EFFICIENT DEMODULATION METHOD FOR FMCW RADAR SYSTEM

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Abstract

An efficient demodulation method designed for high frequency surface wave radar system is presented, which is a modified SRC (sampling Rate Conversion) algorithm without any anti-aliasing filter used inside. This paper demonstrates the new method by analyzing the structure of FMCW (Frequency-modulated continuous wave) radar and the computing complexity compared with the traditional methods is given. The algorithm has been implemented in our newly designed HF radar system.

1 Introduction

HFSWR (High Frequency Surface Wave Radar) is a promising technology of all-weather and reliable surveillance for both remote sensing of sea surface and vessels in excess of the visible horizon [1]. Although HF radar system has been exploited for decades, such radar still presents great challenges to designers, there is a great demand for highly integration, and the hardware must be configurable and programmable based on the idea of Software Defined Radio (SDR). In many wireless communication applications, it is required to change the sampling rate by a certain ratio [2]. Programmable digital down-converters have been widely used to isolate the desired channel from signal spectrum, and the programmable part of decimation structure includes FIR filter, comb filter and so on, which are quite popular in software-defined radio system, and the SRC turns out to be mainly a problem of designing appropriate filters (anti-imaging and anti-aliasing), the main concern is not decimation but anti-aliasing operation [3, 4]. In most cases, researchers are trying to pursuit optimal methods for the filters with better magnitude and phase response [5]. But in this paper, a new structure is proposed to process SRC and demodulation, where no filter is used. Firstly, the signal processing of FMCW radar is briefly introduced.

2 Signal processing in FMCW Radar

Frequency-Modulated Continuous Wave mechanism has been widely used in modern radar system. Both transmitter and receiver are left on for expanded periods and optimum processing leads to estimates of range and velocity with greatly improved signal-to-noise ratio [6]. In each repetition period, the waveform parameter of the mechanism is described as follows:

$$x_T(t) = e^{j(2\pi f_c t + \pi \alpha t^2)}, \quad -T_s/2 < t \le T_s/2 \quad (1)$$

Where f_c is the radar carrier frequency, α is the sweep rate and T_s is the sweep repetition period, the received signal is a delayed (time t_d) and scaled (factor A) replica of the transmit signal:

$$x_R(t) = Ae^{j(2\pi f_c (t-t_d) + \pi \alpha (t-t_d)^2)} \quad (2)$$

The time delay t_d can be extracted from $x_T(t)$ and $x_R(t)$ by $x(t) = x_R(t) \otimes x_T(t)$, which is a simple function of the target's range and velocity:

$$t_d(p,t,v) = F[x(t)] = \frac{2R(p,t,v)}{c} = \frac{2}{c}[R_o + v(pT_r + t)] \quad (3)$$

Where R_0 is the initial target range, v is the velocity, c is the speed of the light and p is the sweep number [6]. Figure 1 gives the sketch of transmit and receive signal of FMCW system.

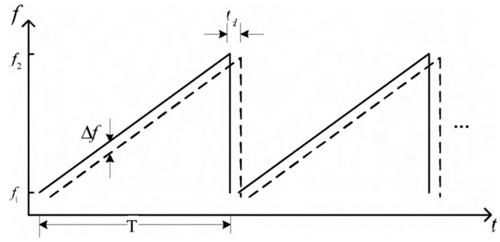

Fig. 1 Sketch of transmit and receive signal of FMCW.

As shown by D. E. Barrick, etc [6,7,8,9], the frequency of the demodulated signal f_τ is approximately proportional to the target's delay time τ_o by $\Delta f = \alpha \tau_o$. The range of specific targets can be extracted from the frequency of demodulated signal $x(t)$. ADC is used to generate the discrete sequence $x[n]$, and according to the sampling theory, in normal situation, if sampling rate f_s is much higher than the band of desired signal ($0 \sim \Delta f_{\max}$), the higher Signal-to-

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Noise Ratio can be achieved [3]. In HF radar measurement, the corresponding maximum frequency offset $\Delta f_{\max}$ (calculated from the detecting physical limitation) is only several hundred Hz [6]. But the HF radar system operates in high frequency environment. The noise outside the receiver is dominant other than noise inside. So we set an appropriate sampling rate according to noise level and the SNR which is desired for further signal processing.

The radar is a multi-state system. It can not only receive the backscatter echoes, but also acquire the noise outside the system between each repetition period, That is to say, the sampling clock for x(t) doesn't exist all the time, the sampled sequence is not continuous, and even more, the system can be configured as a multi-frequency radar system(with Time-Division Multiplexing method). While operating in different frequency, we can use different sampling clock, so we can get the limited-length x1[n] and x2[n] etc. if we take use of the traditional SRC method, an L-length filter h[n] is utilized to finish the convolution calculation with N-length sequence x[n], the length of result is L+N-1, and zeros need to be added for the convolution, the radar system is a phase-coherent system and phase information is very strictly controlled. If the length $N >> L$, this influence can be ignored, which is a normal situation of continuous sampling structure. But in some periodic transmit and receive systems like radar and sonar etc, filters may not be the best solution for the limited-length sampled sequence.

3 Principle of the Novel Algorithm

As we discussed above, the frequency information can be extracted from the Fourier transform (FT) of x[n]. $X[k]$ may be given by

$$X[k] = DFT\{x[n]\} = \sum_{r=0}^{N-1} x[n]\, e^{-j2\pi(kn/N)} \quad (4)$$

Where $W_N = e^{-j2\pi/N}$, N points are sampled in each repetition period. An evaluation of (4) using Fourier transform (FT), would result in N points. But according to the maximum offset frequency related to the maximum detecting range, we just need M points of FT result ($k = [1,2,...,M]$, $M << N, Q = N/M$). If the processor is very powerful and with enough storage memory, surely, it can process the whole sequence with once N-Point FFT calculation. But usually, the system needs to process multi-channel received signals and some further processing including detecting and tracking the targets. So the efficiency of the processing must be considered.

The new structure is described as follows: we expand (4) and get (5)

And we calculate the sum of each column, for example, the sum of qth column:

$$\begin{aligned} X_q(k) &= x(q)W_N^{k\times q} + x(Q+q)W_N^{k\times(Q+q)} + ... + x((M-1)\times Q+q)W_N^{k\times((M-1)\times Q+q)} \\ &= \sum_{m=0}^{M-1} x(m\times Q+q)W_N^{k\times(m\times Q+q)} \\ &= \sum_{m=0}^{M-1} x(m\times Q+q)W_N^{k\times(m\times Q)}W_N^{k\times q} \end{aligned} \quad (6)$$

According to the Fourier transform formula:

$$X_q'(k) = DFT\{x[m\times Q+q]\} = \sum_{m=0}^{M-1} x(m\times Q+q)W_M^{km} \quad (7)$$

And $W_M^{km} = e^{-j2\pi km/M} = e^{-j2\pi km/(N/Q)} = W_N^{k\times(m\times Q)}$ (8)

From formula (6), (7) and (8), we will get

$$\begin{cases} X_q(k) = X_q'(k)W_N^{k\times q} \\ X_0'' = \sum_{q=0}^{Q-1} X_q(k) = \sum_{q=0}^{Q-1} X_q'(k)\, W_N^{k\times q} = X(k) \quad \{k=1,2,...,M\} \end{cases} \quad (9)$$

If we separate the result of total FT of $x[n]$ into Q segments, we get

$$\begin{cases} X_0'' = X(k) \quad \{k=1,2,...,M\},... \\ X_p'' = X(k) \quad \{k = p\times M+1, p\times M+2,...,(p+1)\times M\},...., \\ X_{Q-1}'' = X(k) \quad \{k=(Q-1)\times M+1, (Q-1)\times M+2,...,Q\times M\} \end{cases}$$
(10)

Where p is a value of $[0,1,2,...,Q-1]$, the Factor is defined as $W_N^{k\times q}(q=0,1,...,Q-1; k=1,2,...,M)$, and the output is X_0'' .If we define

$$W_N^{k\times q}(q=0,1,...,Q-1;\ k=p\times M+1, p\times M+2,...,(p+1)M) \quad (11)$$

Then we can get X_p'' , which is the pth segment of the total FT (as shown in (10)).

Fig 2 shows the block diagram of the principle of the proposed structure. X_0'' is the result of the SRC and it is also the result of the demodulation. So the N-point Fourier transform of sequence $x(n)$ can be replaced by the sum of a relative shorter M-point FFT which will be multiplied by a series of factor $W_N^{k\times q}$. So the ratio of SRC is $Q = N/M$ and

$$\begin{aligned} X(k) &= \sum_{r=0}^{N-1} x[n]\, W_N^{k\times r} \\ &= x(0)W_N^{k\times 0} + x(1)W_N^{k\times 1} + ... + x(Q-1)W_N^{k\times(Q-1)} \\ &+ x(Q)W_N^{k\times Q} + x(Q+1)W_N^{k\times(Q+1)} + ... + x(2Q-1)W_N^{k\times(2Q-1)} +, ..., \\ &+ x((M-1)\times Q)W_N^{k\times(M-1)\times Q} + x((M-1)\times Q+1)W_N^{k\times((M-1)\times Q+1)} + ... + x(M\times Q-1)W_N^{k\times(M\times Q-1)} \end{aligned} \quad (5)$$

the analytic frequency band is converted to Fs/Q, which meets the need of the echo signal processing as analyzed in section 2. If p is defined as some certain value $\left(p \neq 0\right)$, the structure seems like a "Band-Pass" filter, the corresponding frequency band can be extracted from the original signal. And also if we set $p = p1, p2$, that is to say, we define the different factor $W_N^{k \times q}$ and multiplied by the result of M-point FFT simultaneously to get multi-channel output X_{p1}'', X_{p2}'' and so on. As shown in Fig2, the dashed line means that we can add another more output channels if need.

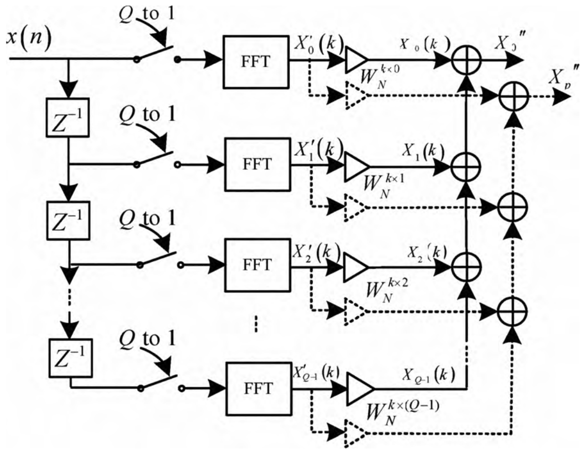

Fig.2 Structure of the proposed demodulation method

4 System test and Complexity Comparison

In our HF radar system, we sample N (N =16384) points in each sweep repetition period with sampling rate f_s=60 KHz ($T_s = N/f_s$). Because during our field test, the echo is not so strong to demonstrate the algorithm, we set a simulated target echo by delaying and attenuating the transmit signal, feed it back into the receiver, the simulated target echo (FFT result, as shown in Fig3.a) appears at the 80th point.

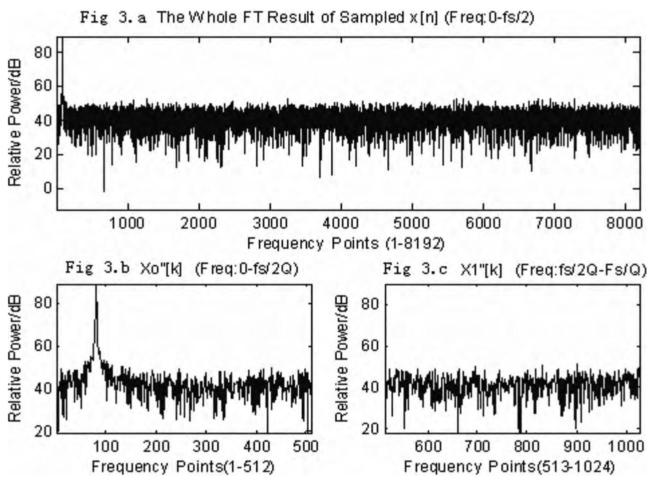

Fig.3 Results Processed by the New Method

The offset frequency is 300Hz ($\Delta f = 80 \times \left(1/Ts\right)$), but the bandwidth of sampled data is $f_s/2$ ($f_s/2 = 30\,kHz$). After processing with new structure, we get 512 points of the first segment (M=512, Q = N/M =32), and the analytic frequency band is decreased by Q times. $f_s' = f_s/2Q$, as shown in Fig3.b. And if we set $p = 1$, so in factor $W_N^{k \times q}$, $k = [M+1, M+2, ..., 2M]$, we can also get the second segment(Fig 3.c).

The proposed algorithm is more efficient than the whole FFT calculation, the computing complexity of FFT is $N \log_2\left(N\right)$,while the complexity of proposed structure is $N \times \left(1 + \log_2\left(M\right)\right)$ for calculating X_O'' , During the calculation in DSP or FPGA, FFT calculating block and parallel technique are used, so Q channels M-point FFT can be processed simultaneously, less time is used for the processing in reality.

And we also tested the traditional methods, which utilizes anti-aliasing filter and decimation method. We select the filters with the linear phase response, because the radar system takes use of the echo's phase information to derive the location and movement information of targets. We compare the performance of the proposed method with FIR, HB and CIC filters by analyzing the computing complexity and the magnitude and phase response of them. As analyzed in section II, the filter method is not very suitable for limited length sequence, but still we can analyze the complexity of these methods. Table I gives the computation complexity comparison of these methods. The FIR filter designed based on optimum equiripple approximation method and Parks-McClellan method is used to compute the parameters of the filter [10]. FIR and computation complexity of FFT, the computation complexity comparison of FIR+FFT, two-stage filters decimation structure, and HB+CIC+FFT structure and the proposed structure. We take the number of multiplication for the comparison. As we can see from Table 1, the proposed method is a relative efficient demodulation method. Although CIC filter is also with less complexity, it is with the characteristic of the stop-band attenuation and high bandwidth gain [11].

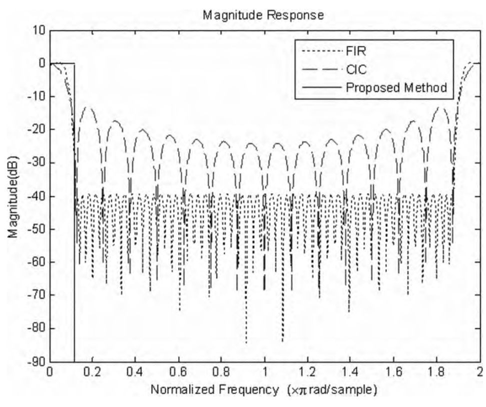

Fig.4 Comparison of Magnitude Response

Fig 4 and Fig 5 illustrate the typical magnitude and phase response of FIR and CIC filters, actually the novel

Table 1 calculating Complexity Comparison (N is the length of sequence, $L=2^{14}$, N>>L, M =N/Q)

Demodulation Method	Decimation Ratio (Q)	Parameters of filters	Filter Length (L)	Calculating Complexity	Number of multiplication
FIR+ FFT	32	$\delta_p = \delta_s = 0.01, \Delta f = 800Hz, fs = 60KHz$	147	$N \times L + M \log 2(M)$	2413056
FIR1+FIR2+FFT	4*8	$\delta_p = \delta_s = 0.01, \Delta f = 2800Hz, fs = 60KHz$ $\delta_p = \delta_s = 0.01, \Delta f = 800Hz, fs = 15KHz$	L1=45 L2=40	$N \times L1 + (N/Q1) \times L2$ $+M \log 2(M)$	905728
HB+CIC+FFT	4*8	$\delta = 0.001, \alpha = 0.02$ $(KAISER)$ * /	L1 = 7* /	$N \times L1 + (N/2) \times L1$ $+M \log 2(M)$	176640
PROPOSED STRUCTURE	32	/	/	$N \log 2(M) + N$	163840

* parameter of HB filter

method has no magnitude or phase response, according to analyses above, we can draw the magnitude and phase response similarly to the traditional methods. From the figure, we can find that the magnitude response of the novel method just like an ideal complex low-pass filter. And if we define p, it can be configured as an ideal band-pass filter. The rectangular means the algorithm just only separate the desired signal from spectrum. And the values of the original data will not be shaped by the filter.

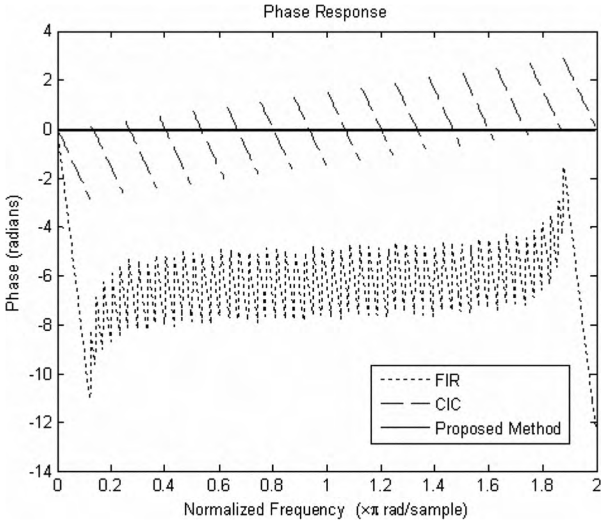

Fig 5 Comparison of Phase Response

Conclusion

This paper describes a new structure to extract the desired frequency band from the sampled data, which is an efficient demodulation method in some periodic transceivers like radar and sonar systems, the sequence sampled from the echo should be processed before the next sequence is generated. So we can take use of the sampling time for next sequence as the processing time for the demodulation and further processing. And these systems usually take use of the Doppler frequency shift of echo to get the information of the targets. The sampling rate conversion technique is often used in wireless system as an important application of Software Defined Radio technique. From the analyses of the sampling structure and the comparison with the traditional SRC methods, this paper introduces a better solution with a high efficiency and performance.

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