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LABORATORY EXPERIMENTS WITH AN ADAPTIVE THRESHOLDING SYSTEM FOR RANGE RESOLUTION IMPROVEMENT IN SONAR

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ABSTRACT

The laboratory experiments and performance of a system that is capable of improving the range resolution in a conventional sonar are presented. The model of the system examined is a double differentiator based Adaptive Thresholding System - ATS, whose main task is to resolve overlapping sonar returns. Laboratory experiments proved that for a 1.3 ms "ping", with the bandwidth limited to 2.6 kHz, the ATS discussed was able to resolve up to 7 returns that overlapped within one pulse-length. This figure has been obtained within the amplitude range of 40 dB provided that the signal-to-noise ratio was better than 20 dB. Further improvement in the resolving capability can be obtained by broadening the signal bandwidth, which on the other hand requires better input signal-to-noise ratio. It is believed that apart from improving the quality of the echograms, the ATS here proposed can be useful for a more accurate estimation of the TS depth distribution in low and medium density fish schools.

INTRODUCTION

A considerable problem in numerous areas of science and technology is the unambiguous resolution of overlapping pulses. One such area is acoustics, specifically in sonar systems, where the solution to the above problem enables us "to kill two birds with one stone", namely to improve both the resolution in range and the target detectability in reverberation. The general approach to this problem includes the matched filter pulse compression e.g.,[6], deconvolution, (inverse), filtering e.g.,[9], cepstrum analysis e.g.,[7], mismatched filtering e.g.,[4], "moments" methods e.g., [8], and others.

It is well known that in sonar and radar, where the transmitted pulse is known, the matched filter pulse compression is superior to any linear method in the above respect. Moreover, it provides the best possible signal-to-noise ratio. Unfortunately, the signals that are subject to the operation of compression in the matched receiver are substantially broadband ones and therefore their direct application to conventional sonar systems is difficult, principally, due to the narrow bandwidth of transmitting transducers. In addition, the high cost and complexity of pulse compression systems means that most commercially available sonars and echo-sounders use finite-length rectangular pulses of constant frequency. Unfortunately, the use of such pulses results in a conflict between the maximum range, (signal-to-noise ratio), and the range resolution, (signal-to-reverberation ratio). In other words, the longer the sounding pulse ,the longer the system range (or equivalently the better target detectability in noise), but at the same time the poorer both the range resolution and the target detectability in reverberation. One can partly resolve these conflicting demands by allowing for long pulses so that the system range is not sacrified and using special filters which are capable of recovering the range resolution and improve the target detectability in reverberation at short range. Unfortunately, such resolution improvement filters are not consistent with the idea of matched filter which allows for the optimum detection of targets in noise, and therefore their use results in the degradation of target detectability in noise, relative to the matched filter case. This degradation can be considered as the price to be paid for the improvement in range resolution and target detectability in reverberation.

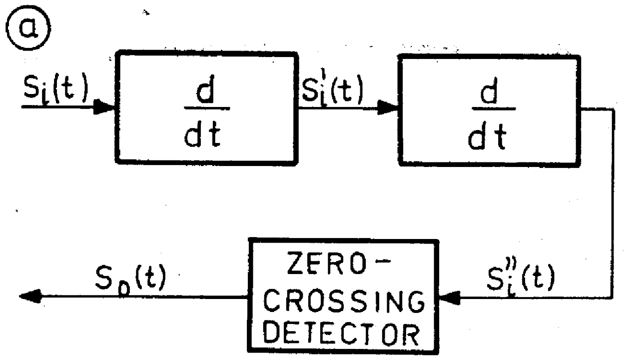
The filters that are capable of resolving nearly rectangular pulses, called the envelope constrained filters have been introduced first by Evans and Fortmann and applied in radar, [4]. The Single Pulse Resolution Filter - SPRF, based upon the idea of deconvolution filter has been developed and applied in fisheries acoustics by Dyka, [1]. An improved version of the SPRF that is immune to the mutual interferences between overlapping returns, called the Interference-Immune Resolution Filter - IIRF has consequently been proposed by Dyka, [2]. The main disadvantage of all of the aforementioned filters is the appearance of undesirable sidelobes at their output which might be a cause of ambiguous target detection.

The aim of this paper is to present a nonlinear filter based upon an Adaptive Thresholding System - ATS, that is capable of resolving unambiguously overlapping returns. Though the form of the filter here presented is quite different from its precedessors i.e., the SPRF and IIRF, the underlying definition of deconvolution filter is the same in all three cases. Therefore the filter here discussed can be, to some extent, viewed as an improved version of the IIRF.

DOUBLE DIFFERENTIATOR BASED ADAPTIVE THRESHOLDING SYSTEM

An adaptive thresholding system determines the position of a pulse in a way which is independent of the pulse amplitude. The most important systems of adaptive thresholding are the level adjuster, the constant-fraction discriminator and the double differentiator. All of them have been discussed in detail by Torrieri, [10].

Compared with other adaptive thresholding systems the double differentiator ATS has the advantage that it does not require an analogue delay line, which makes it simpler. For that reason it was chosen for the filter here discussed. The principle of operation of the double differentiator ATS consists in detecting the inflexion point in the edge of the pulse, which is equivalent to detecting the time when the second derivative of the pulse crosses zero, fig.1.



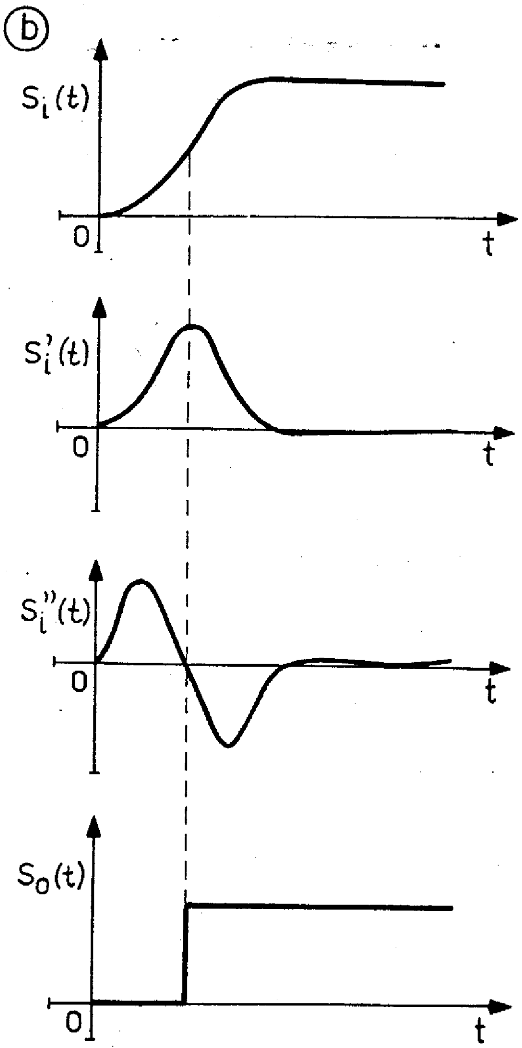


Fig.1. Double differentiator ATS

a – block diagram

b – voltage waveforms

where:

$s_i(t)$ – is the ATS input signal,

$s_i'(t)$ and $s_i''(t)$ – are the first and second derivative of $s_i(t)$, respectively,

$s_o(t)$ – is the ATS output signal

The waveforms in fig.1.b show an idealized situation in which the input pulse is noise free. In a real world however, a certain amount of input noise must be taken into account. As the double differentiator is very sensitive to noise, even a small amount of input noise may have a very detrimental impact on the correct detection of the input pulse. Therefore, the False Alarm Protection – FAP circuit, which protects us against ambiguous pulse detection must be inherently built-in into the double differentiator ATS [10].

RESOLUTION IMPROVEMENT FILTER

The principle of operation of the ATS based resolution improvement filter is based on the assumption that the envelope of a sonar return can be modelled as the output of a linear, low-pass filter with a rectangular pulse input, the latter being the envelope of the sounding pulse, [10]. Then, it can be proved that the distance between the inflection points on the leading and trailing edge of the so modelled return equals the length of the rectangular envelope of the sounding pulse. Using the double differentiator ATS one may detect all inflection points in the envelope of sonar received signal. As can be seen in fig. 1.b. every inflection point in the ATS input signal $s_i(t)$ corresponds to a positive or negative zero-crossing in the ATS output signal $s_o(t)$. Now, suppose, that by a slight change in arrangement, which consists in connecting a monostable flop to the ATS output, every zero-crossing corresponds to a short rectangular pulse, the latter much shorter than the sounding pulse, fig.2. Some of the flop output pulses will correspond to the edges of the return, while others will inevitably be due to noise. An appropriate setting of the FAP circuit described by Torrieri [10] can significantly reduce a number of noise-caused pulses. In order to detect returns one has to determine all pairs of the monostable output pulses that are separated by the time equal to the known pulse-length of the sounding pulse. Such a task can be performed by a special linear sidelobe reduction filter, [3], followed by a fixed threshold,[2]. A simple arrangement that consists of a digital SISO shift register, that delays its input signal by the time equal to the duration of the sounding pulse, and the AND gate can do this job as well. Notably, the latter arrangement is very similar in concept to the pulse-length discriminator, [5].

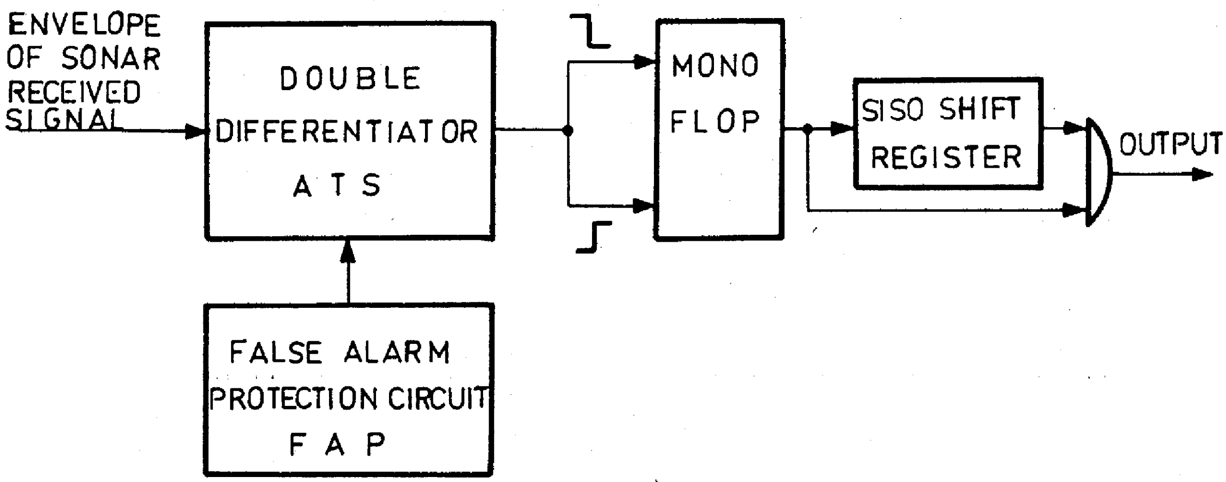


Fig. 2. ATS based Resolution Improvement Filter

CONSTRUCTION AND EXPERIMENTS WITH RESOLUTION IMPROVEMENT FILTER

On the basis of the diagram shown in fig. 2 a model of the Resolution Improvement Filter using standard off-shelf electronic components, both analogue and digital has been constructed. The parameters of the model were set up for handling the 1.3 ms return. The input of the Resolution Improvement Filter was fed with the output signal of the simulator of overlapping returns. The simulator provides facility for the control of the position, pulse-length, and phase of three overlapping returns. This facility enables us to produce a signal with any type of phase interferences, (constructive or destructive), between overlapping returns. Actually, the output signal of the simulator is the low-pass filtered envelope of overlapping returns. It is essential that there are no overshoots in the step response of the low-pass filter, otherwise, due to the operation of differentiation in the ATS, the ambiguous detection with undesirable sidelobes is inevitable. For that reason the Thomson transfer function for the low-pass 1.3 kHz filter was chosen. This figure corresponds to the 2.6 kHz overall bandwidth of the hypothetic sonar system. The measurement arrangement is shown in fig. 3.

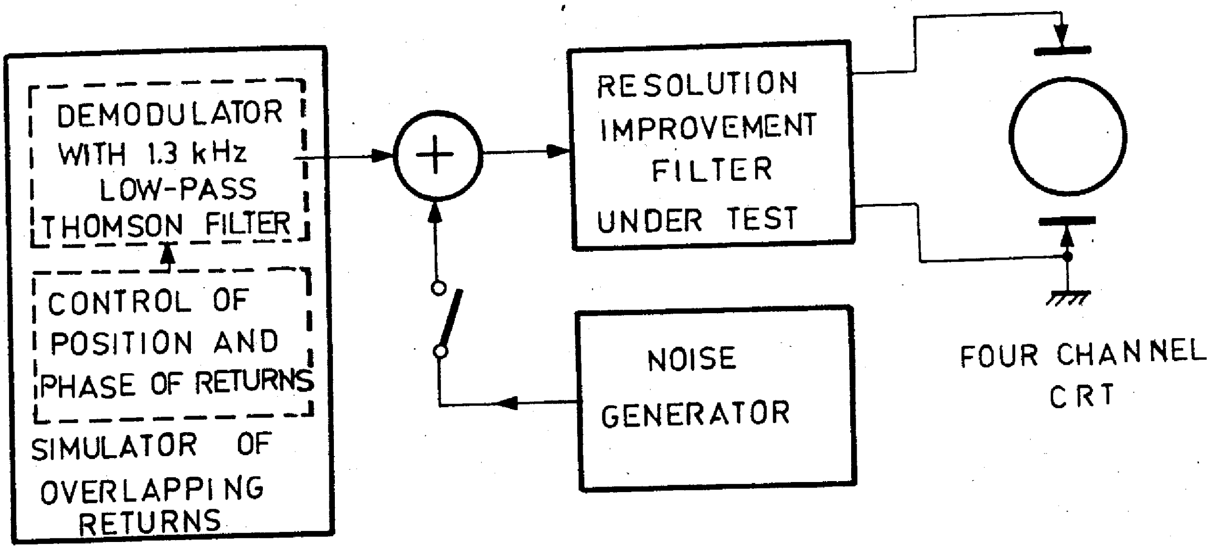


Fig. 3. Measurement arrangement for the tests with the Resolution Improvement Filter

Three samples showing the performance of the filter in the above arrangement are shown in fig. 4. The first of these samples shows the case of constructive interferences between overlapping returns, the other two show the mixed, (constructive and destructive) interferences case.

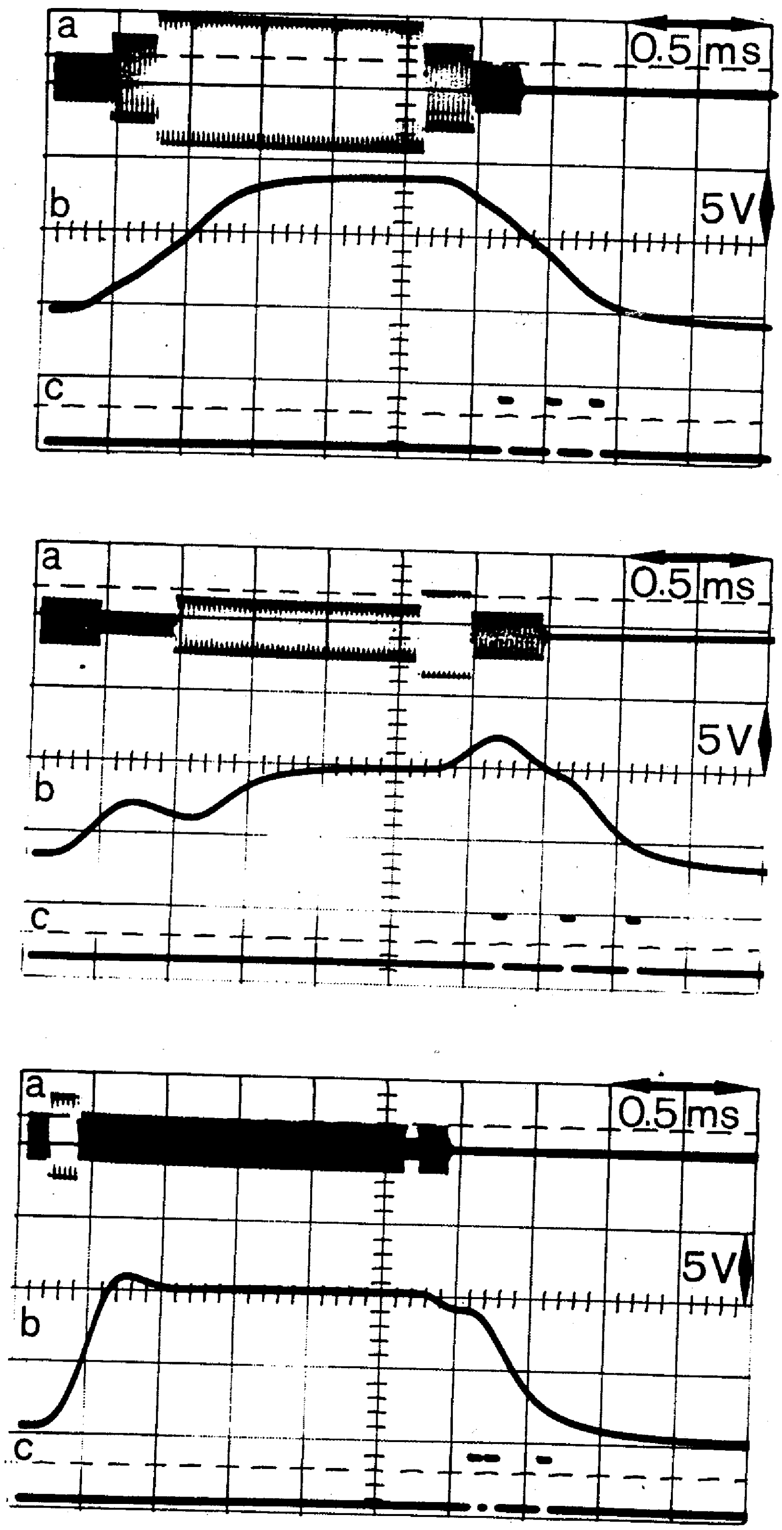


Fig. 4. Samples showing the performance of the ATS based Resolution Improvement Filter

a - three 1.3 ms overlapping pings

b - input signal of the filter being the bandwidth limited, (1.3 kHz) envelope of a

c - output signal of the filter showing the overlapping pulses resolved

The experiments proved that the Resolution Improvement Filter here discussed was capable of resolving up to 7 returns that overlapped within one pulse-length. This result was obtained for most cases of mutual interferences between overlapping returns for varying the input signal amplitude in the range of 40 dB. In some cases however, the correct detection was not possible and the three following situations were observed:

1. The number of detected returns was correct, (three), but their position was incorrect.

2. The number of detections was larger than the number of returns, (up to five).

3. The number of detections was smaller than the number of returns.

It is believed that number of incorrect detections can be reduced by improving the original FAP circuit. All experiments were carried out with the signal-to-noise ratio of about 20 dB and it was found that further increasing this figure did not influence the correctness of the detection. On the other hand, it was observed that for signal-to-noise ratios below 10 dB the number of noise-caused detections increased dramatically. Theoretical considerations and the results of the above experiments show that the maximum number of overlapping returns within one pulse-length that can be resolved with the filter here discussed equals roughly 2WT, where W is the overall bandwidth of the sonar system and T is the duration of the sounding pulse. Notably, for a given duration T of the sounding pulse an improvement in the resolving capability of the filter can be obtained by broadening the overall sonar bandwidth W. The latter however, in practical systems is limited by the bandwidth of the transmitting transducer, which in this way sets a limit for the degree to which the overlapping returns could be resolved. Further improvement in the resolving capability could be obtained by using broader bandwidth transmitting transducers, which, on the other hand, would require better signal-to noise ratio. Eventually, one should note that the received signal amplitude, i.e., the information about TS is lost in the filter but it can be recovered easily by delaying the ATS input signal by the time equal to the sounding pulse duration T and multiplying it with the filter output signal. It is believed that apart from improving the quality of the echograms, the Resolution Improvement filter here proposed can be useful for a more accurate estimation of the TS depth distribution in low and medium density fish schools.

ACKNOWLEDGEMENT

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