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PULSE COMPRESSION IN MEDICAL ULTRASOUND IMAGING

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Abstract: *Pulse compression theory and techniques were developed originally for radar systems but could be carefully adopted for the ultrasound specific problems In past ten years a considerable amount of research has been done in area mismatched filtering for the purpose of it application in ultrasound. However, similarly to radar systems for distributed targets, pulse compression has not achieved a success of penetration in comercialy available systems.*

1. INTRODUCTION

Pulse compression is employed in radar (and sonar) to increase the signal energy transmitted without sacrificing range resolution, nor encountering excessively high peak powers than can cause electrical breakdowns. Therefore, pulse compression permits, decoupling useful signal bandwidth (range resolution) from the transmitted pulse length.

Modulating the transmitted pulse increases transmitted signal bandwidth. This modulation may consist of amplitude, phase, or frequency changes of signal carrier within the pulse. Target echo signal are then passed through filters matched to the transmitted signal. Therefore the energy is compressed into a pulse having a time duration T, which is approximately equal to the reciprocal of he transmitted bandwidth B (Fig. 1). The ratio of he transmitted to compress pulse length is called pulse compression ratio or TB product.

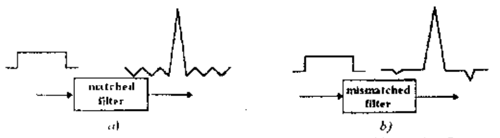

*Fig. 1 Pulse compression: **a)** Matched filter (MF); **b)** Mismatched filter (MMF).*

In medical ultrasonic imaging the peak acoustic pressure cannot be arbitrarily increased because of patient safety. FDA (Food and Drug Administration) has set maximal acoustic peak pressures in order to reduce potential risk of damaging the biological tissue. The pulse compression methods has the potential of not exceeding FDA's limit of acoustic pressure while the improving SNR (signal to noise ration) if compared to conventional system.

2. WAVEFORM SELECTION AND PROCESSING

Pulse compression theory and techniques were developed originally for radar systems but could be carefully adopted for the ultrasound specific problems.

First of all, pulse compression is introduced to improve range resolution. Two mutually close targets cannot be distinguished without properly selected pulse duration. Pulse compression is achieved by the intrapulse signal coding and by matched filter use. Fig. 2 shows the resolution improvement when intrapulse coded signal is applied.

There is no one waveform that satisfies all requirements. The applications of waveforms can be summarized as follows:

Simple pulse used:

- In conventional systems;
- Where range accuracy and resolution requirements can be met with a pulse wide enough to provide sufficient energy for detection;
- Where signal generation and processing costs must be minimized.

Linear and nonlinear FM (chirp)

- Commonly used to increase range accuracy and resolution when long pulses are required to reasonable signal to noise ratios (10 to 20 dB).
- Variety of hardware is available to generate and process this waveform type.

The step-chirp and frequency hopping (FH)

- Provides an approximation of the chirp signal, which consists of linear frequency sweep versus time. The step-chirp transmitted waveform usually consists of a sequence of different tones or concatenated frequencies.

Pseudo - chirp

A binary approximation to a chirp derived from chirp waveform in time domain. The start of this waveform is synchronized with the master clock and sampled at every clock period over the duration of the chirp and at each sampling point the sing is determined. If the sign is positive, then pseudo chirp is set to "1", if the signal is either zero or the sign is negative, then the pseudo chirp is set to zero.

Binary phase (biphase) codes

There are two major types:

- **The Barker Code** waveforms are short binary phase sequences that have the property of unit sidelobe level at the matched filter output. The peak response is N, the length of the code. The longest binary Barker sequence is of length 13. This limitation is the major reason that this type of signals not too practical for most large TB signal applications.

- **Pseudo noise (pn) binary** phase sequences are conceptually related to the Barker waveforms albeit longer. The 0 - 180 ^{0} phase code is implemented easily an can be processed simply in the time domain with digital techniques, lumped constant delay lines, or the more recent surface wave acoustic device.

Polyphase codes

Well known polyphase sequences: Frank, P1, P2, P3 and P4 are related to the sampled step-chirp waveform. The Frank polyphase code waveform may be desired and generalized by considering a hypothetically sampled step-chirp waveform.

3. SIDELOBES AND SUPPRESION

Signal coding within a transmitted pulse is often used in order to increase spatial resolution. Some code sequences can give appreciable processing gain. The major disadvantage is that the compressed pulse has range sidelobes, which limit the spatial resolution for closely spaced targets (Fig. 2). The problem of sidelobe suppression has been recognized as a major problem of pulse compression techniques.

The first techniques for sidelobes suppression were based on mismatched receiver. Most of time they were added in front of match filter. This was complicated and was making equipment more expensive and bulky.

These problems was reason for introduction of single mismatched filter (Fig. 1.b), which would combine mismatched receivers and match filter. So, the objective is to design filter, which simultaneously performs compression, and mismatching according to given criteria. Such solutions are economical and also give better overall results.

Different methods of filters mismatching can be roughly classified into two classes of filters. First that suppress maximal sidelobes (MX filters) and second which suppress RMS sidelobes (LS and similar ones). In [1] new algorithms are presented, IRLS (Iterative Reweighted Lest Square) and DIRLS (Doppler optimized IRLS). These new filters combine properties of MX and LS filters and enables considerable simplification of the designing procedure and what is more important, it can be applied to all types of sequences. Paper [2] proposes a mismatching approach to Frequency Hopping (FH) technique.

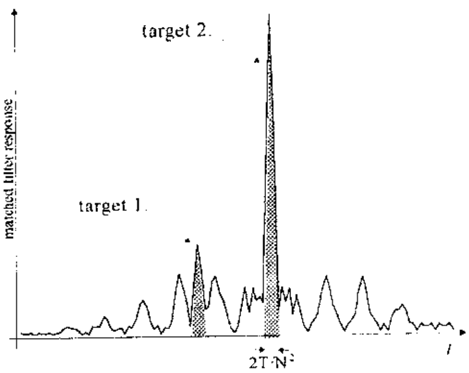

Fig 2. Matched filter response to the signal with compression when there are two targets.

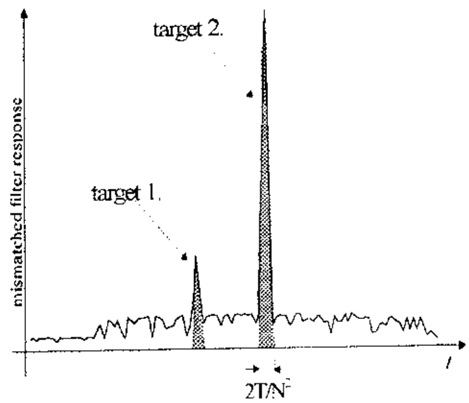

Fig. 3. Mismatched filter response to the signal with compression when there are two targets

In [3] a procedure has been described for self - clutter suppression filter design using the modified RLS algorithm This procedure is applicable both for real and complex sequences. Modified RLS algorithm also offers an advantage compared to the (D)IRLS approach because it is possible to attain a tradeoff among two criteria: suppression of the peak sidelobes and suppression of the mean square sidelobes. Additional benefit in application of the proposed method is its reduced computational complexity compared to the (D)IRLS method. Thus particularly obvious in case of designing Doppler optimized self-clutter suppression filters.

4. IMPROVED RANGE RESOLUTION ACHIEVED BY MISMATCHED FILTER

A complex signal with phase coded pulse is given by

$$u(t)=\sum_{i=1}^{L}u_i(t-nT), \tag{1}$$

where

$$u_i=\begin{cases}e^{j(\omega t+\theta_i)}, & 0\le t\le T_i\\ 0, & elsewere,\end{cases} \tag{2}$$

and θ_i is the phase sequence element, $i=1,2,\ldots,L$.
The sequence at the output of the coherent demodulator is $\{s_n\}=\{s_1,s_2,\ldots,s_i,\ldots,s_L\}$,
where

$$s_i=\begin{cases}e^{j\theta_i}, & 0\le t\le T_i\\ 0, & elsewere,\end{cases} \tag{3}$$

is the complex signal envelope, and L is the sequence length.

If there are more close targets with reflected signal delays within a subpulse equal to T_i, the information about their existence is enclosed in the received signal envelope, but not in the demodulated sequence $\{s_n\}$.

The oversampling process should 'retrieve' that information from the envelope and allow the close range target distinction in further processing.

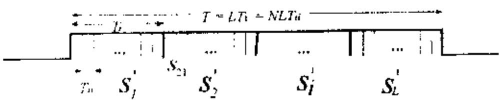

Figure 4: Oversampled pulse train, targets.

The signal at the output of the coherent demodulator is sampled at N time greater frequency than the bit rate. So, every subpulse contains N samples (Figure 4), and the sequence $\{s_n^{'}\}$ can be expressed as

$$\{s_n^{'}\}=\{s_1^{'},s_2^{'},\ldots,s_i^{'},\ldots,s_L^{'}\}, \tag{4}$$

where $\{s_i^{'}\}=\{s_{i1},s_{i2},\ldots,s_{iN}\}$.

That means that in the absence of noise and superposition signals from different targets, each element from the original sequence $\{s_i\}$ is repeated N times.

The matched filter is designed for receiving the $\{s_n^{'}\}$ sequence. There is no resolution improvement so far. To improve the resolution, in order to separate components within a subpulse of the length T_i, a mismatched filter was designed with a response of which the mainlobe is $T_n=2T/N$ wide

Hypothetical filter bank contains a matched filter, a mismatched filter without resolution improvement and mismatched filter with resolution improvement, as shown in [4]. This structure, besides its educational value, can be also implemented practically. By adding a filter in parallel on the receiver input, better resolution for a close target can be achieved, without modifying the main processing channel.

IRLS procedure

The iterative reweighted LS procedure for the design of mismatched filters proposed in [1] can be described as follows:

$$\hat{x}(k)=\left[\mathbf{S}_\Phi^H(0)\mathbf{W}_\Phi(k-1)\mathbf{S}_\Phi(0)\right]^{-1}\cdot \mathbf{S}_\Phi^H(0)\mathbf{W}_\Phi(k-1)\Delta_\Phi(k-1), \tag{5}$$

where $\hat{x}$ are estimated filter coefficients, $[.]^H$ stand for the Hermitian matrix, and $S(0)$ is the signal matrix, with a constant value during the iterative procedure, and has a value for the oversampled sequence.

In Equation (5) $R(k-1)$ is a diagonal matrix, of weighted coefficients in the $(k-1)$th iteration, made by $R(k)=diag(r(k))$. The weight vector $r(k)$ is formed by adaptive adjustment in order to minimize the maximum sidelobe levels of the signal at the mismatched filter output. The role of the window, included in the matrix, can be interpreted as a LS algorithm corrective factor.

The desired autocorrelation function which corresponds to the filter response in the $(k-1)$st iteration, is labeled $\Delta(k-1)$, in the design of the improved resolution filter, and is also equal to the Dirac pulse.

Earlier mentioned the IRLS algorithm for 'normal' resolution can be used for sidelobe the suppression of the compression filter for zero Doppler shift as wall as for segment of ambiguity function. Further more, it can be applied for the sidelobe suppression of periodic and aperiodic sequences and also it can be used on binary, polyphase, chirp and frequency hopping sequences as well.

5. MISMATCHED FILTER RESULTS

For the Barker sequence of length 13, oversampled with $N=4$ times greater frequency, mismatched filter coefficients were designed by the IRLS algorithm. The designed filter length was 52 (4×13), and the desired function width of one sample was T_i. The designed filter was found able to separate close range targets. In Figure 5, the comparison of a matched and a mismatched filter response to a signal of two close targets is shown.

With the mismatched filter, a reduced signal to noise ratio is the price that must be paid. In the case of improved resolution mismatched filter design, this undesired effect represents the main limitation. In Figure 5 a decrease of the mismatched filter main lobe level can be seen, which corresponds to the signal to noise ratio loss measure. With the increase of N, i.e. the ability of close target distinction, the loss of signal to noise ratio is increased too.

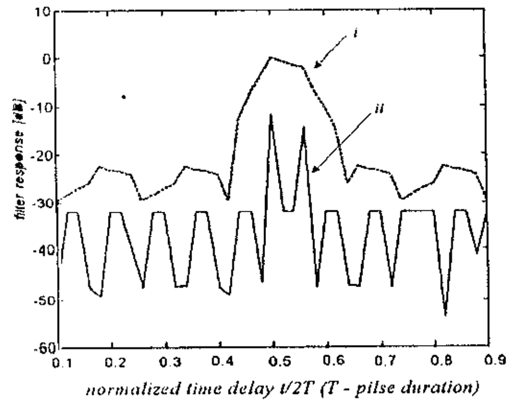

Figure 5: Comparison of i - a matched filter response and ii - a mismatched filter response for the Barker sequence of length 13, when a two close targets signal is present at the input.

As this device does not degrade the basic functions of a compression filter, additional information about close targets are valuable for the primary signal processing.

V. CONCLUSION

Pulse compression technique is a mature technique. It has been hot research topic almost fifty years ago and it has produced excellent results in the area of radar systems for the point targets. Similarly pulse compression has been successfully used for sonar systems.

Approach similar to pulse compression, spread spectrum, has produced extraordinary results in area of wireless communications. The spread spectrum technique is one of the major drivers for recent boom of adoption of wireless voice and Internet communications.

Application of pulse compression for radar with distributed targets (weather radars) has been delayed by pulse sidelobe problems.

In past ten years a considerable amount of research has been done in area mismatched filtering for the purpose of it application in ultrasound [6-10]. However, similarly to radar systems for distributed targets, pulse compression has not achieved a success of penetration in comercialy available systems. GE General Electric) has claimed that it succeeded in adopting pulse compression in comercialy available Logic 700 scanner.

Mismatched filters have not been extensively considered for application in ultrasound area [37]. Simplicity and effectiveness of mismatched filters may make them appropriate for ultrasound systems. However, a lot research should be performed.

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KOMPRESIJA IMPULSA U ULTRAZVUČNOJ DIJAGNOSTICI

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