Digital Filters for Radar Signal Processing

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ABSTRACT

Digital filtering is one of the most powerful tools of Radar Signal Processing. Filtering of radar signals frequently take place to realize a certain task, such as interference reduction or Doppler processing to remove clutter. In this paper a digital filter is proposed to be designed to reject the out of band interference. Higher SNR can be obtained to enhance the detection of targets inside noise if the filter is matched to the expected radar signal, even if it caused signal distortion. Doppler processing makes use of digital filters to cancel signals from fixed and slow targets. The proposed filter would be designed as digital filters to eliminate interference and clutter and blindness that caused by targets of higher speeds. Matlab simulations are made to observe features and limitations of the proposed digital filters. Windowing function is used to modify to achieve the best outcome.

Keywords- Signal, Filter, Matched, Noise, Doppler, Windowing.

1. INTRODUCTION

Digital filtering is one of the most powerful tools of DSP. Apart from the obvious advantages of virtually eliminating errors in the filter associated with passive component fluctuations over time and temperature, op amp drift (active filters),etc., digital filters are capable of performance specifications that would, at best, be extremely difficult, if not impossible, to achieve with an analog implementation. In addition, the characteristics of a digital filter can be easily changed under software control. Therefore, they are widely used in adaptive filtering applications in communications such as echo cancellation in modems, noise cancellation and speed recognition.

The rest of the paper is organized as follows: Section 2 reviews some filtering method. The design of digital filters is described in section 3. The simulation results and performance analysis are presented in section 4. Section 5 concludes the paper.

2. BACKGROUND

The purpose of surveillance radar is target detection, in addition to other tasks, such as Tracking-While Scanning (TWS). In target detection, the requirement is to detect the presence of a moving target like aircraft in the presence of unwanted signals in a reliable manner. Those unwanted signals consist of interference caused by signals produced by other nearby transmitters that could be operating in the same band as the radar transmitter itself, and the ubiquitous noise produced by electronic devices at the front end of the receiver. Other source of unwanted signals is clutter, which is radar backscatter from objects other than the target that lie in the path of the transmitted signal. Filtering of signals is the major tool used in the signal processing of radar signal returns. Most of analog filters have been replaced by digital filters, particularly in baseband. Filters are used to shape the spectrum and rejects interference, while passing the wanted echo signal with minimum distortion and realizing more desirable higher Signal-to-Noise Ratio (SNR). However, the same filter performing the task of rejection of the out-of-band interference .Here, the multifunction filter serves as a matched filter as well as a Doppler processor.

2.1 Generation of Signal

Several techniques exist when it comes to generate the transmitted signal. Signal generation can be passive or active. Fig 1 describes the passive method of producing a compressed signal. This method is defined as passive because a specific filter is used to produce the frequency modulation from a basic rectangular pulse. The radar uses digital computation. The use of digital calculation allows to store several pulses in a memory bank and to select a given pulse at any given instant according to the situation. Indeed, compressed pulses have specific characteristics that have a direct impact on the overall radar performance. For instance, pulse duration T and τ impacts the covered range or the range discriminator ; the frequency drift Δf has an impact on the Doppler filtering and can also be selected.

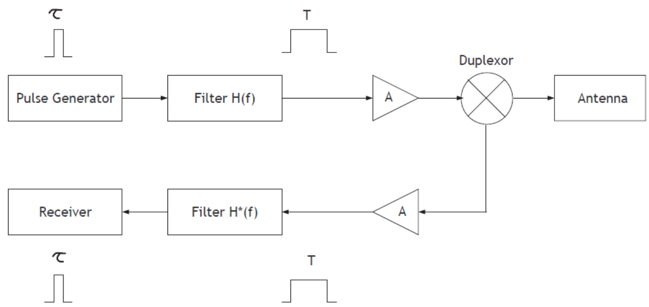

Fig-1: Pulse Compression signal generation

2.2 Matched Filter

Matched Filter (MF) is a filter that is designed according to the waveform of the transmitted radar signal. In presence of white noise only, the MF achieves in its output the maximum SNR, compared to all other linear filters [1]. It is used in pulse compression to attain very narrow pulse out of long pulse of low peak power [2]. The transfer function of a filter matched to a signal s(t) is given as the complex conjugate of the spectrum, or Fourier transform, of that signal S(f) [3], i.e.

$$H(f) = ks^*(f)e^{-j2\pi f t_0} \quad (1)$$

where k is an arbitrary scalar that depends on the gain of the filter, and to is a delay required for best sampling instant. Equivalently, the impulse response of the MF can be found in terms of the signal as follows

$$h(t) = ks^*(t - t_0) = ks(t_0 - t) \quad (2)$$

Since not all waveforms have realizable matched filters, new waveform designs are sought after. The waveform design should satisfy a number of requirements that depend on the application and the nature of noise or clutter exists. However, correlation between the transmitted and received signals can be used instead of matched filtering. When a correlator is used, the delay to is essential to be synchronized so as to obtain the maximum SNR, while it is not critical in a MF. Matched filter is active in the presence of white noise rather than colored-noise. Therefore in presence of colored-noise, a whitening filter has to be applied prior to the matched filter or correlator, in order to alter the spectrum of noise and avoid mismatch with the signal. Integration is a class of matched filtering techniques that is used in Radar Signal Processing (RSP) in order to enhance the SNR of the received echo signal. The integrator accumulates the energy of subsequent echo pulses that are reflected by the same target. On the other hand, noise signals are not correlated and thus their integrated energy tends to vanish, and hence the SNR at the output of the integrator is going to increase. Predetection integration takes place within the IF stage of the receiver, realizing better SNR compared to post-detection integration

2.2 Doppler Processing

Doppler processing of radar signals is applied to remove fixed targets and raise the Signal-to-Clutter Ratio (SCR).Currently, there are two major processing techniques to enhance the SCR in pulse radar, which are the Moving Target Indication (MTI) used in short pulse radar, and Moving Target Detection (MTD) that is used in pulse Doppler radar [4]. If the wavelength of radar signal is λ, the Doppler frequency shift in the signal received from a target moving with a relative velocity v_r (to or from the radar) is given as

$$f_d = \frac{2v_r}{\lambda} \quad (3)$$

An MTI filter is easier to be implemented and costs a little, compared to MTD processors. In baseband, an MTI filter is Highpass Filter (HPF) that filters out signals from fixed targets, which are of zero Doppler frequency, and attenuates signals come from slow targets. Unfortunately, MTI filter is either a digital filter or a discrete analog filter, and both of them have periodic frequency response that repeats each multiple of the sampling frequency, i.e. the Pulse Repetition Frequency (PRF) of the pulse radar.MTI filter acts a proper HPF only within the Doppler frequency range -PRF/2 ≤ fd ≤ PRF/2, as illustrated in Fig. 2 below, where f_d is the Doppler frequency of the echo signal, as defined in equation (3). That complies with the Nyquist rate that restricts fs ≥ 2 f_d , where f_s is the sampling frequency. For targets of higher Doppler shifts, velocity ambiguity occurs.

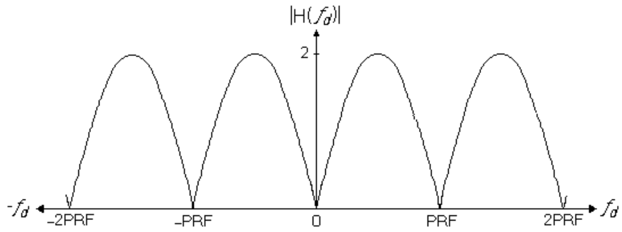

Fig.-2: Frequency Response of MTI Filter.

Consequently, MTI filter rejects some moving targets of nonzero Doppler frequencies. Those targets have frequencies multiples to the PRF and they are known to move with blind speeds, as the radar will be blind and wont detect them. The first blind speed of MTI radar is related to its PRF and wavelength as follows

$$v_B = \frac{PRF\lambda}{2} \quad (4)$$

Generally, MTI filters avoid blind speeds difficulty by using a staggered PRF, rather than using fixed PRF. Hence, the probability that a target moves with one of the blind speeds will be considerably reduced [5]. In pulse Doppler radar, a bank of filters is used to achieve larger dynamic range in the Doppler frequency domain. This procedure avoids filtering out moving targets due to blind speeds and minimizes ambiguity in the velocity measurement.

Commonly, both MTI and MTD take place in the video stages of the receiver (baseband), after the signal demodulation by means of phase detection, rather than using envelope detection. Coherent demodulation is essential so as to preserve the phase information of the received signal. However, the IF signal should be demodulated with the aid of quadrature basis in order to avoid phase angle ambiguity. It is resolved into in phase and quadrature channels using two coherent signals cos ω_ot and sin ω_ot, where ωo is the IF frequency. Each channel has its own MTI filter, whose output is orthogonally summed to that of the other channel filter [3].

2.3 Digital Filters

Digital filters are used in separation and restoration of signals. Signal separation is needed when the signal has been contaminated with interference and noise, while restoration of signal is used when a signal has been distorted in some way. All of those filters used in radar signal processing and mentioned earlier, i.e. Lowpass Filter (LPF), MF and MTI filters will be more reliable if they are implemented as digital filters. Digital filters are replacing analogue filters because of their reliability and their compatibility to Digital Signal Processing (DSP). They are available in two flavors, Finite Impulse Response (FIR) and Infinite Impulse Response (IIR) filters [6]. Although IIR filters are faster and have fewer components compared to FIR filters, they should be tested for stability per each design changes. On the other hand, FIR filter is stable by definition and has linear phase response.

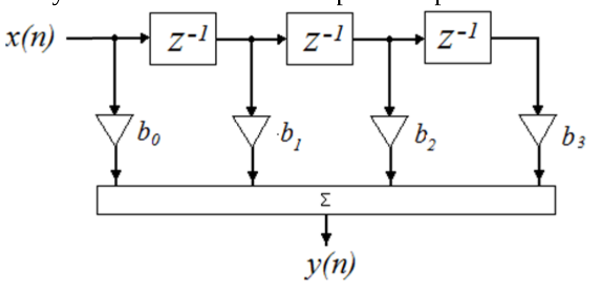

Fig-3: FIR Filter

From Fig 3 above, we can notice that the current sample of the output signal of the FIR filter y(n) is the sum of the current sample of the input signal x(n) and previous samples multiplied by the filter coefficients b_k as follows

$$y(n) = \sum_{k=0}^{N-1} b_k x(n-k) \quad (5)$$

where N is the number of taps or filter coefficients. N−1 is the filter order, which equals the number of delays marked Z^{-1}.

3. DESIGN OF DIGITAL FILTERS

The actual procedure for designing digital filters has the same fundamental elements as that for analog filters. First, the desired filter responses are characterized, and the filter parameters are calculated.

3.1 Design of FIR Filter

To design an FIR digital filter as that performs as a multifunction digital filter for radar signals.

Firstly, we are going to focus on the design of FIR filters in the frequency domain. An FIR filter is totally described by its impulse response h(n) that is formed by the filter coefficients. The design of highpass, bandpass or bandstop filter typically starts with a prototype lowpass filter, which is then converted to the required filter [10].

For better filtering, signals within the passband are required to propagate through the filter, while those signals within the stopband should be adequately blocked or at least face high attenuation. Moreover, minimum level of ripples should exist in both passband and stopband, and the transition band between them is to be as fast as possible.

The impulse response of a HPF (such as MTI filter) is formed by changing the sign of each sample of the impulse response of the prototype LPF, and adding one to the sample at the center of symmetry. This action in the time domain inverts the frequency spectrum as stated in the conversion equations below

$$H_{HPF}(e^{j\omega}) = 1 - H_{LPF}(e^{j\omega}) \quad (6)$$

$$h_{HPF}(n) = \delta(n) - h_{LPF}(n) \quad (7)$$

where H(ejω) and h(n) represents the transfer function and the impulse response, respectively. The combination of cascading the noise reduction LPF with the MTI HPF can be replaced by one BPF, which has an impulse response that is given through the convolution of the two impulse responses as follows

$$h_{BPF}(n) = h_{LPF}(n) * h_{HPF}(n) \quad (8)$$

The design of an FIR filter involves finding the coefficients h(n) that result in a frequency response that satisfies given filter specifications. For an ideal LPF, its impulse response is the sinc function, which extends to infinity. Convolving an input signal with this response provides perfect low-pass filtering. However, the sinc function should be truncated to have h(n) of a finite number of samples. Due to truncation, the frequency response of the filter involves ripples and slower transition band. This results from the discontinuity at the ends of the truncated sinc function. Increasing samples of h(n) does not reduce problems since discontinuity is of a major effect. Several windows are designed for FIR filters, so as to reduce side-lobes that occur due to rectangular windowing [6].Hamming window is given for an FIR filter of length N as follows

$$win(n) = 0.54 - 0.46\cos\left(\frac{2\pi n}{N}\right) \quad 0 \le n \le N\text{-}1 \quad (9)$$

To design a windowed FIR filter, two parameters must be selected: the cutoff frequency f_c and the length of h(n), N. The frequency is expressed as a fraction of the sampling rate fs and lies between 0 and 0.5 (the folding frequency). Value of N sets the roll-off according to an approximation of N = 4/BW, where BW is the transition bandwidth that is measured from where the curve just leaves 1 to where it almost reaches 0. The transition bandwidth as well is expressed as a fraction of the sampling frequency, between 0 and 0.5. Matlab is frequently used to simplify filter design. The FIR1 built-in function offers FIR filter design using the window method to design an Nth order

FIR digital filter of any of the four types, and returns the filter coefficients in vector b of length N+1 as follows

$$b = \text{fir1}(N,\ \omega_c, \text{'ftype'}, \text{window}) \quad (10)$$

The text ftype indicates the type of the filter, where window indicates the window used in the design of the FIR filter. The cut-off frequency ωc must be between 0 and 1.0, where 1.0 is the folding frequency. For BPF, ωc is a vector of two samples, which are the lower and the upper cutoff frequencies

3.2 Design OF Matched Filter

As any signal transmitted over the air, the chirp signal encounters noise in its two way trip from the transmitted antenna to the target and back to the antenna. In order to maximize the signal-to-noise ratio at the receiving stage, the matched filter is the optimal solution. Here we give a summarized overview. For a received signal y(t) which is a time-shifted (delayed) replica of the transmitted signal with additive noise n(t):

$y(t) = x(t - t_0) + n(t)$ (11)

The frequency response function of the linear, time-invariant filter which maximize the output peak-signal-to-mean-noise (power) ratio for a given input signal-to-noise ratio is given by:

$H(f) = Ga\ Y^*(f) \exp(-j2\pi f t_0)$ (12)

where:

- Y (f) is the Fourier transform of y(t)
- t_0: fixed value of time at which the signal is oberved to be maximum (equals the round trip delay of the transmitted signal)
- Ga: filter gain (generally set to unity)

The noise that accompanies the signal is assumed to be stationary and to have a uniform spectrum (white noise). It does not need to be Gaussian. In the time domain, using inverse Fourier transform, the matched filter impulse response can be expressed as follow using the fact that

$Y^*(f) = Y(-f)$

$$h(t) = \int_{-\infty}^{\infty} H(f) e^{j2\pi f t} dt$$

$= Ga\ y(t_0\text{-}t)$ (13)

The impulse response of the matched filter is simply the image of the received signal; that is, it is the same as the received signal but run backward in time starting from instant t_0.However, since the noise n(t) is an unknown signal, the filter is matched to the transmitted signal x(t):

$h(t) = G_a \bullet x(t_0 - t)$ (14)

The output of the matched-filter is expressed as the convolution in the time domain of the received signal with the matched-filter impulse response:

$$s(t) = \int_{-\infty}^{\infty} y(\tau(h(t-\tau)\, d\tau$$

$= z(t) + w(t)$ (15)

where z(t) represents the noise-free output from the matched filter and w(t) represents the filtered noise. Note that since the transmitted signal spans a wide frequency range, the matched filter cuts out some signal as well as some background noise since it has a limited bandwidth.

In radar signal theory, the matched filter output is affected by the Doppler effect. The apriori unknown frequency shift introduced by the Doppler effect can also be seen in the time domain as an additional def, say t_D. Therefore, since both target range and target Doppler shift are unknown to the system, a bank of matched filters is used to determine the overall delay. Doppler filtering output, i.e. a frequency shift, helps to best estimate the target range by substracting the delay t_D to the total delay.

If the noise is not white, assumes that the noise has a spectrum that is independant of frequency (constant). If this assumption is not true, then the filter that maximizes the output SNR is different. It has been showed that the matched filter frequency response in such situation could be derived as follow:

$$H(f) = \frac{1}{N_i(f)} G_a \left(\frac{Y(f)}{N_i(f)} \right)^* e^{-j2\pi f t_0} \quad (16)$$

where Ni(f) denotes the noise spectrum.

This indicates that the non-white noise matched filter can be considered as the cascade of two filters. The first filter, with frequency-response function 1/Ni(f), acts to make the noise spectrum uniform, or white. It is sometimes called the whithening filter. The second is the matched filter described by equation (14) when the input is white noise and a input signal with spectrum S(f)/Ni(f).

The output from an uniform (non-weighted) matched filter presents some characteristics that might lower the overall system performance. Indeed, the high sidelobes level are likely to hide targets. Such a situation is unaccpetable in a military environment. Weighting functions are here used to compromise between main lobe width and sidelobes relative magnitude. Numerous weighting functions exist with different impact on each parameter. They are mainly used in the frequency domain where bandwidth limitation plays an important role. These functions are used in the same manner in the time domain to lower sidelobes relative amplitude and main lobe width. It appears that radar designers have to face a dilemma because a better performance in term of main lobe width can only be achieved at the expense of higher sidelobes amplitude. Windowing functions are used to modify time domain characteristics, they are implemented using digital filtering techniques (FIR filters). These filters are placed right after the matched filter.

3.3 Design OF Multifunction Filter

Regarding the noise reduction LPF and the MTI filter, which is a HPF, they are designed in a similar manner for design of windowed FIR digital filter described earlier. The resultant BPF is not essential to be of symmetry in the lowpass and highpass parts of its response, since each part may have a different prototype LPF. The foundation for that; interference signals are characterized by different attributes from signals due to targets relative to radar. Concerning the MF, its coefficients can be assumed to be constant for regular radar. On contrary, it distorts the received echo signal waveform to realize better SNR that assists the decision of detection. Hence, if we are going to integrate that filter in the proposed multifunction filter, we should accept compromises. Therefore, we have selected that proposed filter to be a multifunction filter. Rather than using three FIR filters each of self determined coefficients, only one multifunction filter is expected to perform the whole task. Those coefficients will be determined according to the received signal from the wanted echo signal that is supposed similar to the transmitted signal. This proposal provides a solution that applies linear system, rather than searching for a nonlinear processor.

4. SIMULATION AND PERFORMANCE ANALYSIS

At the receiving stage, the signal power is very weak compared to the noise power. The incoming echo signals are first mixed into an Intermediate-Frequency box so that the signal gets shifted to a lower frequency where digital signal processing techniques are achievable. A sharp bandpass filter centered on the carrier frequency in use eliminates out of band noise. In the military domain, a frequency hopping scheme is used in order to avoid frequency jamming issues. Fig 4 shows the output of the matched filter in a noise free scenario. As noise is a complex process, it distords both signal amplitude and phase. Thus, the main lobe amplitude and position that leads to the target position are affected by the noise. The upper-left figure depicts the real part of a chirp signal (the signal is symmetric compared with t = 0 axis. We can see how the frequency increases along with the time. The upper-right picture shows the spectrum of the complex chirp signal. The lower figures show the output of the matched filter first in absolute scale and then in dB. Two important parameters has to be taken into account in the figures:

• main lobe width: it defines how accurate the range of a target can be determinated

• sidelobes relative amplitude level: it affects other targets of being shadowed by the matched-filtered output signal

Without any specific signal processing, it appears that the main lobe has an acceptable width. The more narrow the main lobe width can be, the better the accuracy of the radar. Concerning sidelobes, the Fig 4 show that they have a relatively high amplitude level (an exact value would be -13.2 dB relative to the main lobe magnitude). This is obviously a too high value if we consider that several targets can be detected at the same time. If so ,they might be hidden by the sidelobes if they are located in the same direction.

There exists some mathematical techniques that allow to modify main lobe width and sidelobes relative amplitude. These functions are filtering techniques and are called weighting functions.

Weighting functions are here used to compromise between main lobe width and sidelobes relative magnitude. Numerous weighting functions exist with different impact on each parameter. They are mainly used in the frequency domain where bandwidth limitation plays an important role. These functions are used in the same manner in the time domain to lower sidelobes relative amplitude and main lobe width. It appears that radar designers have to face a dilemma because a better performance in term of main lobe width can only be achieved at the expense of higher sidelobes amplitude. Fig 5 depicts the output of a compressed pulse matched filter. Windowing functions are used to modify time domain characteristics, they are implemented using digital filtering techniques (FIR filters). These filters are placed right after the matched filter.

We can see from Fig 5 that it is possible to achieve great performance in minimizing sidelobes amplitude. Blackman function presents almost -85 dB for its sidelobes relative level but produces an unacceptable main lobe width. Most of the modern radars implement Hamming or Hanning window functions because they provide the best trade-off between main lobe width and sidelobes relative amplitude. Depending on the situation, adaptive windowing can be performed in order to select the best weighting function.

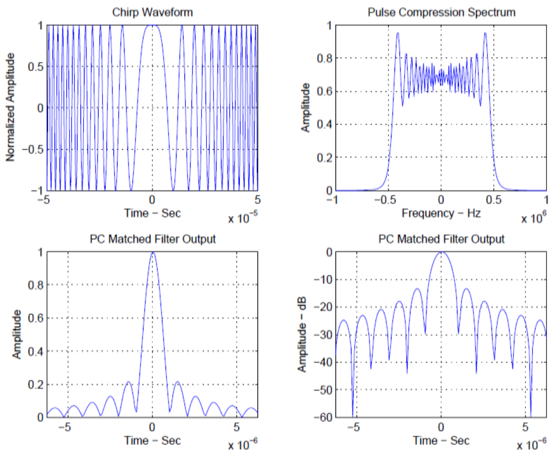

Fig-4: Pulse Compression Matched Filter Output

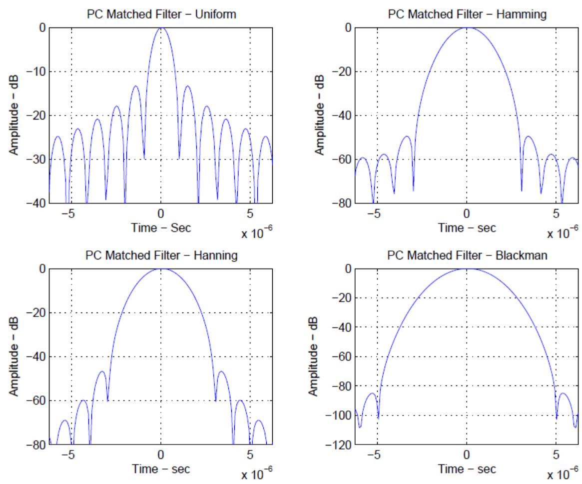

Fig-5: Windowing Function

5. CONCLUSION

In this paper, digital filters have been proposed to perform different tasks, such as filtering of the received signal to reject out-of-band interference, matched filtering and windowing. Matlab program is used in the design of the filter and analysis tools. A code written in Matlab has been designed and simulated for the digital filters.

I recommend that this simulated processing scheme to be realized and applied in a real-time digital signal processor.

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