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Implementation of Space Time Adaptive Processing In Active Sonar Detection

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Abstract: *Active SONAR involves the transmission of a short duration acoustic signal (called a ping), which when reflected from a target provides the SONAR receiver with a basis for detection and estimation. The challenge is to carry out detection in the presence of reverberation. All the literature available in this field points out that the computational burden of solving large equations for STAP in real time and significant performance loss due to limited amount of data available during a 'ping' stands forth as a stumbling block in active SONAR detection. Focus will be on simplifying the overall algorithm for STAP implementation. The Kronecker product of spatial and temporal matrices is carried out for the implementation of STAP. Polyphase FIR filter method is proposed to be handled so that hardware implementation of the STAP technology can be simplified.*
Keywords: *Space Time Adaptive Processing, FIR filter, Kronecker product, polyphase filter method.*

I. Introduction

In an active system, energy is transmitted into the medium and the received echo is processed. In the case of RADAR an electromagnetic wave is transmitted that is reflected by the target and captured by the reception system. SONAR on the other hand, relies on acoustic energy in seawater. Usually the information includes the space-time coordinates of the target like the bearing, range and radial velocity. Adaptive Space-Time processing has become very important technology in SONAR applications. The basis for space time signal processing is an antenna array which provides spatial sampling of the signal field. Sensor systems such as radar or sonar receivers are tools for the interpretation of the wave. Operations such as demodulation, filtering or Fourier analysis of the antenna output signal are temporal processing techniques. Space time signal processing is required whenever there are functional dependencies between the spatial and temporal variable. We are considering two dimensional signals sampled in both time and space. The aim is to include both the spatial and temporal samples into the calculation of weight vector which modifies the input data set. To the study the active SONAR detection process the problem formulated was to detect the target within a range of 750m radius. If we want to do detection of distant targets we need longer pulses. But the problem with this method is that we require large amount of samples to process the data which is both time consuming and memory consuming. Thus for practical applications we need further simplification of the process. Digital filters are implemented for this purpose, because of its wide application in the field of communication and computation. To design a Finite Impulse Response filter that satisfies the required conditions is quite challenging. Target detection will become impossible if the parameter chosen for the filter is not appropriate. Further it has to be cost effective. Thus a simplified, practically reasonable technique of doing Active SONAR detection is discussed in this paper.

1.1 Problem Formulation

The first step towards Active Sonar Detection is beam forming. The fundamentals of beam forming and some elementary experiments with array design are carried out first. The simplest beamformer which is delay and sum beamformer is explained because it explains a tricky concept in simple means. It is often easier to carry out signal processing procedures like filtering in Frequency domain. Thus we move on to implementation of beamformer in frequency domain. In all these types of beamformer detection of the bearing of source and its tracking if it was in motion was implemented. Until now simulations were carried out on passive Sonar which just has the behavior of listening. Next we move on to Active Sonar for detection which shows results of active detection of range in the presence of noise. Space Time Adaptive Processing is to be implemented. The theory of a fully adaptive STAP is studied. Implementation of STAP on a scenario where reverberation is present is done. It is found that STAP takes lot time and requires great memory consumption. Due to this we move forth to a simplified implementation of STAP that is using FIR filtering technique.

II. Beamforming

Beamformer is used in the SONAR to convert the acoustic power in water into a beam. It consists of an array of sensors which captures the power and a processor which works on this input to estimate the signal arriving from a desired direction. Thus Beamformer consists of a multi- input single output system. Since it emphasizes signals from a certain direction while attenuating interferences from other directions it makes a spatial filter. We have signal or multiple signals that are located in some region of a space-time field. Array of sensors is used to filter signals in a space-time field by exploiting their characteristics. Usually we want to spatially filter the field such that a signal from a particular angle, or set of angles, is enhanced by a constructive combination and noise from other angles are rejected by destructive interference. A basic beamformer equation is of the form

$y = w^H . x$

where w is the weight, x is the input signal and y is the output signal. The superscript ^{H} represents the Hermitian of a vector(the conjugate transpose).In delay and sum beamformer we can say that it is just the appropriate delay given to each sample of the input signal which modifies it to obtain the output. So the delay vector is the weight vector here.

The various steps for implementation of time domain beamforming are

a) Input signal generation with the help of an inverse beamformer, for a desired number of samples.

b) Construction of a delay table for beam steering which stores for each bearing what will be the delay produced at each sensor.

c) Construction of the beam which is in phase at a particular instant of time ; for all the sensors for each bearing , by picking up the appropriate delays we have stored in the delay table and applying it to each of the samples in the beam.

d) Addition of all the in phase samples to get a reinforced beam at each bearing.

To get the beamforming in frequency domain we use Fast Fourier Transform and beamforming is done separately for each particular bin.

A Minimum Variance Distortionless Response Beamformer was designed in frequency domain. We derive the weight vector for MVDR by taking the auto correlation **R** of the spatial sampling matrix and the steering vector is **S** .**S** denotes the theoretical signal response vector obtained from the signal modal of the array. **R** is also called Cross Spectral Matrix.

Optimality can be achieved only if we have perfect knowledge of the second order statistics of the signal plus noise input to the array, ie **R**. In practice these correlations are unknown and must be estimated from the data. We use Sample Matrix Inversion technique where we turn to the maximum-likelihood estimate of the correlation matrix given by the average of outer products of array snapshots. The output of the array (signal estimate) at the time of k th sample is

$$y(k) = W^T x(k) = x(k)^T W$$

The expected output power of the array is obtained from

$$E[y^2(k)] = E[\,W^T X(k)\; X(k)^T W\,]$$
$$= W^T \mathbf{R\,xx}\, W$$

The problem of MVDR is min $(w^H R\, w)$ such that $w^H s = 1$.

Using the constraint on weight vector and obtain the result as $w = (R^{-1} s) \;/\; (s^H R^{-1} s)$

When strong reverberation signals are present we need better techniques for signal detection and one such methodology is using fully adaptive STAP. We have already seen that the covariance matrix plays an important role in eliminating the noise in the received signals. But the correlation matrix that we used earlier was just dependant on the spatial properties of the signal. The steering vector has also to be modified in such a way that the final weight vector **w** which operates on the input data set includes both spatial and temporal properties. By calculating the average of the outer products of successive range bins we can get a good estimate of this matrix.

Thus to obtain the modified correlation vector we rearrange the data vector which were originally an NxM matrix into an array of NM size. The pre-whitening operation to be performed on the data set has to be averaged from this data array. The next step is to modify the steering vector to incorporate both spatial and temporal properties. For this we make use of a mathematical operator called the Kronecker product. The spatial steering vector and temporal steering vectors are calculated as given below.

$sv_{spatial}(\Theta_0)= e^{(-j*2*pi*(f)nd \sin\Theta_0/c)}$ n = 0,1,2............................. N-1

$sv_{temporal}(\Theta_0)=e^{-j2*pi*(f)mTs}$ m = 0,1,2............................. M-1

Now the Kronecker product of these are taken to obtain the spatio-temporal steering vector. Kronecker product of a matrix A_{mxn} by B_{pxq} is a new matrix with mp rows and nq columns.

Now we want to combine both of these to obtain the STAP weight vector. Using a unit gain on target constraint, the optimum STAP weight vector is $w_{opt}= R^{-1}\, sv_{spatial,temporal}(\Theta_0, f + f_d)$ where f is the center frequency of the signal and f_d is the Doppler Frequency shift.

III. Sub-Array Techniques

Here a large antenna array is split into sub-arrays. Simulations were made for different sub-array techniques. That is implemented by not using the whole sensor arrays continuously for the beamforming section. The sensor spacing should not increase more than λ/2 otherwise problem of Grating lobes will occur. The range of the target can be estimated by operating this weight vector upon successive blocks of data vector. The simulation results were carried out using STAP technology and the target was successfully identified amidst the presence of strong reverberation signals.

IV. FIR Filter Method

Our intention is to put forth a much simplified, practically reasonable technique of doing Active Sonar Detection. The basic concept which were adopted is to baseband convert the signal. Then we can do filtering and re-sampling (according to Noble identity the order is not significant).The advantage of first down sampling is that we can perform filtering at lower rates. These basic steps is then simplified and combined to form polyphase architecture.

A basic filter section consists of adders, multipliers and delay elements. To explain the filter theory consider the filter H(Z) = $\sum_{n=0}^{N-1} h(n)Z^{-n}$

$= h(0) + h(1)Z^{-1} + h(2)Z^{-2} + h(3)Z^{-3} + \ldots + h(N-1)Z^{-(N-1)}$

We are considering that there will be decimation by a factor say M. Thus for the better implementation of decimation let us reshape the filter coefficients into a matrix form with M rows and N/M columns.

$$H(Z) = \sum_{r=0}^{M-1} Z^{-r} H_r(Z^M)$$

$$= \sum_{r=0}^{M-1} Z^{-r} \sum_{n=0}^{(N/M)-1} h(r+nM) z^{-Nm}$$

The aim of the project was to implement a simplified STAP algorithm so that detection of range of the target becomes computationally efficient. Thus prior to this beamforming task we wish to perform the filtering section which is termed a polyphase filter since we are splitting up the signals into different phases and we process signals only for similar phases which is required. Thus as we move to the final step we use the property of Z-transform that if h(n) the impulse response of a base-band filter , has a Z-transform H(Z) , then the sequence h(n)e$^{(+j\theta n)}$) , the impulse response of a pass band filter , has a Z-transform H(Z $e^{-j\theta n}$).

If $H(Z) = h(0) + h(1)Z^{-1} + h(2)Z^{-2} + h(3)Z^{-3} + \ldots + h(N-1)Z^{-(N-1)}$

$= \sum_{n=0}^{N-1} h(n)Z^{-n}$

and $G(Z) = h(0) + h(1)e^{j\theta}Z^{-1} + h(2)e^{j2\theta}Z^{-2} + \ldots + h(N-1)e^{j(N-1)\theta}Z^{-(N-1)}$

$= h(0) + h(1)[e^{-j\theta}Z]^{-1} + h(2)[e^{-j\theta}Z]^{-2} + \ldots + h(N-1)[e^{-j\theta}Z]^{-(N-1)}$

$= \sum_{n=0}^{N-1} h(n)[e^{-j\theta}Z]^{-n}$

then $G(Z) = H(e^{-j\theta}Z)$

Thus we replace $Z^{-1}$with $Z^{-1}e^{j\theta}$. The θ value corresponds to k(2Π/M).Thus Z^{-1} corresponds to $Z^{-1}e^{jk(2\Pi/M)}$ and Z^{-M} corresponds to Z^{-M} $e^{jk(M2\Pi/M)}$.Hence the final implementation comes like this

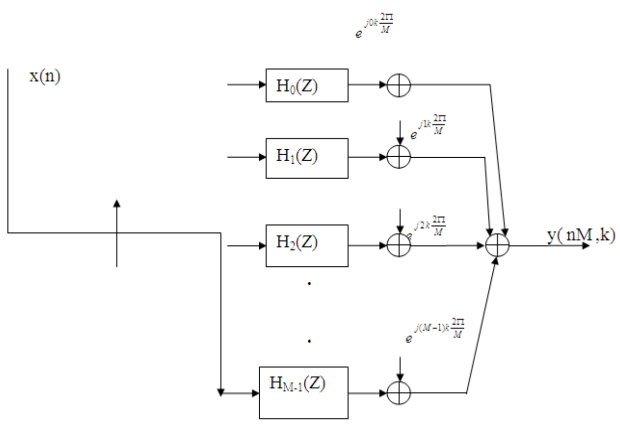

Fig:1 Re sampling M-path Down Converter

This M path down-convertor structure is called polyphase filter. The intention is that we want to perform beamforming using only that bin which corresponds to our central frequency, thus saving a lot space and time for hardware implementation. Referring Fig: 1 we will discard all those paths which doesn't corresponds to our frequency. The samples required for beamforming is now reduced to $1/M^{th}$ of the original samples.

V. Simulation Results

In order to study the system active SONAR was simulated. Along with the transmitted pulse, noise signals and reverberation were also added. The system was used to estimate the velocity of the target from the Doppler vs Look angle plot. The procedure for calculating the velocity is to consider the range of the target to be known and STAP is to be applied on expected range of velocities.

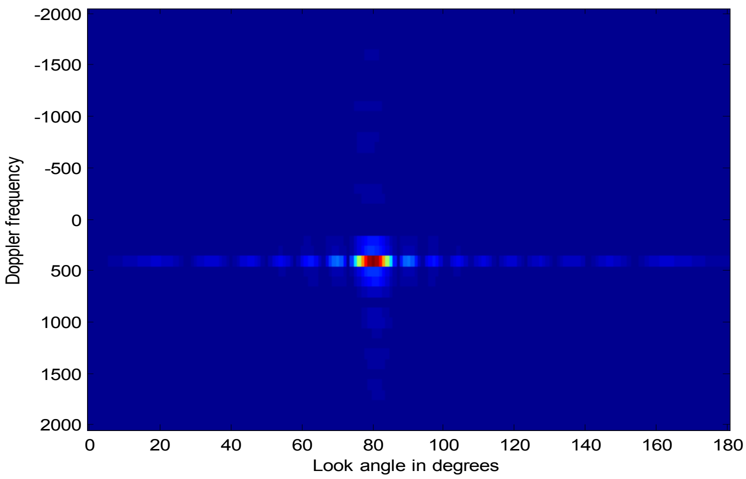

Fig 2: STAP implementation for Doppler frequency vs Look angle plot

VI. Conclusion

A study on implementation of active sonar detection using space time adaptive processing is carried out. Range and velocity detection in the presence of reverberation using **STAP** was done**.** Fully adaptive **STAP** is impractical for reasons of computational complexity and estimation with limited data. A low complexity STAP algorithm was tried to implement using FIR filtering. The hardware implementation of the algorithm using a DSP processor has to be carried out.

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