algorithm, we decompose the input signal using the hybrid tree structure in ATF algorithm, a large attenuation factor is applied to the severely affected sub bands, thus, the initial value of the tap filter coefficient of the adaptive algorithm is better, which can effectively improve the convergence rate of the algorithm.

1. The model of interference suppression receiver based on the improved algorithm

The discrete wavelet packet decomposition and reconstruction algorithm is implemented using a Perfect Reconstruction Quadrature Mirror Filter (PR-QMF) group [5]. The basic idea of using adaptive wavelet packet transform to suppress the narrow band interference is shown in Fig 2.1, the receiver is sampled at chip rate to receive signals $r(n)$, and each time the wavelet packet transform is carried on to all chips in a symbol. In the specific description of the model, we first make a note to some symbols and marks:

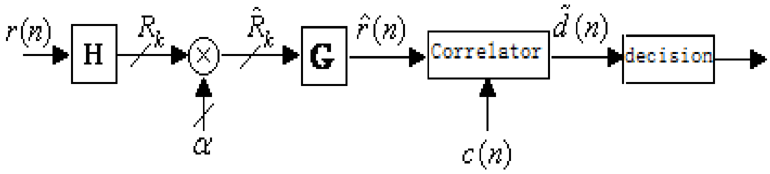

Fig 2.1: The model of adaptive wavelet packet interference suppression

$\mathbf{c}=\left[c_0\ c_1\ \cdots\ c_{N-1}\right]^T$ is the spread spectrum sequence vector for users;

$\mathbf{C}=\left[C_0\ C_1\ \cdots\ C_{N-1}\right]^T$ is the wavelet packet transform domain vector for the spread spectrum sequence vector of the user $\mathbf{c}$;

$\mathbf{r}=\left[r(0)\ r(1)\ \cdots r(N-1)\right]^T$ is the received signal vector, referred to $\mathbf{r}=\left[r_0\ r_1\ \cdots r_{N-1}\right]^T$;

$\mathbf{R}=\left[R_1\ R_2\ \cdots R_M\right]^T$ is wavelet packet transform vector of received signal;

$\hat{\mathbf{R}}=\left[\hat{R}_1\ \hat{R}_2\ \cdots \hat{R}_M\right]^T=\left[\alpha_1 R_1\ \alpha_2 R_2\ \cdots \alpha_M R_M\right]^T$ $\boldsymbol{\alpha}=\left[\alpha_1\ \alpha_2\ \cdots \alpha_M\right]^T$ is adaptive filter tap coefficient vector;

$\hat{\mathbf{r}}=\left[\hat{r}(0)\ \hat{r}(1)\ \cdots \hat{r}(N-1)\right]^T$ is the reconstruction signal vector of received signal $\mathbf{r}$;

N is the length of spread spectrum sequence;

M is the number of adaptive filter tap coefficients.

$\mathbf{H}$ and $\mathbf{G}$ respectively represent the analysis and synthesis filter matrix in PR-QMF, corresponding represent wavelet packet decomposition and reconstruction operation.

Wavelet packet decomposition is carried on to the received signal $r(n)$ which is sampled at chip rate according to the law of the energy accumulation [6]. We hope that in the transform domain, we have a large attenuation of the sub bands which are affected by the interference signal, while smaller or less of the sub bands which are affected little or no influence. Therefore, we use the appropriate factor α_k and R_k to get multiplication $\hat{R}_k$ to adjust the vector $\mathbf{R}$ adaptively, in order to achieve the purpose of suppressing the narrowband interference. The transform domain signal will be re-transformed to get a recovery signal, and then the decision variable is obtained by correlation with the local spreading sequence. Finally the signal is sent to the judge for a decision.

In the Gauss channel DSSS system, the discrete form of the baseband received signal can be written as

$$r(n,k)=d(n)c(k)+N(n,k)+I(n,k) \qquad k=0,1,\ldots,N-1$$

In the above equation, $d(n)$ is the data symbol, a random variable uniformly distributed on $\{+1,-1\}$, $c(k)$ is the spread spectrum sequence, the processing gain is N; $N(n,k)$、$I(n,k)$ are the Gaussian noise and narrow-band interference signals of N-th symbols and K-th chip. So we have

$$\mathbf{R}=\mathbf{Hr}$$

$$\hat{\mathbf{R}}=\left[\alpha_1 R_1\ \alpha_2 R_2\ \cdots\ \alpha_N R_N\right]^T$$

$$\hat{\mathbf{r}}=\mathbf{G}\hat{\mathbf{R}}$$

$$\tilde{d}(n) = \hat{\mathbf{r}}^T\mathbf{c} = \hat{\mathbf{R}}^T\mathbf{G}^T\mathbf{c}$$

Note that the operator $\mathbf{H}$ and $\mathbf{G}$ are perfect reconstructed, so $\mathbf{G} = \mathbf{H}^T$. Then we can get that

$$\tilde{d}(n) = \hat{\mathbf{R}}^T\mathbf{G}^T\mathbf{c} = \hat{\mathbf{R}}^T\mathbf{H}\mathbf{c}$$

In the above equation, $\mathbf{Hc}$ happens to be the wavelet packet transform of user spread spectrum sequence C, so the decision variable $\tilde{d}(n) = \hat{\mathbf{R}}^T\mathbf{C}$.

This shows that, at the receiver end, as long as the spread spectrum sequence is carried out wavelet packet decomposition according to the same mode of the receive signal, it will be multiplied with the receive signals in the transform domain component and corresponding adaptive filter tap coefficient, then take their sum directly to the judge. Thus, the reconstruction process can be omitted, and the calculation amount is reduced effectively. Based on this, we obtain a new structure of the receiver, as shown in Figure 2.The simulation results are based on the model of the receiver in Fig 2.2.

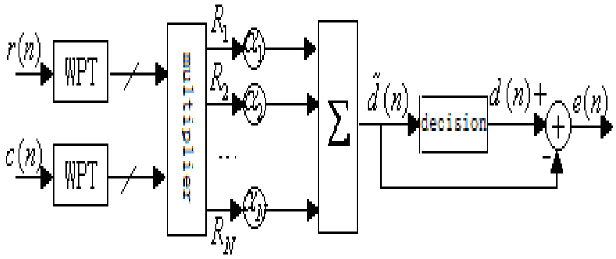

Fig 2.2 A receiver model based on improved algorithm

LMS algorithm is used in the adaptive notch filter, and it is a stochastic gradient algorithm. Theoretically speaking, as long as the iteration step size factors are small enough, the adaptive filter tap coefficients will converge to optimal Wiener solution, and can get the optimal filtering of minimum square error. The adaptive filter has two weights, which can respectively tracking the phase and amplitude changes of interference. By adjusting the two weights, achieve the purpose with the original input end of the interference component, to achieve cancellation.

The specific steps of adaptive suppression narrow-band interference algorithm based on the improved ATF wavelet packet transform are as follows:

(1)In the traditional ATF algorithm, according to the energy aggregation of the M band transform, the received signal and the spread spectrum sequence are decomposed into binary or trinary [6].

$$G = \frac{\delta_x^2}{\left[\prod_{i=1}^{M}\delta_i^2\right]^{1/M}}$$

Where δ_x^2 is the variance of input signal, δ_i^2 is the output variance of the i-th sub-band. The bigger the G is, the less flat the frequency spectrum of the output signal is, and the signal energy is more focused on some sub bands. If $G \le T$, means that the signal spectrum is flat, and the decomposition will stop (T is the pre-defined threshold). Otherwise, if $G(2) \ge G(3)$, two band decomposition is used, and if $G(2) < G(3)$,then three band decomposition is used.

(2)If the decomposition should continue, then take δ_i as reference values: if δ_i is greater than prior given threshold f (f determined by the characteristics of wavelet packet decomposition coefficient of spread spectrum signal), means that the node corresponding to δ_i is focused on more narrow band interference energy, and will require to be decomposed, until it does not satisfy the condition $G > T$ or reach the expected frequency resolution.

(3)See all the terminal nodes of the irregular tree after decomposition, the corresponding sub band components in which δ_i is greater than the threshold f are subjected to a smaller weight factor, because according to the algorithm here they focus more narrowband interference energy, and the remaining sub with weight factors were taken 1, this initial weight factor can significantly improve the convergence speed of the algorithm.

(4) Suppress the narrow band interference by using adaptive notch processing for the nodes presence of interference by LMS algorithm.

In the adaptive LMS algorithm, the tap coefficients of the filter are trained with a training sequence. The concrete realization of the algorithm is divided into three steps:

①Calculate the decision variables

$$\tilde{d}(n) = \hat{\mathbf{R}}^T\mathbf{C}$$

②Estimate the error

$$e(n) = d(n) - \tilde{d}(n)$$

③Modify adaptive filter's tap coefficient

$$\boldsymbol{\alpha}(n+1) = \boldsymbol{\alpha}(n) + \mu \mathbf{R}(n) e(n)$$

In the Equation, μ is the iterative step.

2. The selection of threshold T and f

Both thresholds T and f are determined by the characteristics of white noise under the wavelet transform. A zero mean white noise sequence is still a zero mean sequence in the wavelet transform, and the coefficient sequence of the wavelet base is still the white noise sequence of the same as variance [7]. Spread spectrum signal is with the characteristics of white noise signal, and the metric G of energy concentration degree defined by wavelet packet algorithm is normalized, so G of sub-band without narrowband interference should be close to the constant value 1; and for the presence of narrowband, G should greater than 1.

In the following simulation experiment, the narrowband interference energy is strong, and the length of the signal is finite, the calculated value of G and the theory value of white noise exist deviation. Combine of all these factors, threshold T can be far greater than 1. Each sub-band coefficient standard deviation of received signal will be increased with the increase of signal-to-noise ratio (SNR). But taking into account the signal to interference is relatively high, i.e. and the narrowband interference energy is strong, under prior knowledge for the spread spectrum signal, for different signal-to-noise ratio, the threshold f can be set to unity for a node to judge the existence of narrowband interference.

3. Convergence rate of the algorithm

LMS algorithm has the characteristics of simple and stable, but the convergence speed is slow. The orthogonal transformation can effectively improve the convergence rate of the algorithm. At present, there are some researches on the adaptive equalization algorithm based on wavelet transform [8]. On these basis, the LMS algorithm based on the wavelet packet transform is proposed by Huang Kui. As indicated by the literature [9], by wavelet packet transform, signal autocorrelation matrix showing zonal distribution, the energy mainly concentrated in the vicinity of the diagonal and in most cases close to diagonal or with diagonal distribution, the input maximum and minimum feature ratio value of the autocorrelation matrix of decreased significantly, so the LMS algorithm convergence speed is improved. And with the increase of wavelet packet decomposition series, the self correlation matrix of signals tends to be more angular distribution, therefore the convergence rate will accelerate, but when the decomposition series reaches a certain value, due to the variation of auto correlation matrix is no longer obvious, the convergence rate will not significantly accelerate with the increase of the decomposition series.

III. SIMULATION AND ANALYSIS

In computer simulation, the narrow band interference is simulated with multiple frequencies sine wave. Different frequency sinusoidal interference distributed in different sub bands, respectively, the system error rate is simulated using adaptive notch and the sub-band elimination, and analysis and comparison of the two methods are made.

In simulation, assuming that the system work in the Gaussian channel, the length of the direct sequence spread spectrum system m sequence is 63, BPSK modulation, using the 4 order Daubechies wavelet filter coefficients for decomposition and reconstruction, the power ratio of the useful signal and the total interference signal is -40dB. The normalized angular frequency of the three single frequency interference were $0.1271\,rad/s$, $2.5322\,rad/s$, $0.9125\,rad/s$, the phase is a random variable uniformly distributed on the interval $[0, 2\pi]$. According to the threshold T and f discussed, we select $T = f = 2$ in the algorithm simulation.

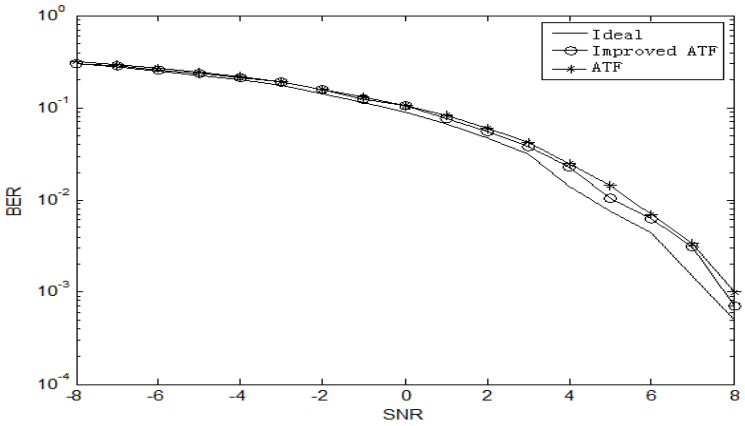

Fig 3.1: BER under a single frequency interference

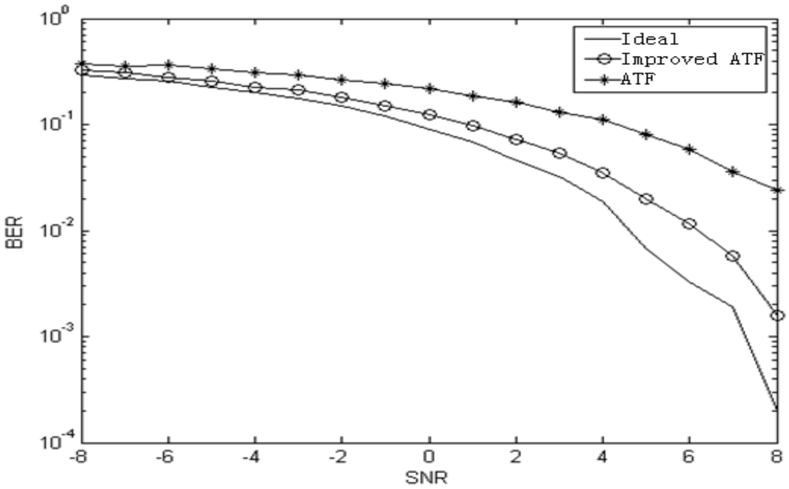

Fig 3.2: BER under three frequency interference

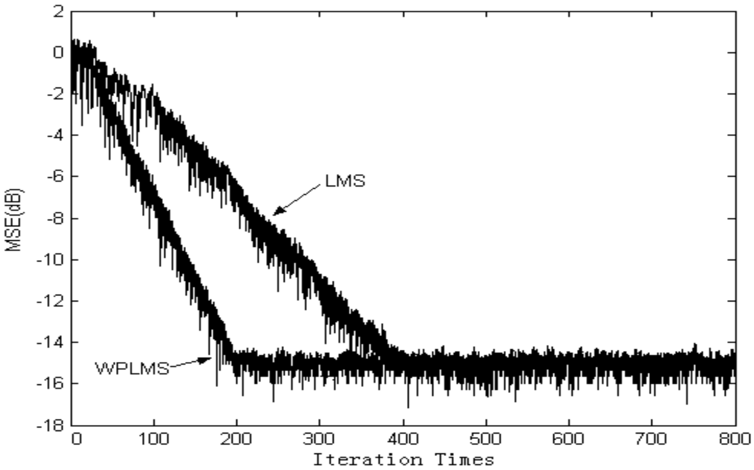

Fig 3.3: Convergence characteristics of adaptive algorithm

Fig 3.1 shows the presence of single frequency interference, the bit error rate（BER） performance curves are compared between the ideal case (in the absence of narrowband interference),the ATF algorithm (sub-band elimination) and modified ATF algorithm (adaptive notch filter). From Fig 3.1, we can see that it is more effective using the adaptive notch filter to eliminate the interference than sub-band elimination, and the performance curve has been close to the ideal curve of no interference. Due to the single frequency interference in sub-band accounts for only a little, sub-band elimination will not cause too big effect to wideband spread spectrum signal receiving, so its performance is also very good.

Fig 3.2 shows the bit error rate curve of the adaptive notch and the sub-band elimination when the three interferences occur at the same time in the low, medium and high frequency bands. As can be seen from the figure, the system performance is significantly better when using adaptive method than sub-band elimination. This is because after decomposition, three frequency interference, occupy more sub-band. Taking up sub-band elimination, all these wavelet packet coefficient will be zero, at the same time the useful signal will also be discarded together, the loss of useful signal is larger, which is equivalent to reducing the signal to noise ratio, so the performance of the signal is poor.

Fig 3.3 shows the convergence of LMS and LMS based on Wavelet Packet Transform (WPLMS) under tone interference. Mean Square Error (MSE) curves are obtained from 800 independent iterative processes. It can be seen that when the LMS algorithm is adopted, it takes 400 iterations to reach the steady state, and the WPLMS can converge to the steady state only 200 times. Thus, the orthogonal transformation can improve the convergence rate of the algorithm.

IV. CONCLUSION

In this paper, an improved adaptive wavelet packet interference suppression technique based on ATF is studied. When there are multi tone interferences, the bit error rate performance of the receiver based on this algorithm is simulated. Simulation results show that the improved algorithm can effectively improve the BER performance of the system with high power narrow-band interference, and compared with the traditional LMS adaptive algorithm, it can effectively improve the convergence speed of the algorithm.

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