



Digital down Converter for Pulse Radar Receiver

Ganesh P. Sonar¹, Prof R.A.Patil², Mr J.S. Pillai³

M Tech IInd Year, Department of Electronics Engineering, Veermata Jijabai Technological Institute Mumbai, India¹

Assistant Professor, Department of Electronics Engineering, Veermata Jijabai Technological Institute, Mumbai, India²

Scientist E, Department of atmospheric and radar Society for Microwave Electronic Engineering & Research, Mumbai, India³

ABSTRACT: In Pulse radar receiver system Digital down converter is an important part. Digital down converter (DDC) demodulates the base band signal from the transmitted signal and also reduces the sampling rate of the signal. The most important part of DDC is filter structure, it uses large resources which increases the cost of the system. To overcome this problem we propose the new filter structure containing the low pass FIR Half-band filter. The proposed filter structure of DDC reduces the complexity and cost of implementation. The simulation results shows that the proposed filter structure of DDC achieves expected specification in application of pulsed radar receiver.

KEYWORDS: Digital Down Converter (DDC); Half Band Filter (HB filter); Analog To Digital Converter (ADC)

I. INTRODUCTION

DDC demodulates the signal and reduces its sampling rate. Input of DDC comes from ADC. DDC cuts down the cost of ADC and CPU in a system and also reduces the complexity of the system. If the cost of the system is reduced, the efficiency increases. It makes the most flexible structure of DDC in application of pulsed radar receiver.

This application requires both amplitude and phase of the signal. This signal acquires high resolution and high speed ADC [1]. The information content does not occupy the entire Nyquist band of the ADC. For this case sample rate is reduced, only after the frequency shifting of the signal the band of interest down to DC [2]. It preserves the magnitude and phase of the signal along with the frequency of the signal both above and below of the target band. It also separates the component of the signal by 90 degree out of phase which must be retained. This signal is referred as in-phase (I) and quadrature (Q), which are complex in nature i.e. having real and imaginary components.

Digital down converter mainly consists of the following structure- software based numerically control oscillator (NCO), mixer, filter bank and decimation. Filter bank contains low pass Half-band filter.

The paper is organized as follows. Section II related work Section III presents the design of DDC structure, a modified filter bank, and builds a LABVIEW model. Section IV LABVIEW programming based on mathematical model. Section V shows the simulation and analysis results. Section VI the conclusion of this paper.

II. RELATED WORK

[1] Merrill Skolnik, He gave an idea about "Introduction to RADAR systems". [2] Hyung-jung kim, jin-up Kim, jae - hyung kim hong mei Wang and in-sung lee all these proposed a knowledge on "The Design Method and Performance Analysis of RF Subsampling Frontend for SDR/CR Receivers." [3] In this project we have discussed about DDC which is a fundamental part of many communication systems. This allows a signal to be shifted from its carrier frequency down to base band. The incoming analog signal is converted to digital samples by ADC DDC performs the necessary translation to convert the high frequency input signal down to base band signal. Then it is transferred to signal processing units for processing. In this project we have proposed a CHIRP signal using MATLAB code, and the corresponding. DDS generates Sine and Cosine signal with center Frequency of 70Mhz. The mixer multiplies input sample coefficients stored in Block RAM with cosine and sine Signal to generate I and Q signal respectively.

International Journal of Innovative Research in Computer and Communication Engineering

(An ISO 3297: 2007 Certified Organization)

Vol. 3, Issue 5, May 2015

III. DIGITAL DOWN CONVERTER

Function of DDC is to extract the baseband signal from the modulated signal at high sampling rate. And it shifts IF spectrum to the baseband spectrum and decimate the sampling rate by the factor 1/12. In this section, a structure of DDC is designed, and a LABVIEW model is built.

A. DDC Structure

Structure of DDC mainly consists of the two channel I (n) and Q (n) as shown in fig.1. It consists of - mixer, filter bank and decimation factor. RF mixer does multiplication of two signal [3]. Software based NCO generates two complex sequences of signal having same frequency 38 MHz but different phase 0 and 90 degree respectively. This frequency is called center frequency or zero frequency. Multiplication of real signal and complex sequence signal does shifting of IF spectrum into the baseband spectrum by using the mixer. After shifting the signal it needs filtering which removes sideband, imaging, and aliasing of signal. Sampling rate is reduced by decimation. Next section is modified filter bank which reduces the complexity of signal and make it flexible, efficient structure of DDC.

TABLE-1 PARAMETERS OF DDC

Parameter type	value
Sample rate input	250MHz
Sample rate output	20.83MHz
Down sample factor	12
Roll off	0.01db
Stop band	100db
i/p frequency carrier+ baseband	212001000 Hz
o/p frequency of base band	1000Hz

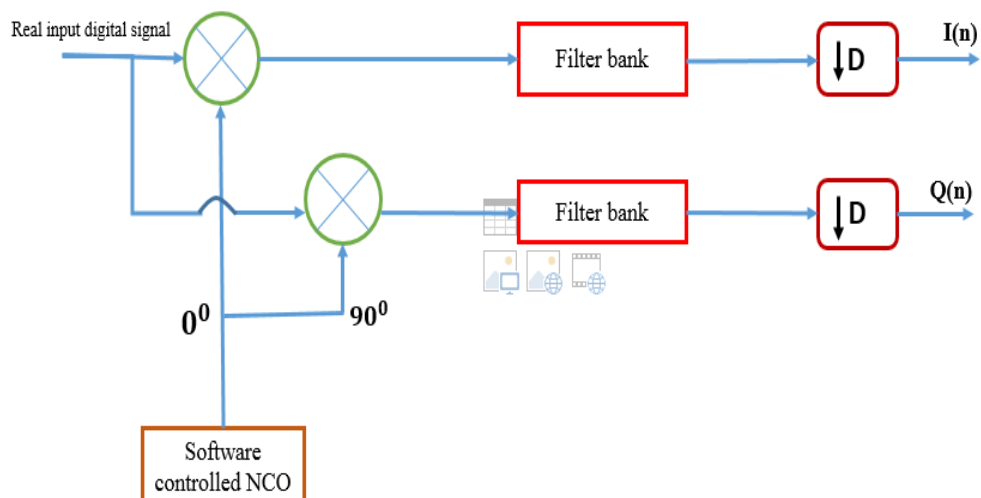


Figure 1. Structure of DDC

International Journal of Innovative Research in Computer and Communication Engineering

(An ISO 3297: 2007 Certified Organization)

Vol. 3, Issue 5, May 2015

B. Modified Filter Bank

In this paper modified filter bank contains 17 –order half band filter. Filters work on the principle of multi-rate signal processing.



Figure 2. Modified filter structure

a. Half Band Filter:

HB filter is FIR filter, as its transition region is centered at one quarter of sampling rate or $f_s/4$. The end of pass band and begin of stop band are equally spaced. Half band filter is used as decimation filter because half of the coefficient are zero in time domain. It reduces computation for filtering. Half band filter is M^{th} band filter when $M=2$, which satisfies the following equation:

$$H(2n+k) = \begin{cases} c, & n = 0 \\ 0, & n \neq 0 \end{cases}, \quad c \text{ is mostly } 0.5$$

IV. LABVIEW PROGRAMMING BASED ON MATHEMATICAL MODEL

Input of DDC comes from ADC, which is a modulated signal having the frequency $w_1 = 212$ MHz carrier and the baseband signal having frequency $w_2 = 1$ kHz and sampling frequency is $f_s = 2000$ MHz. The number of sample is 2000000.

$$X(t) = \sin w_1 t + \sin w_2 t \quad X(n) = \sin \theta_1 n + \sin \theta_2 n$$

$$X(n) = 2 \sin \left(\frac{\theta_1 + \theta_2}{2} n \right) \cos \left(\frac{\theta_1 - \theta_2}{2} n \right) \dots \dots \dots (1)$$

Resampled by $f_s = 250$ Mhz. In equation $1 \frac{\theta_1 + \theta_2}{2} = \theta_3, \frac{\theta_1 - \theta_2}{2} = \theta_4$

$$X(n) = 2 \sin \left(\frac{\theta_3}{f_s} n \right) \cos \left(\frac{\theta_4}{f_s} n \right) \quad X(n) = 2 \sin \Omega_1 n \cdot \cos \Omega_2 n \dots \dots (2)$$

NCO generates two signal at frequency $\Omega_3 = 38$ MHz and the phase is 0 and 90 degree respectively. Mix the input signal and the complex sequence generated by NCO. This will result in the IF band shifting to baseband spectrum [5].

$$X_Q(n) = I^{\text{st}} \text{ Q signal} = X(n) \cdot \sin \Omega_3 n \dots \dots \dots (3)$$

$$X_I(n) = I^{\text{st}} \text{ I signal} = X(n) \cdot \sin(\Omega_3 + 90)n \dots \dots \dots (4)$$

$$= X(n) \cdot \cos \Omega_3 n \dots \dots \dots (5)$$

Half-band low pass filter and the corresponding orthogonal o/p signal are described-

$$I(n) = \sum_{k=0}^N x_1(n-k) \cdot h(k)$$



International Journal of Innovative Research in Computer and Communication Engineering

(An ISO 3297: 2007 Certified Organization)

Vol. 3, Issue 5, May 2015

$$Q(n) = \sum_{k=0}^N x_Q(n - k) \cdot h(k)$$

$$I(n) = \sum_{k=0}^{N/2} x_I(n - 2k) \cdot \cos\left(\pi \frac{(n - N/2)}{2}\right) \cdot h(2k) + 0.5x_I(n - N/2) \cdot \cos\left(\pi \frac{(n - N/2)}{2}\right) \dots \dots \dots (6)$$

$$Q(n) = \sum_{k=0}^{N/2} x_Q(n - 2k) \cdot \sin\left(\pi \frac{(n - N/2)}{2}\right) \cdot h(2k) + 0.5x_Q(n - N/2) \cdot \sin\left(\pi \frac{(n - N/2)}{2}\right) \dots \dots \dots (7)$$

When n = 0, 1, 2

Then we discuss (6) & (7) according to two situation of variable n. when variable n is even (6) & (7) be rewritten as follow-

$$I(2b) = \sum_{k=0}^{N/2} x_I(2b - 2k) \cdot \cos(\pi(b - k)) \cdot h(2k) \dots (8)$$

$$Q(2b) = 0.5x_Q(2b - 2m - 1) \cdot \sin(-\pi(b - m - 0.5)) \dots (9)$$

When n is odd –

$$I(2b + 1) = 0.5x_I(2b - 2m) \cdot \cos(\pi(b - m)) \dots (10)$$

$$Q(2b + 1) = \sum_{k=0}^{N/2} x_Q(2b - 2k + 1) \cdot \sin(-\pi(b - k + 0.5)) \cdot h(2k) \dots (11)$$

For N-order FIR filter, when N is even the symmetrical characteristics can be described as follow-

$$h(n) = h(N - n), \quad n = 0,1,2,3, \dots \dots n$$

According to equation, we can know that the order of half-band low pass filter is even, equation (8) & (11) can be rewritten once again according to the symmetrical characteristics of filter tap coefficient.

When variable n is even (8) & (9) can be rewritten as follow-

$$I(2b) = \sum_{k=0}^m [x_I(2b - 2k) \cdot \cos \varphi_1 \cdot h(2k) + x_I(2b + 2k - N) \cdot \cos \varphi_2] \cdot h(2k) \dots (12)$$

$$Q(2b) = 0.5x_Q(2b - 2m - 1) \cdot \sin(-\pi(b - m - 0.5)) \dots (13)$$

$$\text{When } \varphi_1 = \pi(b - k), \varphi_2 = \pi(b - k - N/2)$$

When the variable n is odd

$$I(2b + 1) = 0.5x_I(2b - 2m) \cdot \cos(\pi(b - m)) \dots (14)$$

$$Q(2b + 1) = \sum_{k=0}^m [x_Q(2b - 2k + 1) \cdot \sin \varphi_1 + x_Q(2b + 2k - N + 1)] \cdot h(2k) \dots (15)$$

$$\text{Where } \varphi_1 = \pi(b - k + 0.5), \varphi_2 = \pi(b - k - N/2 + 0.5)$$

International Journal of Innovative Research in Computer and Communication Engineering

(An ISO 3297: 2007 Certified Organization)

Vol. 3, Issue 5, May 2015

It is easy to notice that the value of sine and cosine function in equation (12) and equation (13) are merely some constant such 0,1and -1 there by the multiplication in (14) & (16) can be replaced by simple data assignment operation , reducing signal processing complexity and logical significance.

V.SIMULATION AND ANALYSIS RESULT

Input signal to DDC is 212 MHz + 1 kHz frequency of signal, Input of Digital down converter comes from the ADC. Simulated signal is shows as follow-

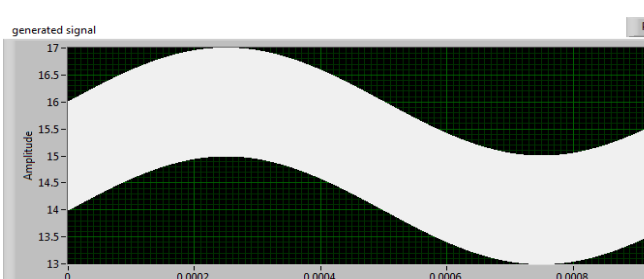


Figure 4. Simulated signal

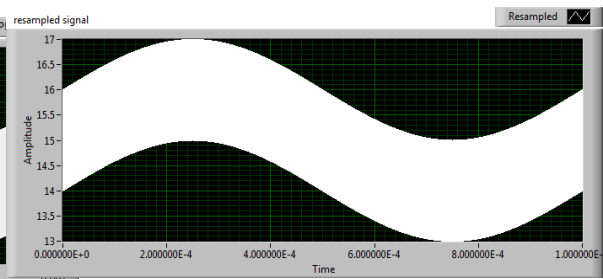


Figure.5 Resampled signal

Then the signal is resampled at the 250 MHz, because the two different frequency signals should have same start and end point of the signal as follows- Resampling signal is mixed with the sine and cosine signals. This produces first I (n) and Q (n) signal respectively and shifts the spectrum to the base band shows in fig 6 and fig.7

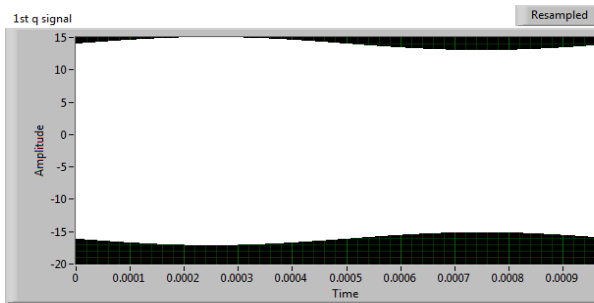


Figure 6. 1st q signal

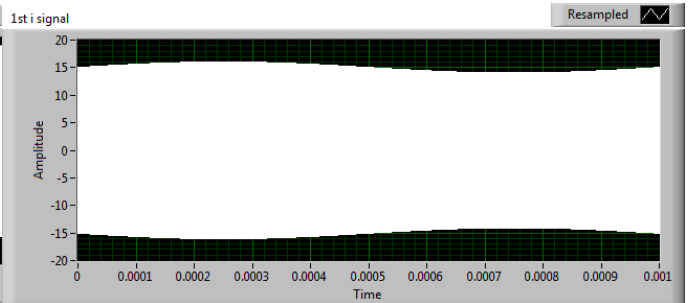


Figure 7. 1st i signal

Then signal is passed through the HB filter order is -17, roll of 0.1 and stop attention is 100 dB then find the magnitude response of the HB filter , and signal is multiplied by the coefficient of half band filter. Show in figure 8 and figure 9. Multiplied signal and filters decimated signal are show in the figure 10 and figure 11.

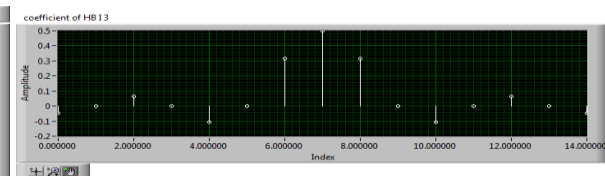
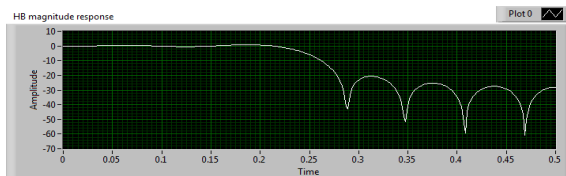


Figure -8 coefficient of half band filter, Figure -9 magnitude response half band filter

International Journal of Innovative Research in Computer and Communication Engineering

(An ISO 3297: 2007 Certified Organization)

Vol. 3, Issue 5, May 2015

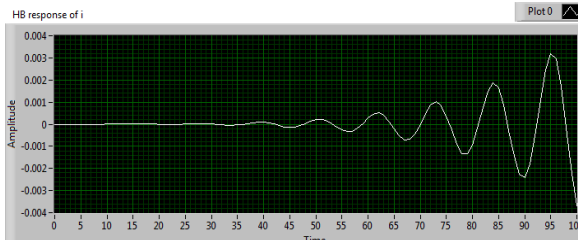


Figure-10 o/p I signal form half band signal

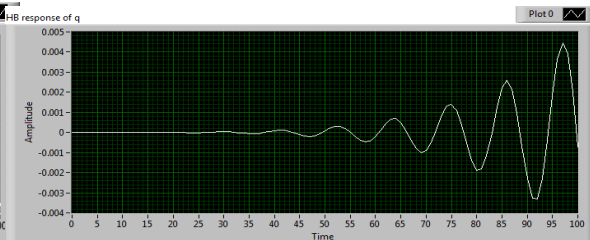


Figure -11 o/p Q signal form half band signal.

Decimating the signal by certain factor to get a decimated signal and its sampling rate is low. This is I (n) and Q (n) signal. This signal have same frequency and different phase that is 90⁰ out of phase show in figure12 and figure 13.

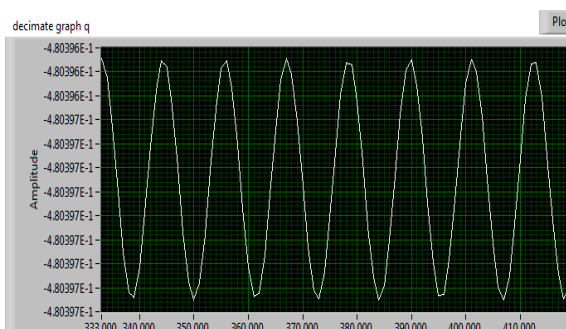


Figure-12 Q signal

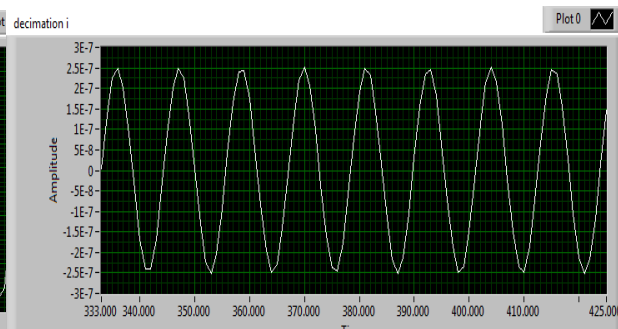
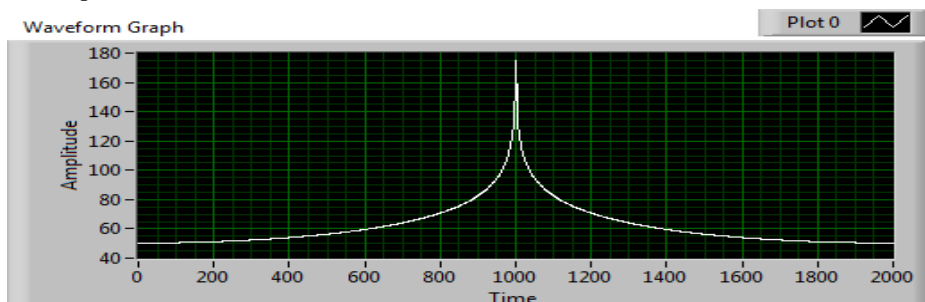


Figure 13 I signal

Particular frequency of signal and reduced sampling rate is detected by using the FFT transform technique shown by this following graph. Our input frequency is 1000Hz is detected and sampling rate is reduced by the factor 12 is detected and reduced sample rate is 20.18MHz.



V. CONCLUSION AND FUTURE WORK

In this paper we implemented DDC with new filter bank on LABVIEW for pulse radar appliances. DDC's main part is filter bank which uses the resources of a system. It comprises the low pass half band filter. In typical decimation application we need high performance filter, reasonable flat pass-band and narrow transition band, however this property are not shown by CIC filter. The pass-band of CIC roll off quickly in frequency domain. Conventionally we need to flatten the pass band with the method of following the CIC filters with compensation FIR filters, increased the bandwidth of signal of echo signal, the compensation of filter tap will decrease in HB filter due to its property, hence minimum resources compensation. Half band filter's half of the coefficients are zero, therefore the length of the filter is decreased and requires low computation for the realization of the whole operation of DDC. The modified filter bank requiring less resources as compared to the conventional filter bank, thus reducing the size, cost and complexity and hence increasing the efficiency.



ISSN(Online): 2320-9801
ISSN (Print): 2320-9798

International Journal of Innovative Research in Computer and Communication Engineering

(An ISO 3297: 2007 Certified Organization)

Vol. 3, Issue 5, May 2015

ACKNOWLEDGEMENTS

The authors acknowledge very helpful discussions and support provided by Vishnu Patil, Sankalp Govalkar and Abhishek Kodilkar (Research scientists) of ARI Division, Society for Applied Microwave Electronics Engineering and Research (SAMEER), Mumbai.

REFERENCES

1. Long Pang, "A Highly Efficient Digital Down Converter in Wide Band Digital Radar Receiver" ICSP2012 Proceedings, 2012
2. lin Zhou, qingchen zhou, jie shi, and yuxiang, " Chirp radar based on modified orthogonal transformation" IEICE electronics express, vol.8, no.20, 1736-1742
3. Henrik Ohlsson and Lars Wanhammar, "A digital down converter for a wideband radar receiver" IEEE transaction 2012.
4. Youn-Hui Jang , Hyun-IK Shin, Bum-Suk Lee, Jeong-Hwan Kim, "the loss of down converter for digital radar receiver" IEEE transaction 2012.
5. J. Alter Jeffrey O. Coleman naval, "radar digital signal processing" page no- 1 -40, march 2006.
6. By Joonam Park, SuninCNS, "A digital down converter for NI 5734" publish date-8 November 2013.

BIOGRAPHY

Ganesh sonaris is a M tech IInd year student of M Tech IInd year, Department of Electronics Engineering Veermata Jijabai Technological Institute Mumbai 400 019, INDIA. Doing the project on digital down converter for pulse radar in atmospheric and radar department in SMEER, IIT Pawai campus, MUMBAI.