AN ADAPTIVE BEAMFORMING ALGORITHM FOR BROADBAND ACTIVE SONAR

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ABSTRACT

An adaptive beamforming algorithm was developed for broadband active sonar with a convergence time on the order of the pulse duration and effective interference nulling capabilities which enhance desirable echoes. The algorithm is an element based time domain implementation in which beam data that is formed in the direction of each interferer is successively subtracted from the element data using an adaptive FIR filter. The algorithm was applied to impulsive source sonar data received on a bottomed hydrophone array. Collected data contained interfering active returns and noise from nearby shipping as well as desirable echoes from passive reflectors placed near the array. In a representative example, the algorithm adapted fast enough to null out active interference as well as shipping noise, which enhanced the signal to noise ratio of a passive reflector echo by 6 dB.

1. INTRODUCTION

Active sonar operates by transmitting a pulse of energy that reflects off of targets and to the receiver, which may or may not be in the same place as the transmitter. In the process, unwanted returns are also received in the form of volume reverberation, and echoes from the bottom and surface. This is in addition to any nearby shipping noise that may be present. Conventional array beamforming helps mitigate the noise problem by providing spatial filtering gain against these noises. Broadband arrays, however, are typically sparce (unevenly spaced) because they have to provide array gain over a wide bandwidth with a limited number of elements due to the cost of underwater hardware. This results in high sidelobe levels in the beampattern. Adaptive beamforming is useful as a way to improve on the system performance by placing nulls in the sidelobes at directions where interference is louder than the sidelobe level can eliminate.

When the interference is shipping noise or any other slowly varying noise, standard adaptive beamforming techniques can be used [4-8] to null out the interference. For broadband adaptive beamforming, the data is typically spectrally decomposed using a fourier transform (FFT) and the adaption is applied independently to each frequency bin. This limits the adaption to the spatial domain whereas the alternative, a two dimensional tapped delay line in the time domain, can lead to a large number of degrees of freedom. Whether the technique is iterative [4] or a sample matrix inverse type approach [5-8] a number of frequency samples are required to estimate the statistics of the noise field. For the sample matrix inversion approaches, the number of samples required before convergence of the adaptive weights to within 3 dB of optimal is approximately 2 times the number of degrees of freedom [6]. Convergence for the iterative techniques is dependent on the eigenvalue spread [6] but takes a number of samples to converge as well.

For the active signal case, the received waveform is either a pulsed modulated source or an impulsive source. Modulated source waveforms can have a long pulse duration, which would seem to suggest that the adaption rate can be slower, but they should be matched filtered or compressed in time before adaption so as to remove any overlap in time of interferences. This makes it easier for the adaptive beamformer as it can handle the interferers independently, assuming the adaption rate is fast enough. Impulsive sources, which the algorithm in this paper was designed to adapt against, are short in duration to begin with.

Whether a modulated waveform is received and compressed, or an impulsive signal is received, the signal duration is generally much less than even 1 FFT so there is no sample support for nulling out active interferers. There are techniques that mitigate the problem by reducing the effective number of degrees of freedom: Beam based adaption [8], dominant mode rejection [7], and other eigenstructure based techniques. One can also use samples across frequency as opposed to across time, but this is limited by fractional bandwidth considerations. A signal that has frequency extent is related by fractional bandwidth to an interferer with angular extent. the more one averages in frequency, then, the more nulling the adaptive beamformer has to do for each interferer in the data. This effectively reduces the number of interferers that can be nulled out.

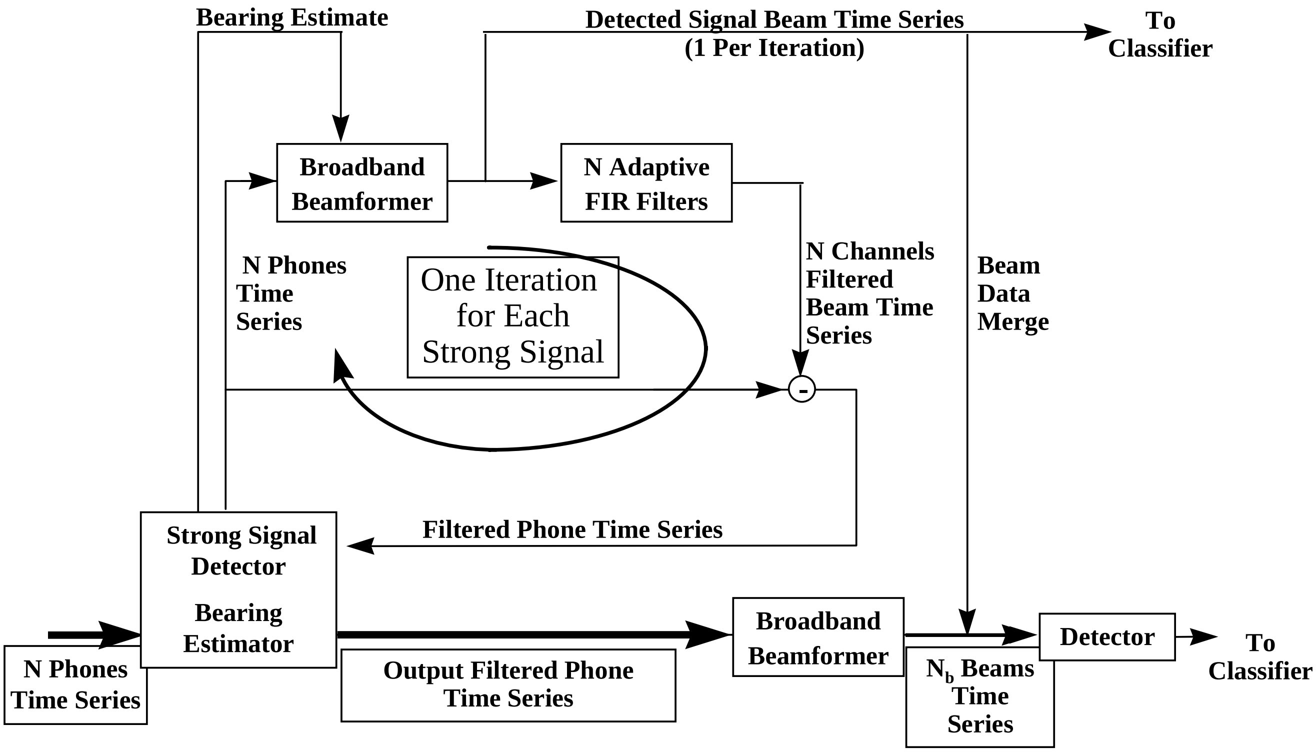

Figure 1. Block diagram of active adaptive beamformer

These techniques place a null in the direction of active interferers, since the interferers statistical information is present in the covariance matrix, but the null is placed over the duration of the integration window and not the duration of the active signal, which is much shorter. For the times when the active signal is not present (even within the same FFT sample), the resulting beampattern is suboptimal and can degrade to the point where the conventional beam response is better.

Anderson [1-3] developed a technique for removal of coherent arrivals on an array that is a time domain implementation, so there is no spectral decomposition, and has the potential for fast convergence since there is no covariance estimate needed for the nulling of interferers. His algorithm removed interferers by forming a beam in the direction of the interferers and subtracting the resulting beam time series, appropriately aligned, from each of the element time series data channels. The direction of the beams was manually set and the interferers being cancelled were passive interferers so his integration time was not particularly fast.

The algorithm studied here is an application of Anderson's technique to active sonar with automation in the bearing estimation and additional filtering in the subtraction part of the algorithm. Adaption rate is also much faster because of the fact that active data is being processed. A block diagram of the algorithm is shown in figure 1. Interferers are removed in a two stage process that is repeated for each interferer in the data segment being processed. In the first stage, strong interferers are detected and the time delay of arrival of the interferer on each element relative to a reference end element is estimated. In the second stage, the element time series data is summed after compensating for the measured time delays to form a "beam" in the direction of the interferer. The beam data is then aligned with each of the elements by delaying the beam data by the inverse of the time delay measured earlier. The beam data is then subtracted from the element data. To further compensate for mismatches between the beam data and the element data, the beam data is also filtered with an adaptive FIR filter that is designed to minimize the output power after subtraction.

This process is repeated for each interferer in the data. The interference free element data is then conventionally beamformed after filtering. This results in 2 sets of beam data, the residual beam data formed from the filtered element data and the reference beam data, both of which are kept for detection and classification.

Adaption rate is determined by the amount of time needed to estimate the bearing of interferers. For active returns, the most time one can use is the (compressed) pulse duration plus the travel time across the array as there is no information outside of this time window. For this application, the data was segmented into 50% overlapped blocks of duration equal to the travel time across the array and the pulse duration. For passive interferences, the integration time could be longer but at this time, there is no logic to distinguish between the two.

Section 2 will discuss the interference bearing estimation stage of the adaptive cancellor. Section 3 describes how the cancellation is implemented given the estimated bearing of the interference. Section 4 details how the algorithm handles multiple interferences. Section 5 describes the results of applying the algorithm to real data, and section 6 is a summary of the results and some of the outstanding issues associated with the algorithm.

2. INTERFERER BEARING ESTIMATION

In order to place a null in the direction of the interferer, a beam is formed in the direction of the interferer. This direction is unknown a priori and so has to be estimated. The technique chosen here is broadband crosscorrelation of each element with a reference end element. This yields a response whose peak location gives the estimated time delay between channels. One could alternatively choose to conventionally beamform the data and then pick the beam having the peak response (assuming they are close enough together) but this assumes the hydrophone locations are known accurately, which is not always the case, and that the signal is in the far field and is a plane wave. Since the interferer signal is strong, the signal to noise ratio is high enough to do element based processing and determine the actual time delays between all of the channels for a particular data segment. Element level correlation processing doesn't assume knowledge of the hydrophone positions or that the signal arriving is a plane wave. The minimum measurement time for the crosscorrelation estimate should be the signal duration plus the travel time across the array so that all of the hydrophones will have full signal data to correlate.

3. INTERFERENCE CANCELLATION

Once the interferer time delays across the hydrophones have been determined (bearing estimation step), the interference is removed. The element time series data is first aligned by delaying each element by its measured interference delay relative to the reference element. The element time series data is then summed, resulting in a beam formed in the direction of the interferer. This beam is then subtracted from each of the element time series after applying an inverse time delay on the beam data as was applied to the element data.

There are a number of real world effects that reduce the effectiveness of the subtraction. There are variations in response of each of the elements as well as decorrelation of the signal with distance from the reference element. The time delay estimation is not exact either as there could be other interferers present that bias the estimate. One could also simplify the algorithm by shifting the beam data to the nearest time delay when aligning it with the element data just before subtraction. In order to compensate for these effects, an adaptive FIR filter is first applied to the interference beam data to make it look as similar to each of the element time series segments as possible. Filtered beam data is then subtracted from each element time series. For this study, 7 taps were used.

It should be noted that this algorithm is not sensitive to correlations between signals as in other traditional adaptive algorithms since the interferer is estimated as beam time series and then subtracted from a limited time delay window on the element time series data so that there is no cancellation of the desired signal if it is correlated with another return.

4. ITERATION PROCESS

There will be multiple strong interferers present in the general case so the process described above needs to be repeated. If, after the initial interference cancellation, there is still sizeable energy, the residual element time series is input back into the bearing estimation stage and the process is iterated on until there are no more strong interferers left. This is determined by setting a threshold on the correlation output so that iterations stop if the summed correlation doesn't exceed a predetermined threshold. An advantage of the iterative process is that the number of iterations is data dependent so that there are only about as many iterations as interferers. This is in contrast to the traditional systems where the processing has to be done on a predetermined estimate of the number of necessary degrees of freedom, which is usually constant and conservative.

In the process of iterating on the interference energy, it's entirely possible that the return of interest will be detected as a loud signal and subsequently removed from the element time series. In the event that this happens, the target return will be contained in the reference beam itself. This can actually be advantageous as the beam will be pointed directly at the target so there will be no scalloping loss as there is normally in conventional beamforming due to the fact that the target could be at an angle between beams. This underscores the necessity of keeping the interference beams and passing them on to the detection and classification stage.

Another scenario is that the bearing estimation accuracy is such that total cancellation is not achieved. In this case, the iteration is useful in that multiple iterations can be used to improve on the rejection of the interferer. If the accuracy is so bad that there is no appreachable rejection and the same interferer gets detected each time and the same bearing estimates get used each time, the algorithm can get stuck in an infinite loop. To account for this, there has to be a failsafe that doesn't use the same bearing twice but perturbs it about the estimated one. This has the effect of broading the null and helps to effectively remove the troublesome interference.

5. RESULTS

The algorithm mentioned above was applied to impulsive sonar data received on a bottomed array. Passive reflector targets were placed nearby the bottomed array. The active adaptive algorithm was applied to a subset of the data containing a passive reflector echo, noise from 3 nearby ships, and active echo interference that was time coincident with the passive reflector echo. The receive array used had a broadband sidelobe response that levelled out at 10 dB down from the peak so that noticeable energy from the interferences leaked into the passive reflector beam. For the purposes of bearing estimation, data was segmented into 50% overlapped blocks of 1/2 second, which is the pulse duration plus the travel time across the array. After 4 iterations of the active adaptive algorithm, the interferences were effectively removed from the passive reflector beam so that the passive reflector echo had a signal to noise ratio 6 dB higher than with conventional beamforming.

6. SUMMARY

An algorithm for processing broadband active sonar data has been presented. The algorithm has demonstrated its ability to null out active as well as passive interferers. Convergence time is approximately the pulse duration plus the travel time across the array. Further work needs to be done to determine the number of iterations to compute for a given data segment. Additional improvements in the bearing estimation can also be made so that continuous interferers are tracked and their resulting bearing estimates can be associated across consecutive data segments. This would lead to a higher accuracy in forming the reference beam.

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