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Extraction of Polyphase Modulation Parameters Using Cyclic Spectral Analysis

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Abstract. This paper presents an efficient algorithm to extract the polyphase modulation parameters from intercepted signals. The algorithm is essentially based on the cyclic spectral analysis method. A calculation of parameters extraction relative error as a function of different signal patterns reveals the performance of the algorithm, which is validated via simulation examples.

I. INTRODUCTION

Polyphase modulation technique is widely used in low probability of intercept wideband signal [1], which makes Non-cooperative intercept receivers difficulty in looking for these signals. The conventional signal processing methods such as energy detection and power spectral density analysis cannot provide details about the polyphone modulation parameters. Recent research [2] have shown that intercept receiver can increase its processing gain by implementing cyclic spectrum signal processing method so as to raise the effect of parameter extraction.

This paper investigates an algorithm to efficiently extract the key polyphase modulation parameters (bandwidth B, code length N_c, code rate R_c, codes period t_m, modulation period T and carrier frequency f_c etc.) using cyclic spectral analysis. The method based on cyclostationary characters is particularly applicable since the majority of polyphase modulation signals have the first-and second-order statistic periodicity [3].

We evaluate the sensitivity of the algorithm using typical polyphase signals for signal-to-noise ratios (SNRs) of 0dB and -6dB. To illustrate the algorithm, a Frank code is used with M sub-codes, a carrier frequency of $f_c = 1000$ Hz and sampling frequency of $f_s = 7$ kHz with SNR=0dB [4]. The Frank code is a polyphase code with each sub-code phase defined as

$$\phi_{i,j} = \frac{2\pi}{M}(i-1)(j-1) \tag{1}$$

where $i = 1,2,\cdots M$, $j = 1,2,\cdots M$, $N_c = M^2$. The number of codes is M, total code length is N_c and code period is t_m [5].

II. CYCLIC SPECTRAL ANALYSIS DESCRIPTION

A. The Ttime-smoothing FFT Accumulation Method(TFAM). Cyclostationary processing transforms the signals into the frequency-cycle frequency domain. Measurement of cyclic autocorrelation and spectral correlation density (SCD) in signal analysis constitutes what is referred to as cyclic spectral analysis [6]. The time-smoothing FFT accumulation method was developed to reduce spectrum estimation computations [7]:

$$S^{\gamma}_{X_{N'}}(n,k) = \frac{1}{N}\sum_{n=0}^{N-1}[\frac{1}{N'}X_{N'}(n,k+\frac{\gamma}{2})X^{*}_{N'}(n,k-\frac{\gamma}{2})] \tag{2}$$

Where

$$X_{N'}(n,k) \triangleq \sum_{n=0}^{N'-1} w(n)x(n)e^{-(j2\pi kn)/N'} \tag{3}$$

is the discrete Fourier transform of $x(n)$, $w(n)$ is the data taper window, k and γ are discrete equivalents of frequency and cyclic frequency respectively. N represents the total number of discrete samples within the observation time, and N' represents the number of points within the discrete STFT [8].

The TFAM method estimation results of a Frank signal are shown in Fig.1 and Fig.2, which represent signal parameters in bi-frequency plane respectively. Fig.1 illustrates the measurement of N_c and R_c; Fig.2 illustrates the measurement of f_c and R_c [8]. In Fig.1, we can get N_c by counting the number of close curves in supports area, namely $N_c = 9$; we can get R_c by measuring the distance between main and sub- support areas on cyclic frequency axis. In Fig.2, we can get R_c in the same manner and get carrier frequency by projecting the peak value of support area on cyclic frequency axis, namely $f_c = F_c / 2$. We can calculated more parameters in the basis of N_c, R_c and f_c, which will discuss in next section.

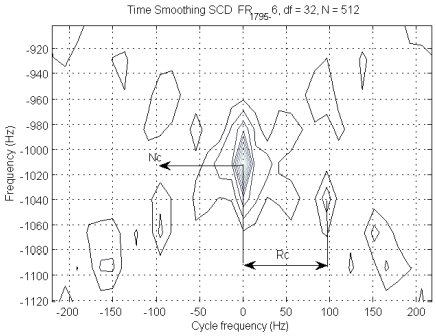

Figure 1. Results of TFAM SCD patterns for the frank code with $N_c = 9$, $f_c = 1$KHz, SNR=-6dB on cyclo frequency 0

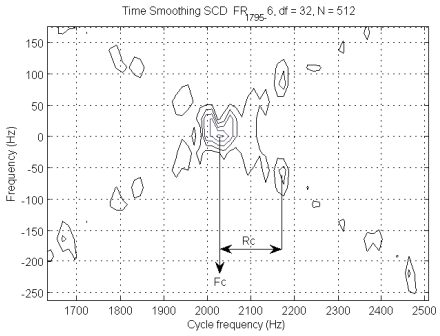

Figure 2. Results of TFAM SCD patterns for the same signal on cyclo frequency 2000Hz

B. Direct Frequency-Smoothing Method(DFSM). The direct frequency-smoothing method is computationally superior to indirect algorithms such as Wigner-Ville Distribution. The basis for the DFSM is the discrete time frequency-smoothed cyclic periodgram represented by [7]:

$$S^{\gamma}_{x_N}(n,k)_{\Delta k} = \frac{1}{N}\sum_{n=0}^{N-1} X_N(n,k+\frac{\gamma}{2})X_N^*(n,k-\frac{\gamma}{2}) \tag{4}$$

Where

$$X_N(n,k) = \sum_{n=0}^{N-1} w(n)x(n)e^{-(j2\pi kn)/N} \tag{5}$$

is the discrete Fourier transform of $x(n)$, $w(n)$ is the rectangular windows of length N ,which is the total number of points of the FFT related to the total observation time, the other parameters are the same meaning as in (2)and (3).

The DFSM method SCD estimation results of the Frank code signal are shown in Fig.3, we can get the 3 parameters in the same method as in Fig.1 and Fig.2.

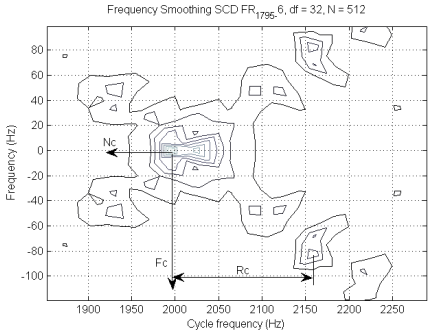

Figure 3. Results of DFSM SCD patterns for the frank code with the same signal on cyclo frequency 2000Hz

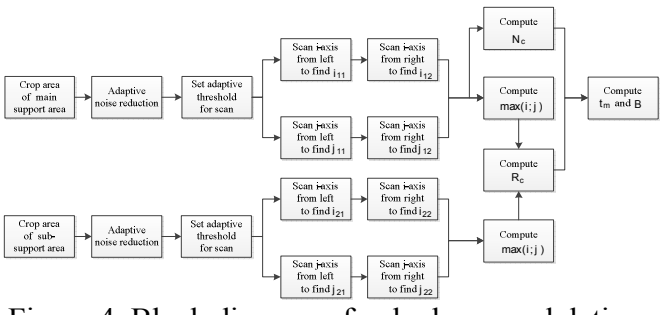

Figure 4. Block diagram of polyphase modulation signal extraction processing using cyclic spectral analysis in bifrequency plane

Since the modulation parameters N_c, R_c and f_c can be obtained from SCD examination results, bandwidth B and code period t_m, are easily calculated by (6)and (7) [1].

$$t_m = \frac{T}{N_c} = \frac{1}{N_c R_c} \tag{6}$$

$$B = \frac{1}{t_m} = N_c R_c \tag{7}$$

With -6dB noise added, measurement ability in Fig.1, 2 and 3 remains fairly robust [9] since white Gaussian noise is not correlated and is suppressed in this cyclic spectral analysis.

III. PARAMETER EXTRACTION ALGORITHM BASED ON CYCLIC SPECTRAL ANALYSIS

A block diagram of the polyphase modulation extraction algorithm that uses the DFSM results is shown in Fig. 4.

The first step in the algorithm is to crop one of the quadrants within the area of supports matrix [5]. After cropping the area of support, an adaptive threshold operation is performed to reduce the amount of noise present in bifrequency plane. This is followed by scanning axis for detection of maximum magnitude in bifrequency plane. The scanning results are used to measure the code rate R_c and code length N_c, subsequently calculate the modulation period t_m and bandwidth B.

An approximation to the PDF is constructed by scanning both the i-and j-axis and finding the magnitude above threshold corresponding to each index in main and sub-support areas simultaneously. The horizontal scan gives the i-index values as

$$x_i = \sum_{j=1}^{N} S(i,j) \tag{8}$$

And

$$PDF_i = \frac{x_{i=(1:N)}}{\sum x_{j=(1:N)}} \tag{9}$$

This energy distribution enables the location of the support areas in the bifrequency plane to be determined. In main support area, the scan on both i- and j- axis from the lower index values to the higher index values to a level above -6dB gives i_{11} and j_{11}; from the higher index values to the lower index values to a level above -6dB gives i_{12} and j_{12}. In sub-support area, the scan remains the same as in main support area and give i_{21}, i_{22}, j_{21}, j_{22}. Thus

$$N_c = \frac{i_{12} - i_{11}}{2} \tag{10}$$

And

$$R_c = \left| i_{S(i_1,j_1)_{\max}} - i_{S(i_2,j_2)_{\max}} \right| \tag{11}$$

which $i_{S(i_1,j_1)_{\max}}$ stands for the value of i corresponding to the maximum magnitude in main support area; $i_{S(i_2,j_2)_{\max}}$ stands for the value of i corresponding to the maximum magnitude in sub-support area.

In terms of a noisy background, the scan method described above will degrade the signal's PDF. The procedure can be modified to use an adaptive noise filter [10].

IV. TESTING THE ALGORITHM

A. Noise Level Setup and Resolution Analysis. To evaluate the algorithm's accuracy, the parameter extraction algorithm is tested in this section.

The algorithm checks the signal for the existence of the noise and uses cyclic spectral analysis method to extract the parameters. The signals with no noise can analyzed with 64Hz of frequency resolution but the signals with -6dB noise cannot analyzed with it due to the computational difficulties. The signal with -6dB noise do not have a code rate solution, since the resolution of the frequency should be smaller than the largest parameter being measured or extracted [11].

$$R_c = \frac{1}{N_c t_m} = \frac{1}{16 * 10^{-3} s} = 62.5Hz < 64Hz \tag{12}$$

B. Results an conclusion. If a^* is a measurement value of a quantity whose exact value is a, then the absolute value of the relative error ε_r is defined by:

$$\varepsilon_r = \left|\frac{a^* - a}{a}\right| = \left|\frac{\text{Error}}{\text{True value}}\right| \tag{13}$$

and gives an indication of how good a measurement is relative to the size of the thing being measured.

Six polyphase modulation signals are extracted with FAM and DFSM [12] method separately, The parameters used to generate the polyphase modulations are f_s =7000Hz, f_c =1000Hz, B=1000Hz, and N_c=4, 9, 16. The results of parameters extraction are showed in Table 1 and Table 2 corresponding to FAM and DFSM separately. The true and measured parameters are plotted in Fig. 5-7.

For the test results, The carrier frequency error is very small for signal without noise, but for signal with -6dB noise higher errors occur for small values of N_c, if the frequency and cyclic frequency resolution of the cyclic spectral analysis is increased, the error in estimating f_c decreases. The error in the estimation of N_c is related to t_m and B since $N_c = t_m \times B$. The overall errors are reasonably small for without noise condition. For the condition SNR=-6dB, the largest errors occur for N_c=9, 16. That is ,the test shows the important result that for smaller values of SNR, the errors in the extracted parameters are smaller for larger values of N_c.This is due to the larger processing gain obtained with larger number of sub-codes [13]. Note that another advantage to the approach is that the extraction algorithm is not affected from the non-cyclostationary noise significantly. The reason is that the non-cyclostationary noise is not correlated and is suppressed significantly in the spectral correlation technique [14].

TABLE I. MEASURED PARAMETERS WITH FAM

Signal	True Parameters					Measured Parameters				
	f_c	t_m	N_c	R_c	B	f_c	tm	N_c	R_c	B
1	1K	1	4	250	1k	1004	1	4	246	1012
2	1K	1	4	250	1k	1108	0.9	4	248	1018
3	1K	1	9	111	1k	998	0.9	10	123	1231
4	1K	1	9	111	1k	1092	0.8	9	138	1236
5	1K	1	16	62.5	1k	998	1	14	68	957
6	1K	1	16	62.5	1k	1006	1	15	66	992

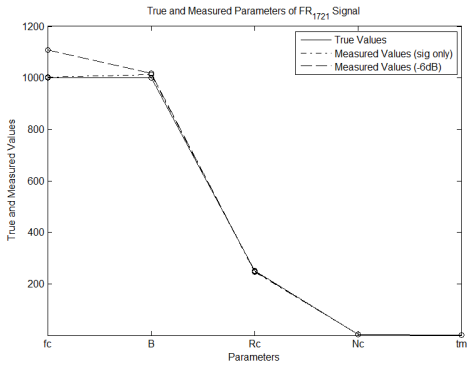

Figure 5. True and Measured Parameters of sig1 and sig2

TABLE II. MEASURED PARAMETERS WITH DFSM

Signal	True Parameters					Measured Parameters				
	f_c	t_m	N_c	R_c	B	f_c	tm	N_c	R_c	B
1	1K	1	4	250	1k	998	1	4	246	991
2	1K	1	4	250	1k	1121	1	3	248	725
3	1K	1	9	111	1k	998	0.8	11	109	1230
4	1K	1	9	111	1k	1198	0.8	9	136	1240
5	1K	1	16	62.5	1k	1148	1	14	55	957
6	1K	1	16	62.5	1k	1150	1	15	66	986

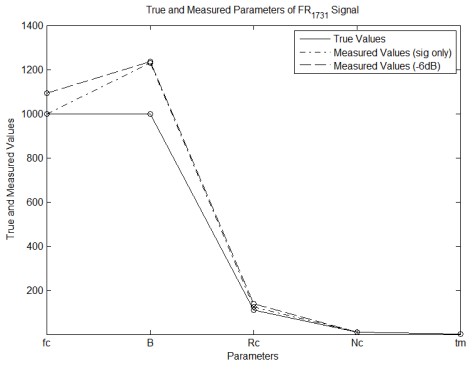

Figure 6. True and Measured Parameters of sig3 and sig4

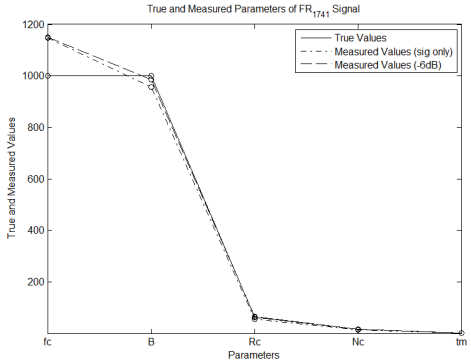

Figure 7. True and Measured Parameters of sig5 and sig6

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