

A Novel Design of Matched Filter for Digital Receivers



D. Likhith Reddy , S.V. Subba Rao

Abstract: In this paper, a brief review regarding introduction to the digital signal processing techniques particularly Digital Pulse Compression and Linear Frequency Modulation involved in matched filtering and some designs being used is presented. Also, the matched filter being developed is discussed by highlighting its pros and cons. The introduction of matched filter in the communication receivers has simplified the design of the system. The matched filter has improved the signal to noise ratio of the receiver system and hence has become an important element in the communication system. This paper also presents the possible challenges; the matched filter design and simulation results in MATLAB have shown satisfactory outputs of the receiver.

Keywords: Aerospace Applications, Digital Pulse Compression, Linear Frequency Modulation, Matched Filter

I. INTRODUCTION

Design of a matched filter has become very critical in the development of digital receivers used for aerospace applications. The matched filters are essential in the receivers to provide high S/N Ratio and for the rejection of unwanted clutter or echoes.

The design of matched filters involves digital signal processing. Different signal processing techniques are used in matched filters designs based on the requirements. In case of development of digital receivers used in aerospace application, digital pulse compression technique is used in the design of the matched filter. Digital Pulse Compression is an advanced technique for processing a radar data [1]. In the process of digital pulse compression, Linear Frequency Modulated (LFM) signal is used to generate the chirp waveform. Chandan Singh D. Rawat, et.al, illustrated that the Polyphase Codes P3 and P4 derived from the LFM signals have better peak side lobe level making them desirable in Radar Pulse Compression [2].

This paper focuses on matched filter design in the receiver system and discusses about Digital pulse compression technique and design of the matched filter, pros and cons of such designs, probable issues encountered during the design.

II. DIGITAL PULSE COMPRESSION

Pulse Compression was developed to overcome the requirement of extreme high peak power to achieve long range detection and to improve the signal level in the presence of clutter. Digital Pulse compression is one of the very first blocks present in the chain of signal processing algorithms for a radar receiver. The received signal at the input of any receiver is highly noisy in nature. As a result, the transmitted signal of interest is not clearly visible to the receiver in the presence of additive white Gaussian noise from the transmission channel. Pulse compression allows to achieve relatively long pulse with limited peak power, while obtaining the range resolution corresponding to a short pulse. Thin Thin Mar, et.al., presented pulse compression with LFM pulse modulation and proved that with the compression of effective pulse width, range and resolution can be enhanced [3].

The process of pulse compression involves modulation of the pulse before the transmission. The received echo is compressed into a narrow pulse using correlation technique, which facilitates long range detection. The radar that implements this technique is called pulse compression radar. In Pulse compression radars, the received signal is convoluted which is an exact replica of the transmitted signal to produce a narrow output pulse at the location when the two convolved signals are an exact match.

Correlation of time domain signals i.e., convolution of signals is equivalent to multiplication of their transforms in frequency domain. When the two convolved signals are an exact match, the frequency domain representation of the two correlated signals is also identical. Its result is totally independent of the time alignment between the two signals. So, the matched filter will respond in a unique way for each target returning regardless of when the echoes are received. Converting the product back to the time domain results in a narrow pulse [4][5], thus enabling us to estimate the target distance i.e., the range of the target. This also enhances the range resolution. The amplitude of the pulse corresponds to the Radar Cross Section (RCS) of the target whereas the delay gives an insight to the range of the target [6][7].

Different types of modulations for the pulse, such as linear or non-linear frequency modulation signals (chirp modulation) or discrete phase code modulation can be used before transmission.

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III. LINEAR FREQUENCY MODULATION

In order to perform pulse compression at the receiving end of the radar system, the transmitter shall output a modulated pulse which is considered to be linear frequency modulated pulse. Linear frequency modulation is a technique in which the frequency varies linearly. When a pulse is linear frequency modulated, the instantaneous frequency linearly varies (either increases or decreases) within the signal thereby producing chirp waveform (upward chirp or downward chirp) as illustrated in Fig. 1 [8].

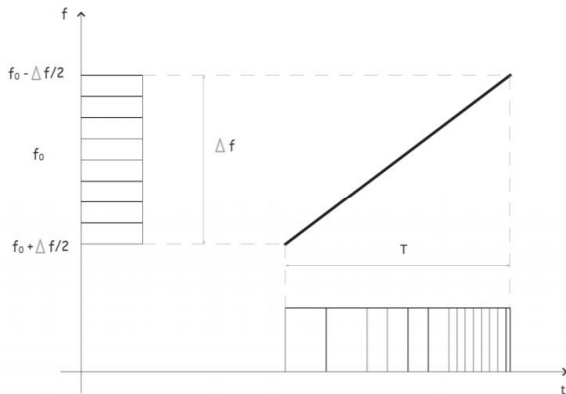


Fig. 1: Generation of LFM Signal – Basic Principle

The time domain chirp signal is given by the following equation [9].

$$f(t) = e^{j\phi(t)} \quad -- (1)$$

Here $\phi(t)$ is the instantaneous phase which is given by the equation

$$\phi(t) = 2\pi \left[\left(f_0 - \frac{\Delta f}{2} \right) t \pm \frac{\Delta f}{2T} t^2 \right] ; 0 < t < T \quad -- (2)$$

where T is the pulse duration for which the transmitter is ‘ON’ considering the duty of the transmitter is 50% i.e., transmitter is ON for 50% of the time and OFF for 50% of the time. Substituting the equation (2) in $f(t)$ and separating the real and imaginary parts, we arrive at LFM pulse. The result is shown in Fig. 2.

$$Re\{f(t)\} = \cos \phi(t)$$

$$Im\{f(t)\} = \sin \phi(t)$$

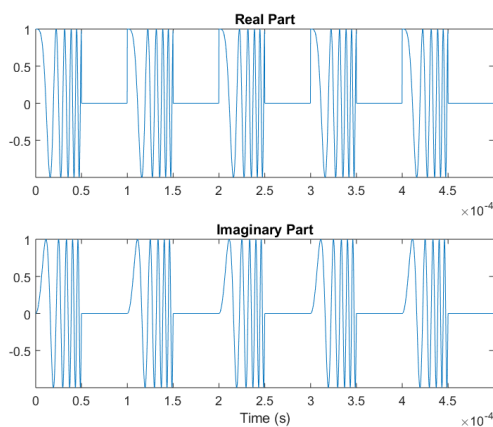


Fig. 2: Linear Frequency Modulated Pulse (Real & Imaginary parts) plotted for 5 consecutive pulses

IV. MATCHED FILTER

Any transmitted signal gets affected by noise when it travels from the antenna to the target and back to the antenna. To retrieve the known transmitted signal from the received noisy signal is where matched filter comes into picture. The matched filter is an optimal filter which maximizes the Signal to Noise Ratio (SNR) at the receiver. The received signal $r(t)$ can be assumed to be a time shifted version of the transmitted signal $x(t)$ with Additive White Gaussian Noise (AWGN) denoted by $n(t)$.

$$r(t) = x(t - t_0) + n(t) \quad -- (3)$$

The impulse response of the matched filter is simply reverse time shifted version of the input signal. However, when the transmitted pulse is symmetrical in nature, like most of the practical signals, the impulse response is same as that of the input signal and the output of the matched filter is expressed as the convolution of the received signal and impulse response of matched filter in the time domain. The convolution output is maximum when the impulse response is exactly a time delayed version of transmitted signal.

In the matched filter algorithms, Fast Fourier Transform (FFT) and Inverse Fast Fourier Transform (IFFT) are used to achieve the radar parameters as they are very fast compared to other algorithms. The matched filter is represented in Fig. 3.

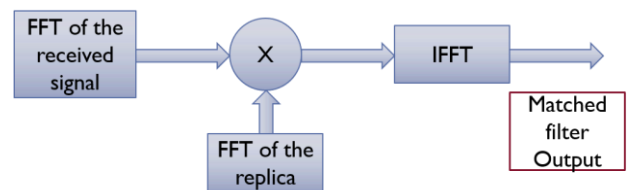


Fig. 3: Matched filter

In matched filtering technique, the system is designed to look out for the known signal (signal with similar characteristics as the transmitted signal) in the received signal. This is the main advantage of using matched filters. But to accomplish this task, the system also needs to have prior knowledge of the signal that is being watched for which it becomes one of the disadvantages. Thus, usage of matched filter in most cases is a trade-off between the receiver complexity and the knowledge of the signal. The biggest challenges in matched filter design is to correctly identify the integration period. To maximize the output Signal-to-Noise Ratio, it is best to integrate over the entire period of the signal. But, practically, the time available for identifying the target and making a decision is much smaller. Hence, integrating over the entire period will delay the result. Hence the period of integration must be chosen such that the SNR is maximized without delaying the decision making.

V. MATCHED FILTER SIMULATION RESULTS

The matched filter was simulated in 5 different ways. The first method involved is the direct method in which the FFT's of the replica of transmitted signal and the received signals are multiplied. Inverse FFT is applied to the product to obtain the matched filter output as shown in Fig. 4.

In the second method, the matched filter coefficients are obtained for the linear frequency modulated waveform and applied for the samples of the received waveform without any additive noise. The third method is similar to the previous method but the received signal was intentionally corrupted by additive white Gaussian noise during transmission. In the fourth method random noise was added to the samples and matched filtering was performed. In the fifth method, Blackman window was applied to the output of fourth method. The second and fifth methods are shown in Fig. 6 whereas the comparison of third and fourth methods is shown in Fig. 5.

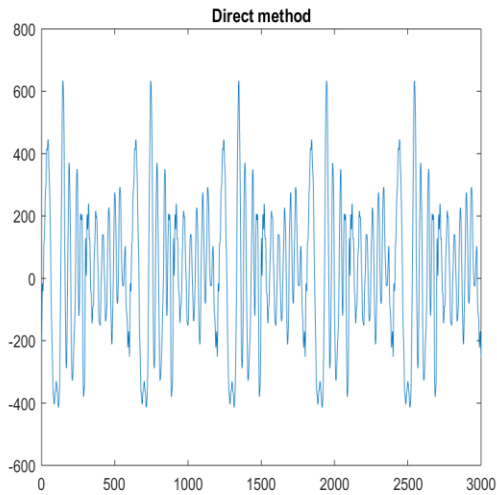


Fig. 4: Direct method of matched filtering

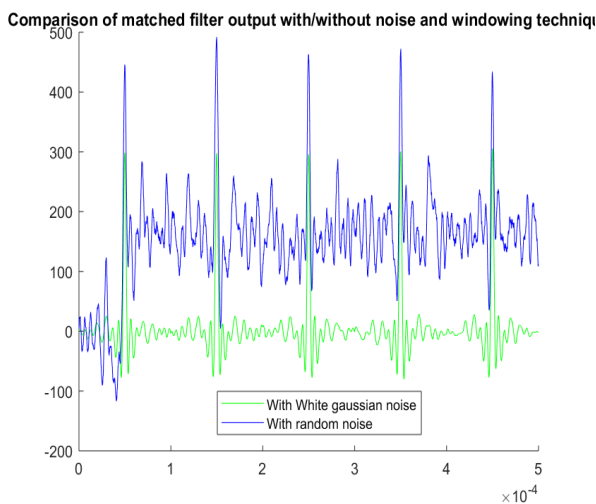


Fig. 5: Comparing matched filtering results in the presence of AWGN and Random noise

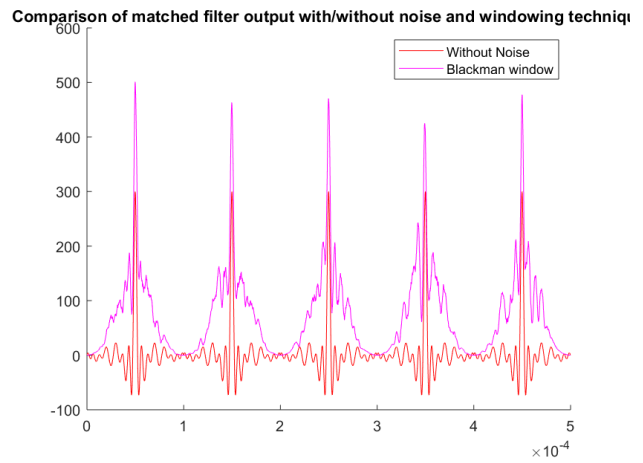


Fig. 6: Comparing matched filtering results without noise and Blackman window with random noise

By comparing the results in Fig. 5 & Fig. 6, it is established that applying Blackman windowing (pink color) technique in the process of matched filtering reduced the sidelobes than that in the case of random noise (blue color) or the direct method shown in Fig. 4. The pulse compression ratios achieved are also higher.

VI. CONCLUSION

The design of the matched filter and the different advanced signal processing techniques involved in it such as Digital Pulse compression and Linear Frequency Modulation were discussed in this paper. Matched filtering techniques have been implemented under various scenarios such as ideal conditions without noise, conditions with additive white gaussian noise, windowing technique and a combination of more than one of the previously mentioned conditions.

A comparison of the different implementation techniques is made and it is shown that the signal to noise ratio of the receiver is improved by the use of matched filter implementation through simulations in MATLAB via windowing technique even in the presence of AWGN.

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