CYCLOSTATIONARY ANALYSIS OF POLYTIME CODED SIGNALS FOR LPI RADARS

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Abstract

In Radars, an electromagnetic waveform will be sent, and an echo of the same signal will be received by the receiver. From this received signal, by extracting various parameters such as round trip delay, doppler frequency it is possible to find distance, speed, altitude, etc. However, nowadays as the technology increases, intruders are intercepting transmitted signal as it reaches them, and they will be extracting the characteristics and trying to modify them. So there is a need to develop a system whose signal cannot be identified by no cooperative intercept receivers. That is why LPI radars came into existence. In this paper a brief discussion on LPI radar and its modulation (Polytime code (PT1)), detection (Cyclostationary (DFSM & FAM) techniques such as DFSM, FAM are presented and compared with respect to computational complexity.

Keywords—*LPI Radar, Polytime codes, Cyclostationary DFSM, and FAM*

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1. INTRODUCTION

The radar is an abbreviation for RAdio Detection And Ranging expression. In general, the radar systems use modulated waveforms and directive antennas to transmit electromagnetic energy into a specific volume in space to search for targets. Objects (or targets) within a special search volume will reflect back to the radar a portion of this energy (radar returns or echoes). These echoes are processed by the radar receiver to extract target information such as range, velocity, angular position, and other target identifying characteristics [1].A low probability of intercept (LPI) radar is defined as radar that uses a special emitted waveform intended to prevent a non cooperative intercept receiver from intercepting and detecting its emission. The LPI radar has different modulation and detecting techniques of them we are going to discuss following. A study conducted on the implementation of Barker code and linear frequency modulation pulse compression technique has shown that the SNR and Range resolution were improved even for targets having very low RCS, but failed to prove quantitatively [2]. A group of scientists presented modeling and analysis of LPI radar signals using Barker and polyphase codes which only gives the time and frequency changes, but could not extract the required parameters such as center frequency, Bandwidth, and code rate [3]. FMCW modulated LPI signals were analyzed using Wiener Ville Distribution, the performance of which is limited for the estimation of the center frequency in the frequency agility conditions [4]. This paper focused on the Polytime codes modulated LPI signals and extraction of its parameters using efficient methods of cyclostationary signal processing. The analysis performed under different SNR conditions with different methods such as DFSM, FAM and compared quantitatively.

2. POLYTIME CODE (T1(n))

The Polytime codes are counterparts of polyphase codes in which phase along with time spent at each phase state will changes. There are four types of Polytime codes T1, T2, T3 and T4 out of which T1 is discussed below. The T1, T2 codes arise from stepped-RF waveform whereas the T3, T4 from linear-FM waveforms. T1(n) is an approximation to stepped-RF waveform with zero beat at its leading edge. The ‘n’ indicates the number of phase states used to approximate the underlying waveform. The proposed work uses only two phase states (0 and 180). The signal is of 16µSec is divided into four segments each of 4µSec. Among the four segments the first segment has no signal, and the second segment has one full cycle (3600). The third segment consists of two full cycles (7200), and the fourth segment has three full cycles (10800) resulting in a total accumulated phase of 21600. The Fig.1 shows the unwrapped accumulated phase and quantized wrapped phase of a stepped-RF signal.

The above wrapped phase quantized to 00 and 1800 can be directly generated by using the equation.(1)

$$\varphi(t) = \mathrm{MOD}\left[\frac{2\pi}{n}\mathrm{INT}(kt - jT)\frac{jn}{T}, 2\pi\right] \quad (1)$$

n = number of phase states
k = number of segments
T = Total code duration
j = segment number

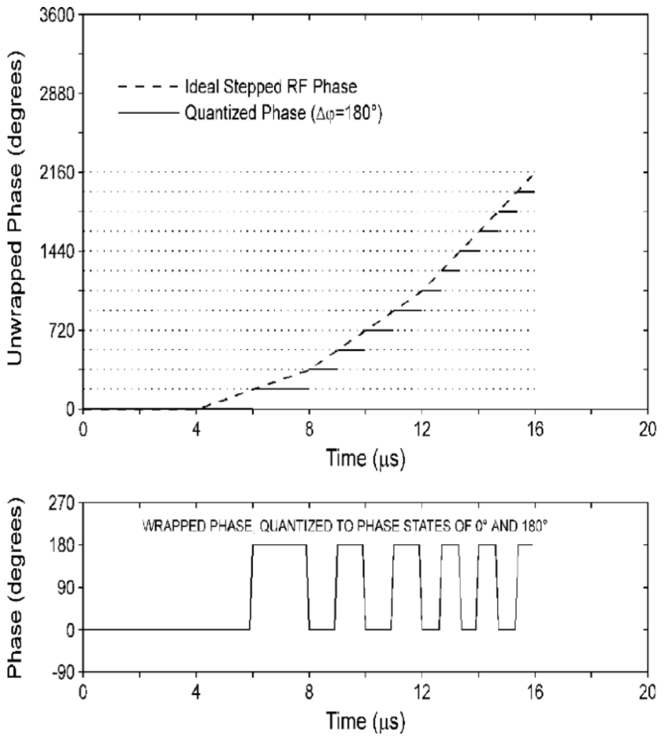

Fig -1: Polytime waveform derived from stepped-RF signal

3. CYCLO STATIONARY SIGNAL PROCESSING

In general all periodic signals are deterministic in nature. By applying directly Fourier transform, we can extract parameters from the signal. Coming to modulated signals they are not truly periodic. By performing a nonlinear transformation, we can convert them into periodic. Previously a quadratic transform is used in which the signal is squared followed by applying FFT to get the spectral lines. However, squaring is not recommended in some cases such as PCM in which only amplitude of ±1 that on squaring hides all the spectral lines giving a DC 1. So a delay must be introduced. Let x(t) be a Polytime coded signal which is not periodic then we convert this signal into a periodic signal y(t) as given in the equation (2)

$$y(t) = x(t)x(t-\tau) \quad (2)$$

In cyclostationary there are two methods time smoothing FFT accumulation method and direct frequency smoothing method. The cyclic auto correlation function and spectral correlation density function are the two important parameters in cyclostationary.

The cyclic auto correlation function can be given in equation (3)

$$R_x^{\alpha}(\tau) = \lim_{T\to\infty}\frac{1}{T}\int_{-T/2}^{T/2} x\left(t+\frac{\tau}{2}\right)x^*(t-\frac{\tau}{2})e^{-j2\pi\alpha t}\,dt \quad (3)$$

The spectral correlation density is given in equation (4)

$$S_x^{\alpha}(f) = \int_{-\infty}^{\infty} R_x^{\alpha}(\tau)e^{-j2\pi f\tau}\,d\tau$$
$$= \lim_{T\to\infty}\frac{1}{T}X_T\left(f+\frac{\alpha}{2}\right)X_T^*(f-\frac{\alpha}{2}) \quad (4)$$

Where $X_T(f) = \int_{-\frac{T}{2}}^{\frac{T}{2}} x(u)e^{-j2\pi fu}\,du \quad (5)$

From the spectral correlation density plot we can extract the parameters easily.

3.1 FFT Accumulation Method

The time smoothing FFT accumulation method is one of the most used techniques which involves a large number of small computations. The Fig.2 shows the block diagram of the FAM method.

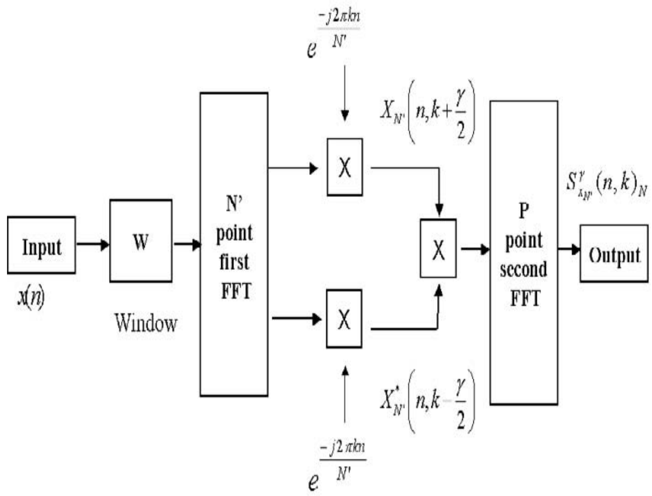

Fig.2.FFT Accumulation Method

The algorithm consists of three stages: computation of the complex demodulates (divided into data tapering, sliding N' point Fourier transform, and baseband frequency translation sections), computation of the product sequences and smoothing of the product sequences. The parameter N represents the total number of discrete samples within the observation time, and N' represents the number of points within the discrete short-time (sliding) FFT. In the FAM algorithm, spectral components of a sequence, x(n), are computed using (4). Two components are multiplied (3) to provide a sample of a cyclic spectrum estimate representing the finite channel pair region on the bi-frequency plane. There are N2 channel pair regions in the bi-frequency plane. A sequence of samples for any particular area may be obtained by multiplying the same two elements of a series of consecutive short-time sliding FFTs along the entire length of the input sequence. After the channelization performed by an N'-point FFT sliding over the data with an overlap of L samples, the outputs of the FFTs are shifted in frequency in order to obtain the complex demodulate sequences. Instead of computing the average of the product of sequences between the complex demodulates, they are Fourier-transformed with a P-point (second) FFT. The computational efficiency of the algorithm is improved by a factor of L, since only N/L samples are processed for each point estimate. With fs the sampling frequency, the cycle frequency resolution of the decimated algorithm is defined as γres = fs/N (compare to Δα = 1/Δt), the frequency resolution is kres = fs/N’ (compare to Δf = 1/TW), and the Grenander’s Uncertainty Condition is M = N/N’>>1.

3.2 DFSM

Direct frequency-smoothing algorithms first compute the spectral components of the data and then execute spectral-correlation operations directly on the spectral components. The direct frequency-smoothing method is computationally superior to indirect algorithms that use related quantities such as the Wigner-Ville Distribution, but DFSM is usually less efficient than a time-smoothing approach.

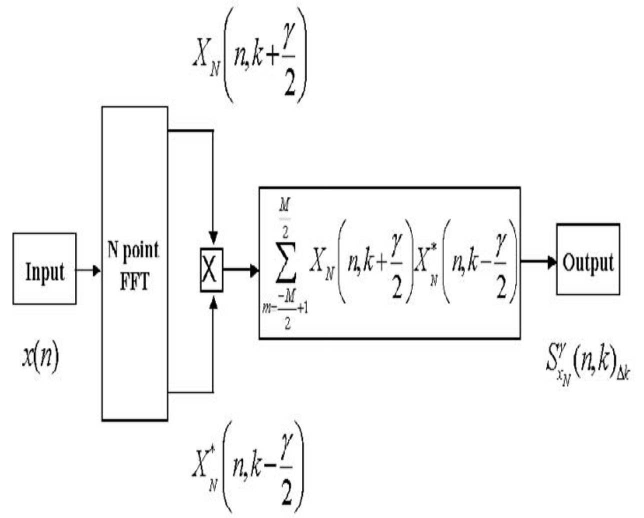

Fig.3.Direct Frequency Smoothing Method

The basis for the DFSM is the discrete time frequency-smoothed cyclic periodogram represented by equation (6)

$$S_{X_N}^{\gamma}(n,k) = \frac{1}{N}\sum_{n=0}^{N-1} X_N\left(n, k+\frac{\gamma}{2}\right) X_N^{*}\left(n, k-\frac{\gamma}{2}\right) \quad (6)$$

Where

$$X_N(n,k) = \sum_{n=0}^{N-1} w(n)x(n)e^{-j2\pi kn/N} \quad (7)$$

is the discrete Fourier transform of x(n), w(n) is the rectangular window of length N that is the total number of points of the FFT related to the total observation time, Δt, γ is the cycle frequency discrete equivalent, the frequency-smoothed ranges over the interval |m| ≤ M/2, and Δk ≈ M · fs/N is the frequency resolution discrete equivalent.

4. RESULTS AND DISCUSSIONS

The Polytime codes best approximates the underlying stepped-RF waveform when compared polyphase codes in which only phase will changes and time spent at each phase state is constant. But in Polytime codes for a fixed number of phase states the time spent on each phase state will changes. If the number of phase states are increased then the time spent on each phase state decreases resulting in a signal which is very difficult to analyze. The minimum bit duration plays an important role which related to bandwidth as $BW=1/t_b$. The minimum bit duration depends on number of sub pulses (k), duration of each sub pulse (τ) and number of phase states (n) as follows

$$t_b = \frac{\tau}{(k-1)n} \quad (8)$$

The techniques FAM and DFSM will give results which agree well with the actual values. The similarities between the DFSM results and the FAM results go until a certain level; in the zoomed plots we see that the channel pair regions are a little different in shape and size, although they occur in the same values for frequency and cycle frequency. This may be the result of the different windows applied in each method (Hamming window for FAM and Rectangular window for DFSM).

The number of computations required for FAM are

N_{comp}=2*P*N’+P*N’*log2N’+2*P*N’+P*(N’)2+(N’)2(P/2) log2P

Where P = N/L and L=N’/4

The number of computations required for DFSM are

N_{comp}=N2+2*N*log2N.

So the computing time is also noticeably larger, two or three times more, for the DFSM routine in comparison to the FAM routine. The FAM implementation is recommended for long signals with a large number of samples. We can extract parameters from the signal even at an extreme signal to noise ratio conditions(-6 dB). The number of phase states and number of segments are difficult to find. If the overall code duration increases the resolution in extracted parameters will also increases.

4.1 Cyclostationary Analysis on Polytime signal T1(n)

Analysis of polytime code signal include signal, with and without the addition of White Gaussian Noise, and extraction of their main characteristics. The results of these signals are shown below. The contour plots show frequency-cycle frequency domain of the results.

First, the carrier frequency fc can be clearly found by the location of the modulation pattern, The T1 modulation shows up in the four quadrants centered on cycle frequency (γ), $\gamma = 2\ f_c = 2$ GHz as shown in fig.4.2. So, carrier frequency is measured as 1 GHz. Secondly, in fig. 4.2, bandwidth can be calculated as width from center of modulation pattern to end of pattern on cycle frequency axis, hence measured as 1750 MHz. Since

$$B = \frac{1}{t_b}$$

For polytime codes Rc = 1/T, The code rate (Rc) is distance between any two adjacent spots on cycle frequency axis and is measured as 62 MHz in Figure 4.3. So the total time duration of the signal will be calculated as T = 1/Rc = 16 nSec.

4.1.1 PT1_1000000_7000000_2_4_s

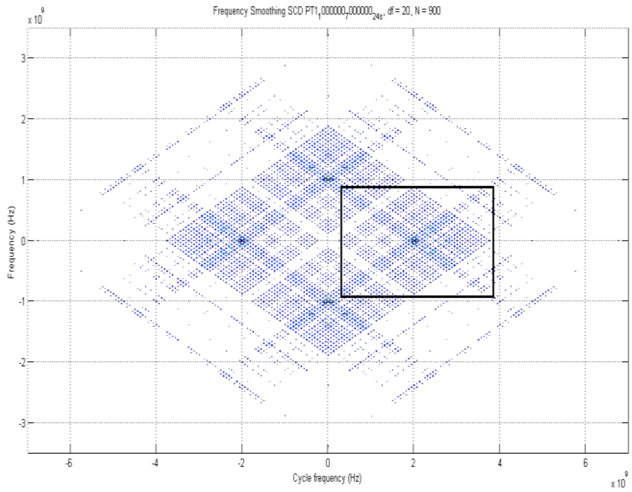

Fig 4.1: SCD patterns for the T1 code with n = 2, fc = 1 GHz, and T = 16 nSec

Fig.4.2 showing measurement of bandwidth which is the distance between center of modulation pattern to end of pattern on cycle frequency axis.

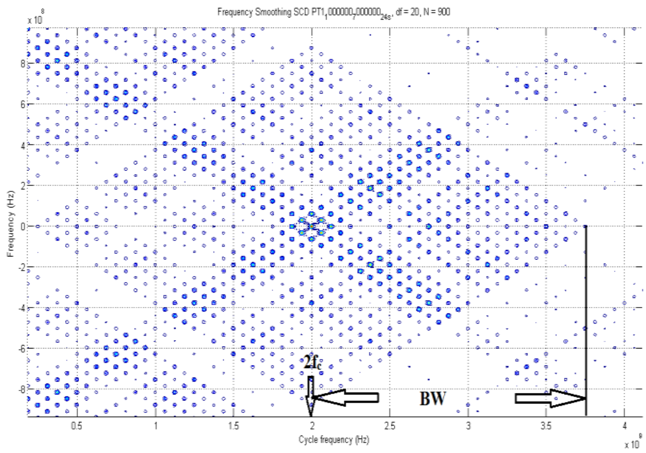

Fig 4.2: A close up of selected pattern and illustrating bandwidth measurement

Fig.4.3 showing measurement of Code-Rate which is distance between any two adjacent spots on cycle frequency axis.

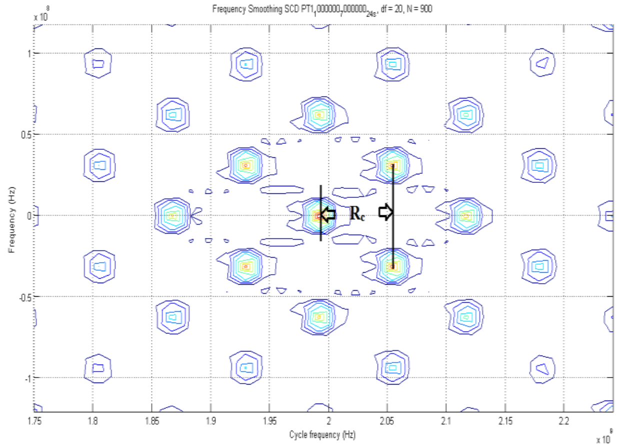

Fig 4.3: illustrating Code Rate measurement

Table-1: Comparison between measured and original Characteristics for signal PT1_1000000_7000000_2_4_s

FEATURE EXTRACTION FOR PT1_1000000_7000000_2_4_s		
CHARACTERSTIC	**ORIGINAL**	**MEASURED**
Carrier Frequency (f_c)	1000 MHz	1000 MHz
Bandwidth (B)	1750 MHz	1750 MHz
Code Rate (R_c)	62.5 MHz	62.5 MHz
Code Period (T)	16 nSec	16 nSec

The above results were analyzed under noise less condition in which last letter for the signal indiacaters different noise conditions.The analysis is carried out for s-signal only condition,0-0dB SNR, and -6dB SNR.

4.1.2 PT1_1000000_7000000_2_4_0

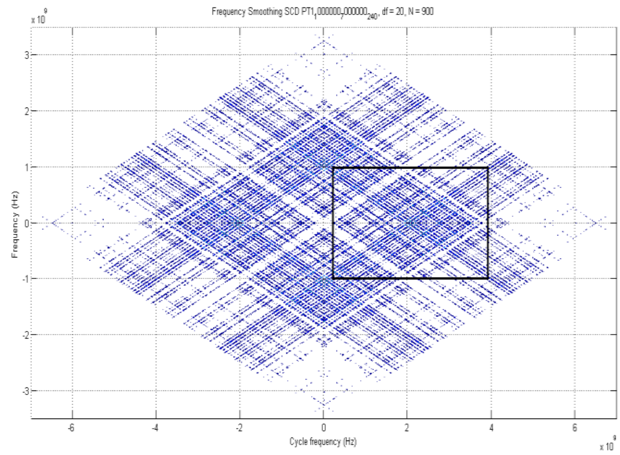

Fig 4.4: SCD patterns for the T1 code with SNR = 0 dB, fc = 1 GHz, and T = 16 nSec

Fig4.5 showing measurement of bandwidth which is the distance between center of modulation pattern to end of pattern on cycle frequency axis

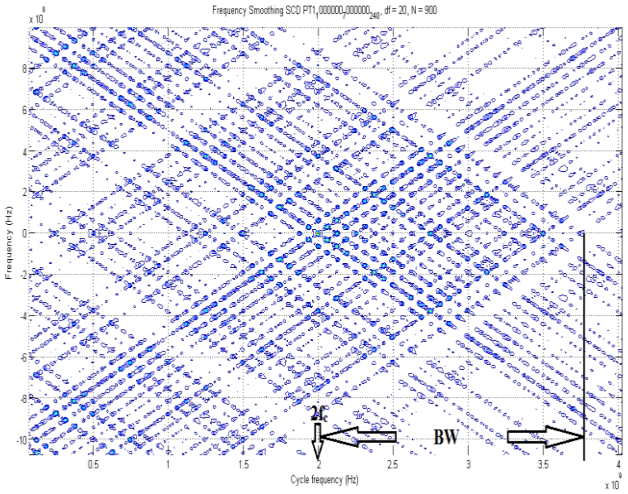

Fig 4.5: a close-up of selected pattern and illustrating Bandwidth measurement

Fig4.6 showing measurement of Code-Rate which is distance between any two adjacent spots on cycle frequency axis.

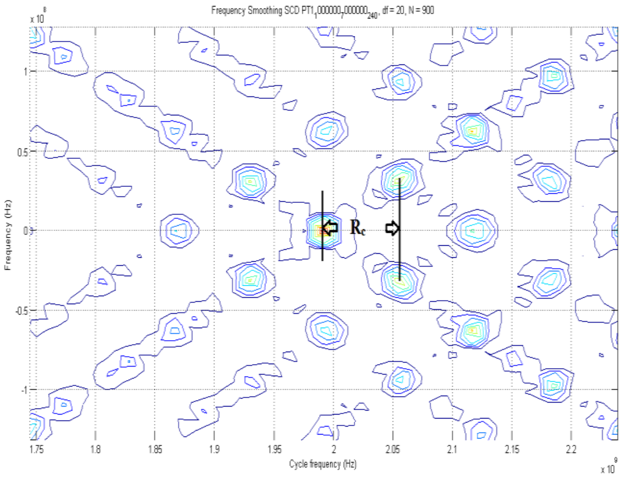

Fig 4.6: illustrating Code Rate measurement

Table-2: Comparison between measured and original Characteristics for signal PT1_1000000_7000000_2_4_0

FEATURE EXTRACTION FOR PT1_1000000_7000000_2_4_0		
CHARACTERSTIC	**ORIGINAL**	**MEASURED**
Carrier Frequency (f_c)	1000 MHz	1000 MHz
Bandwidth (B)	1750 MHz	1750 MHz
Code Rate (R_c)	62.5 MHz	62 MHz
Code Period (T)	16 nSec	16.12 nSec

4.1.3 PT1_1000000_7000000_2_4_-6

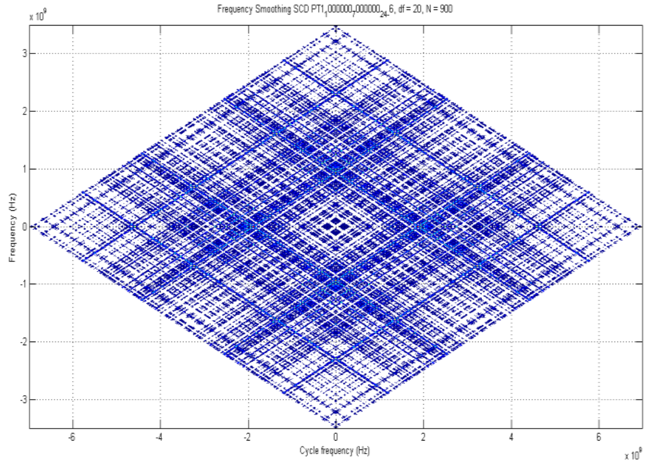

Fig 4.7: SCD patterns for the T1 code with fc = 1 GHz, and T = 16 nSec

Table-3: Comparison between measured and original Characteristics for signal PT1_1000000_7000000_2_4_-6

FEATURE EXTRACTION FOR PT1_1000000_7000000_2_4_-6		
CHARACTERSTIC	**ORIGINAL**	**MEASURED**
Carrier Frequency (f_c)	1000 MHz	-
Bandwidth (B)	1750 MHz	-
Code Rate (R_c)	62.5 MHz	-
Code Period (T)	16 nSec	-

Table-4: Comaparison between generated and extracted paprameters with sampling frequency fs=3GHz for various SNR conditions.

SNR	**High frequency analysis of T1(2) for F_s = 3 GHz**							
	F_c (MHz)		**BW (MHz)**		**R_c (MHz)**		**T (n Sec)**	
	IN	OUT	IN	OUT	IN	OUT	IN	OUT
Signal only	1000	1000	1750	1300	62.5	62	16	16.12
0 dB	1000	1050	1750	1100	62.5	62	16	16.12
-2 dB	1000	950	1750	1050	62.5	65	16	15.38
-4 dB	1000	1000	1750	970	62.5	-	16	-
-6 dB	1000	1100	1750	980	62.5	-	16	-

Table-5: Comaparison between generated and extracted paprameters with sampling frequency fs=5GHz for various SNR conditions.

SNR	High frequency analysis of T1(2) for F_s = 5 GHz							
	F_c (MHz)		BW (MHz)		R_c (MHz)		T (n Sec)	
	IN	OUT	IN	OUT	IN	OUT	IN	OUT
Signal only	1000	1000	1750	1750	62.5	62.5	16	16
0 dB	1000	900	1750	1700	62.5	62	16	16.12
-2 dB	1000	900	1750	1500	62.5	62	16	16.12
-4 dB	1000	1050	1750	900	62.5	62	16	16.12
-6 dB	1000	950	1750	1050	62.5	65	16	15.38

Table-6: Comaparison between generated and extracted paprameters with sampling frequency fs=7GHz for various SNR conditions.

SNR	High frequency analysis of T1(2) for F_s = 7 GHz							
	F_c (MHz)		BW (MHz)		R_c (MHz)		T (n Sec)	
	IN	OUT	IN	OUT	IN	OUT	IN	OUT
Signal only	1000	1000	1750	1750	62.5	62.5	16	16
0 dB	1000	1000	1750	1750	62.5	62	16	16.12
-2 dB	1000	1000	1750	1750	62.5	62	16	16.12
-4 dB	1000	1000	1750	1500	62.5	62	16	16.12
-6 dB	1000	-	1750	-	62.5	-	16	-

5. CONCLUSION

The goal of this work is to implement two cyclostationary processing techniques (Time and Frequency-Smoothing algorithms). The FAM and DFSM methods will give results which are agree well with the actual values. The DFSM took more time to execute than FAM. In DFSM as we use rectangular window the output plot does not have the resolution to calculate the output where as in FAM we use hamming window and the number of values can be truncated and the FFT is used to reduce computations. So for long signals and high data values and for fast computation FAM method is prefferable than DFSM. Extraction of parameters of LPI Radar signal is easier at various sampling frequencies till sampling frequency is considered three times of carrier. It is possible to extract all parameters till SNR =-6 dB, extraction of Code-Rate is difficult under low SNR =-6 dB but it is possible to extract carrier frequency and bandwidth.

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BIOGRAPHIES

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