Time-domain beam signals for adaptive beamforming

Dr. S. Knabe
ATLAS ELEKTRONIK GmbH
Sebaldsbrücker Heerstr. 235, 28309 Bremen, Germany
sebastian.knabe@atlas-elektronik.com

Abstract — Most sonar signal processing chains like e.g. broadband detection or intercept detection are based on the analysis of time-domain signals that are provided by a beamforming algorithm. The performance of the signal processing is therefore limited by the quality of the beamforming process. Conventional Delay-and-Sum beamforming is limited e.g. in signal to noise plus interference ratio, target separation and beam width depending on the shape of the array and the shading. Adaptive beamforming in contrast can achieve an improved performance and is preferred for broadband detection.
ATLAS ELEKTRONIK GmbH has developed an adaptive beamforming (ABF) algorithm which does not only provide a beam-spectrum with low time resolution for broadband detection but does also calculate time-domain beam signals. Thus the advantageous properties of an ABF like improved target separation capability can be combined with other signal processing chains like narrowband detection and audio signal generation for the operator as well. In this paper, the time-domain signal generation using adaptive beamforming is presented. The properties of the method are discussed by evaluation of experimental underwater acoustics data gathered with different passive sonar arrays as well as with simulated data.
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1 Introduction

Modern submarines used for covert maritime surveillance require to operate as quiet as possible. Therefore, active sonar operations are mostly avoided and passive sonar is the best mean for gathering information of the surrounding environment and potential targets. The challenge in passive sonar processing is to extract as much relevant information from incomplete and noisy data as possible.

The basis of a passive sonar system is an array of hydrophones mounted on the hull of the submarine. Typical examples are the cylindrical array sonar (CAS) located at the bow of the submarine or the flank array sonar (FAS) mounted on both sides of the submarine.

In most sonar applications, signals from the hydrophones are first processed by beamforming to obtain audio signals for different directions of incidence. To achieve spatial selectivity, hydrophone signals are delayed according to the desired look direction and hydrophone positions, and are summed up. The latter technique is referred to as delay-and-sum beamforming and has the advantage of a low computational complexity. However, this approach can suffer from side-lobes, which depend on the design of the hydrophone array. These side-lobes prohibit the detection of a weak noise source in the vicinity of a noisy target. To reduce the level of the side-lobes, an amplitude weighting of array outputs (shading) is used. However, the shading increases the beam width, which limits the capability to separately identify multiple targets that are close in bearing (target separation). As a result, a trade-off between side-lobe level and target separation has to be chosen.

Modern submarines have an increased computing power, which has started the research and design of high sophisticated alternatives to delay-and-sum beamforming. Approaches that use the statistics of the input data are collected under the name “adaptive beamforming” (ABF), which provide suppression of interferences and superior target separation.

Up to now ATLAS ELEKTRONIK GmbH (ATLAS) used ABF [1] for broadband detection (BDT) with direct calculation of the received power as a function of bearing for different frequency bands. In parallel conventional delay-and-sum beamforming with the calculation of time-domain signals is performed also for BDT and other processing like detection of envelope modulation on noise (DEMON), low frequency analysis and recording (LOFAR) and especially audio channels for the operator to listen to sound emitted by the different targets.

The disadvantage of such a design is that a target which can only be detected in BDT using ABF cannot be classified using the signals of the conventional beamforming and the advantages of ABF are not available for other processing chains, which require time-domain signals.

As a result ATLAS developed an ABF approach including the calculation of time-domain signals, which provide high quality beam signals for all kind of signal processing chains.

The paper is organized as follows: the design of the ABF with the calculation of time-domain signals is introduced in section 2. Results of the processing of simulated data are presented in section 3. Results from a sea data set are discussed in section 4 and a summary concludes the paper.

2 Adaptive beamforming with time-domain signals

In the residual part of the paper, the far-field assumption is considered and the incoming hydrophone signals are processed in the frequency-domain to reduce the formulation to a narrowband situation. Thus the sound field impinging the array can be described as a plane wave with wavelength λ and propagation direction determined by a vector $\mathbf{u}$ with unity length. The response of the n-th hydrophone at position $\mathbf{p}_n, n = 1, \dots, N$ with N hydrophones is given by

$$\mathbf{a}_n(\mathbf{k}) = \exp(+i\mathbf{k}'\mathbf{p}_n). \quad (1)$$

Here, $\mathbf{k}$ denotes the wave number vector with

$$\mathbf{k} = -\frac{2\,\pi}{\lambda}\mathbf{u}, \quad (2)$$

i is the square root of -1 and $\mathbf{k}'$ the transposed version of $\mathbf{k}$. The vector of all hydrophone responses

$$\mathbf{a}(\mathbf{k}) = [a_1(\mathbf{k}), \dots, a_N(\mathbf{k})]' \quad (3)$$

is called array response.

The array response can be used directly to calculate the signal for the steering direction given by

$$s_m(\mathbf{k}) = \mathbf{a}(\mathbf{k})^H \boldsymbol{y}_m. \quad (4)$$

Here, $\mathbf{a}^H$ stands for the complex conjugated and transposed version of $\mathbf{a}$ and $\boldsymbol{y}_m$ is the vector with values for the m-th snapshot from the N hydrophones.

Such a beamforming approach is called conventional beamforming in the frequency-domain in the rest of the paper. By collection of s_m for all frequencies and using the inverse Fourier-transform, time-domain signals for one beam are calculated.

For the further introduction of adaptive beamforming algorithms, the so called covariance matrix

$$\mathbf{R} = \sigma_0^2\mathbf{a}_o\mathbf{a}_0^H + \sum_{k=1}^{K}\sigma_k^2\mathbf{a}_k\mathbf{a}_k^H + \mathbf{Q} \quad (5)$$

is required. Here, σ_0^2 denotes the power of the signal of interest with array response $\mathbf{a}_0$, σ_k^2 with the associated array steering vectors $\mathbf{a}_k$ are the powers of K additional uncorrelated signals and $\mathbf{Q}$ is the noise covariance matrix.

Since σ_0, $\mathbf{a}_0$, σ_k, $\mathbf{a}_k$ and $\mathbf{Q}$ are unknown in real applications, the covariance matrix $\mathbf{R}$ is replaced by the sample covariance matrix

$$\widehat{\mathbf{R}} = \frac{1}{M}\sum_{m=1}^{M}\mathbf{y}_m\,\mathbf{y}_m^H. \quad (6)$$

Here, M denotes the number of snapshots from the hydrophones.

The weight vector of the Minimum Power Distortionless Response (MPDR) approach are given by the solution of the optimization problem

$$\mathbf{w}_{\mathrm{MPDR}} = \arg\min\left(\mathbf{w}^{\mathbf{H}}\widehat{\mathbf{R}}\mathbf{w}\right)\ \ s.t.\ \ \boldsymbol{w}^H\boldsymbol{a}_0 = 1. \quad (7)$$

The first part of the constraint demands that weight vector $\mathbf{w}$ with the minimum power $\mathbf{w}^{\mathbf{H}}\widehat{\mathbf{R}}\mathbf{w}$ should be used. To prohibit the solution $\mathbf{w}_{\mathrm{MPDR}} = \mathbf{0}$ the constraint $\boldsymbol{w}^H\boldsymbol{a}_0 = 1$ is added.

This standard MPDR beamforming has one significant disadvantage. In the case that a signal of interest is impinging from a direction between two steering directions, it can be treated as interference for all steering directions. As a result, the signal of interest is supressed. To prohibit such a behaviour, the set of used steering directions is replaced by a set of adjoining steering sectors. As a consequence each signal of interest lies in one steering sector. Thus each signal of interest is treated as a signal of interest for one steering sector and as interference in all other steering sectors. This approach is called robust MPDR beamforming and provides the robust MPDR steering vectors $\mathbf{w}_{\mathrm{rMPDR}}$.

Up to now, ATLAS used the direct calculation of the output power as a function bearing and frequency

$$\mathbf{P}(\mathbf{k}) = \frac{1}{\mathbf{w}_{\mathrm{rMPDR}}{}^H(\mathbf{k})\widehat{\mathbf{R}}^{-1}\mathbf{w}_{\mathrm{rMPDR}}(\mathbf{k})} \quad (8)$$

from the robust steering vectors. The resulting matrix with a dimension of number of beams times number of frequency bands is directly used in BDT.

The new approach is to use $\mathbf{w}_{\mathrm{rMPDR}}$ instead of $\mathbf{a}(\mathbf{k})$ in equation (4). The result is transformed into the time-domain which provides a time signal for each beam.

To achieve a high quality of the beam signals, the parametrization of the robust MPDR has to be modified compared to the solution of equation (8) in BDT. The number of frequency bands and the width of the adjoining steering sectors have to be adjusted and a post processing analogue to audio signal processing has to be performed [2].

3 Simulation Results

3.1 Basic Scenario

In a first step a basic simulated scenario is analysed: four targets with a bearing of 45°, 90°, 109° and 120° consisting of broadband noise are simulated. Two frequency lines at about 1 kHz are added to the first target and the other three target signatures include one additional frequency line. All frequency lines have a much higher signal-to-noise ratio than the broadband signals. For the simulation a uniform linear array with 96 elements with a spacing of 0.29 m is used.

Results of spectra integrated over 1.3 s for each beam signal using conventional beamforming are plotted in figure 1. All four broadband targets with a slight sidelobe structure can be observed. The frequency lines produce a strong peak and provide disturbances in all depicted bearings due to the high signal-to-noise level and the limited sidelobe suppression. Due to the used frequency resolution the two frequency lines for the target at 45° cannot be separated.

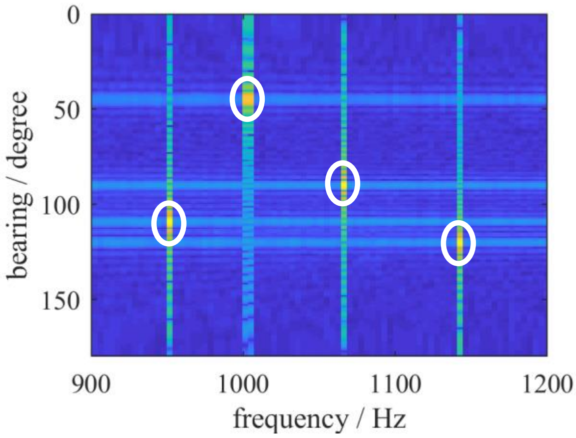

Fig. 1. Matrix of power values as a function of bearing and frequency for the processing of simulated data with four targets using conventional beamforming with the resolution used in the adaptive beamforming. The frequency lines are marked with a white circle.

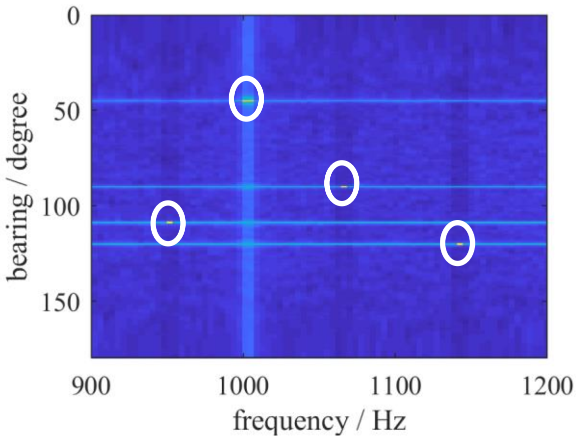

Fig. 2. Matrix of power values as a function of bearing and frequency for the processing of simulated data with four targets using adaptive beamforming following equation (8). The frequency lines are marked with a white circle.

Comparable results using ABF with equation (8) are presented in figure 2. For this image the ABF processes 1.3 s of the stave signals. As in figure 1 all three targets can be detected and an increased level for the frequency lines is observable. But due to the suppression of the interferences, the disturbances by the frequency lines over all bearings are significantly reduced. As in figure 1 the two frequency lines of the target at 45° cannot be separated due to the limited frequency resolution.

One possibility to get a better frequency resolution of the results is to increase the length of the Fourier-Transform. This is easily done for the conventional beam signals e.g. by using LOFAR.

If the number of frequency bands in ABF is increased, the calculation time increases as well: for each frequency band a covariance matrix has to be estimated, inverted, the robust MPDR steering vectors have to be calculated, and equation (8) has to be solved. In addition the number of snapshots for the covariance matrix estimation will be decreased which reduces the robustness of the covariance matrix estimation. Thus, the frequency resolution of the current design of ABF cannot be increased above a certain threshold.

By using LOFAR on the beam signals of conventional beamforming, the frequency resolution in beamforming and signal analysis can be differently selected. Results of LOFAR for the discussed scenario are presented in figure 3. Now, the two frequency lines of the target at 45° are separated.

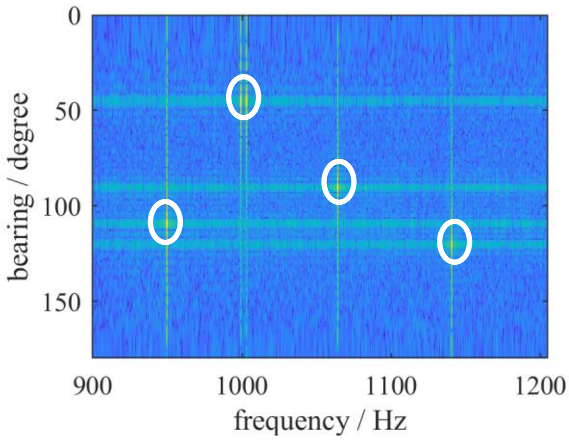

Fig. 3. LOFAR-Gram for the stave signals from conventional beamforming. The frequency lines are marked with a white circle.

The new design for providing time-domain beam signals using ABF makes it possible to also process LOFAR utilizing ABF. Such results are plotted in figure 4. Now the frequency lines of the 45° target can also be separated using ABF with a strong suppression of interferences.

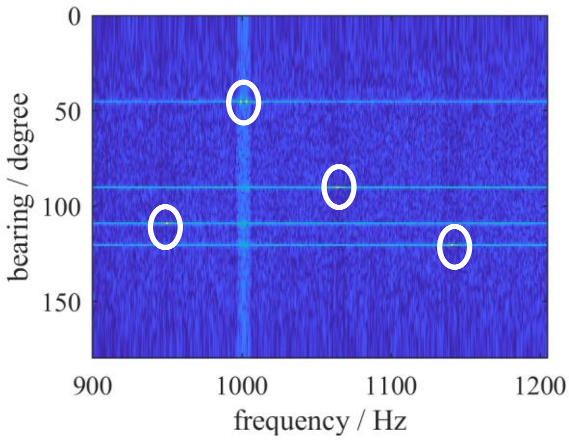

Fig. 4. LOFAR-Gram for the stave signals from adaptive beamforming with time-domain signal calculation. The frequency lines are marked with a white circle.

3.1 Realistic Scenario

To demonstrate the advantages of ABF with time-domain beam signals, a more realistic and complex scenario for the FAS is simulated. The scenario consists of a set of broadband targets with additional frequency line structure. The stave signals are processed using conventional and adaptive beamforming and the beam signals are analysed with DEMON. In DEMON the broadband structure of the targets is removed and only the modulation frequency lines including the bearing information are detected and presented.

Results from DEMON for beam signals using conventional beamforming and ABF with time-domain signals are displayed in figure 5 and figure 6. Both images contain the line-bearing-time-record (LBTR) with the bearing information of the detected DEMON frequency lines over time.

By using ABF a much higher target separation can be achieved, thus it is possible to detect two crossing targets much longer without disturbances. Furthermore a strong target does not mask weak targets as significantly as for conventional beamforming.

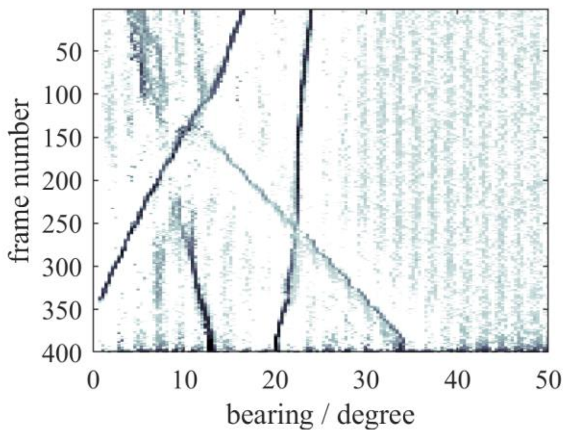

Fig. 5. DEMON-LBTR for the stave signals from conventional beamforming.

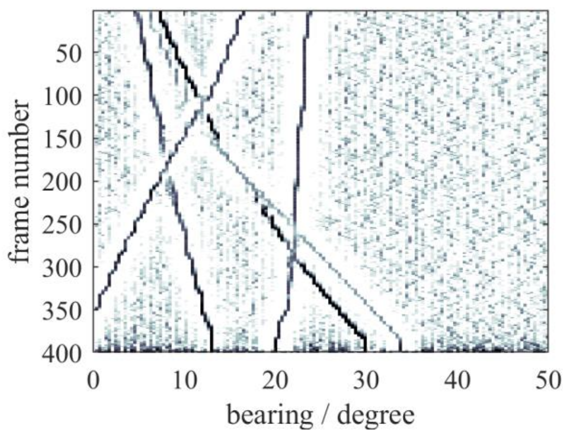

Fig. 6. DEMON-LBTR for the stave signals from adaptive beamforming with time-domain signal calculation.

To sum up, using adaptive Beamforming with the calculation of time-domain signals has the advantage over the former design, that the improved performance of ABF is available for many different signal processing chains.

4 Processing of sea trial data

For a comparison of the different approaches in terms of performance in real scenarios a FAS sea-trial data set is used. The results of standard processing with conventional beamforming and BDT are displayed in figure 7. Here the bearing-time-record (BTR) is displayed together with the own course of the array plotted with magenta marks.

Different target traces with some crossing points can be observed, while the array slowly rotates. Note: due to the reduced performance around endfire-bearings, the target traces are broken for a bearing equal to the own course and equal to the own course ± 180°.

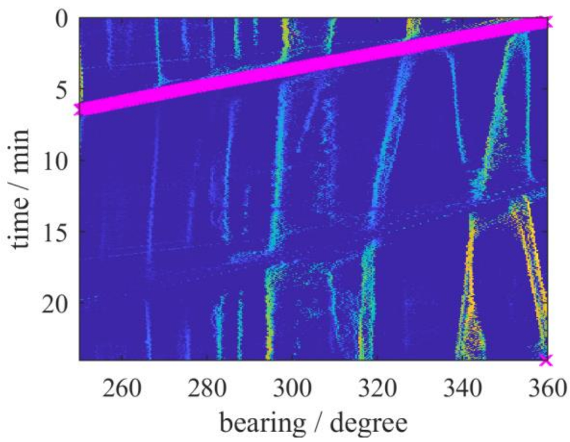

Fig. 7. BTR of BDT for sea-trial data using conventional beamforming.

In contrast, results using ABF with equation (8) and direct processing in BDT are plotted in figure 8. Due to the higher target separation and the interference suppression, much more target traces can be observed. Also crossing targets are much longer separated before the target traces start to interact and merge.

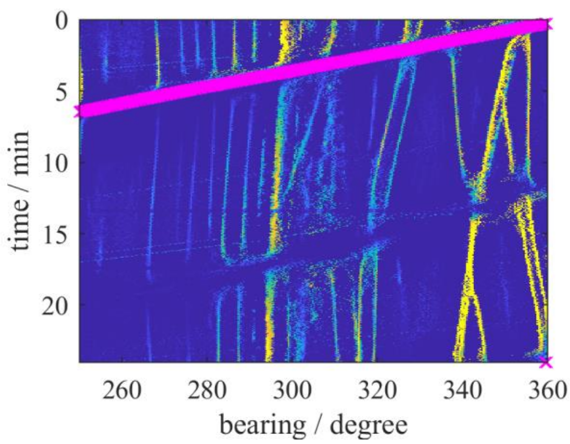

Fig. 8. BTR of BDT for sea-trial data using targets using adaptive beamforming following equation (8).

The corresponding results for the processing using ABF with time-domain signals are presented in figure 9. The results are nearly the same as in figure 8, while mainly the signal-to-noise ratio is different due to the different processing. All target traces can be observed in both figures, the target separation is the same. Only the background noise in figure 9 is slightly higher than in figure 8 while the level of the targets is also slightly higher.

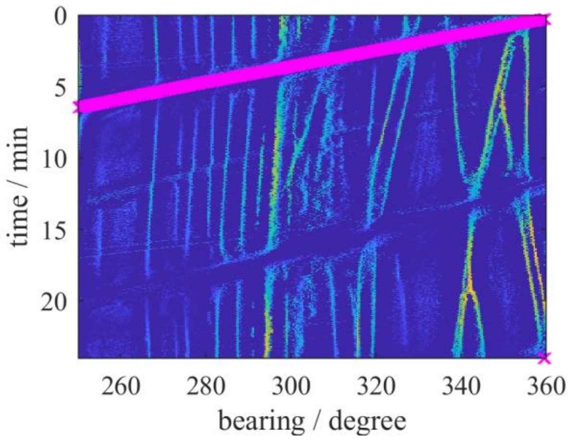

Fig. 9. BTR of BDT for sea-trial data using targets using adaptive beamforming with time-domain beam signals.

The beam-signals for the processing used in figure 9 have one advantage compared to the processing in figure 8: the beamforming produces time-domain beam signals, where e.g. a narrowband processing can be used with or the sonar operator can directly listen to the beam signals.

5 Summary

In this paper a new approach for adaptive beamforming for sonar signal processing by ATLAS ELEKTRONIK GmbH is introduced: the calculation of time-domain beam signals for a MPDR. The enhanced performance of adaptive beamforming like improved target separation and higher signal-to-noise–plus-interference ratio can be combined with other signal processing chains than BDT. High quality time-domain beam signals enable the sonar operator to achieve the better detection results with DEMON and LOFAR. The operator can even listen to the sound produced by the targets. All signal processing chains are now available using ABF without additional delay-and-sum beamforming.

References

[1] S. Knabe and A. Schad, Multi-Purpose Adaptive Beamforming for Passive Sonar, Proceedings of UDT, Bremen, Germany (2017).

[2] S. Leukimmiatis, D. Dimitriadis and P. Maragos, An Optimum Microphone Array Post-Filter for Speech Applications, Proceedings of the International Conference on Spoken Language (ICSLP), Pittsburgh, USA, September 2006, page 2142-2145.