Comparison of the Performance of Time Domain and Time-Frequency Domain Adaptive Beamforming

C. Bao[1], G. Farag[2] and J. Pan[2]
[1]Defence Science & Technology Organisation, Rockingham, WA 6958, Australia
[2]School of Mechanical Engineering, the University of Western Australia, Crawley, WA 6009, Australia

***Abstract*-Adaptive beamforming is a preferable processing technique over its non-adaptive counterpart for obtaining audio signals in array applications due to its excellent anti-interference ability. In terms of implementation, it can be implemented either directly in time domain or in time-frequency domain. This paper compares the performance of time domain adaptive beamforming and time-frequency domain adaptive beamforming in terms of signal fidelity and computational complexity. The limitations of each beamformer are investigated with regard to bearing and frequency separations of the signal and interferences. The time domain implementation is found to perform generally better in terms of signal fidelity, but its computational complexity is much higher than that of the time-frequency domain implementation. In some cases, such as for large separation bearings between the signal and interferences, the time-frequency implementation gives slightly better fidelity performance.**

I. Introduction

Adaptive beamforming, thanks to its excellent performance in suppression of strong interferences, is a preferable technique over its non-adaptive counterpart for obtaining audio signals in array applications. In terms of implementation, it can be implemented either directly in time domain or in time-frequency domain. Each beamforming approach has its own advantages and disadvantages related to the sensitivity of the algorithms to noise and interferences, as well as their computational complexity. Frequency domain adaptive beamforming, for instance, is known to be of less computational complexity, and is widely used in direction-finding applications. However, when used for audio processing it suffers from the latency (signal delayed) problem and its performance may deteriorate due to the leakage in the domain transformation. On the other hand, time domain adaptive beamforming does not have the latency problem but suffers from high computational cost, and therefore may not be realistic for many applications. This paper investigates the limitations of these two adaptive beamforming techniques with regard to the separation in bearing of the signal and interferences, and the separation of their frequency components.

II. Beamforming

A. Time Domain Adaptive Beamforming (TDABF)

The algorithm in time domain beamforming used in this study is a simple stochastic gradient-descent algorithm called the "Constrained Least Mean Squares" algorithm developed by Frost [1]. This algorithm iteratively adapts the weights of a sensor array so as to respond to the signal arriving from a desired direction while minimising the noise power from all other directions. Noise arriving from the look direction may be filtered out by an appropriate selection of the frequency response characteristics in that direction.

The beamformer structure shown in Fig. 1 is used for the simulations of the TDABF. A processor of this kind having L sensors and J taps per sensor, with a delay of τ between taps, has a stacked array vector given by

$$X(k)=\begin{bmatrix} x_1(k) \\ \vdots \\ x_L(k) \\ x_1(k-\tau) \\ \vdots \\ x_L(k-\tau) \\ \vdots \\ x_1(k-(J-1)\tau) \\ \vdots \\ x_L(k-(J-1)\tau) \end{bmatrix}. \tag{1}$$

The vector of weights at each tap is

$$W^T(k)=[w_1, w_2, \cdots, w_{LJ}]. \tag{2}$$

So the output of the array at the kth sample is

$$y(k)=W^T X(k)=X^T(k)W. \tag{3}$$

The following stochastic constrained LMS algorithm is used for obtaining the adaptive weight $\boldsymbol{W}$.

$$\begin{aligned} &W(0)=F \\ &W(k+1)=P[W(k)-\mu\cdot y(k)X(k)] \end{aligned} \tag{4}$$

where the vector $\boldsymbol{F}$ is a LJ-dimensional vector specifying the desired frequency response in the look direction, the matrix $\boldsymbol{P}$ is obtained from the constraints on each vertical set of weights and μ is the convergence factor.

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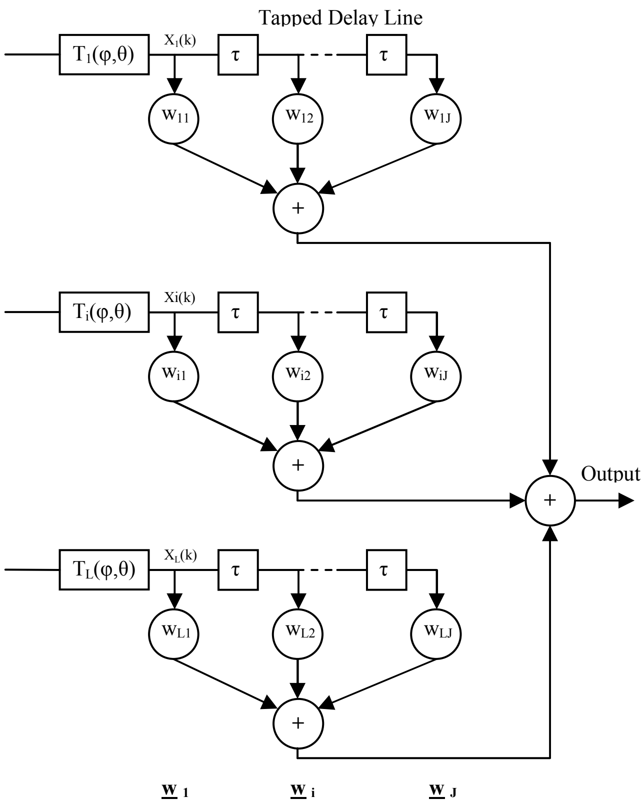

Figure 1. Time domain beamformer structure.

B. Time-Frequency Domain Adaptive Beamforming (TFDABF)

The time-frequency domain beamforming begins with time series array data. The array data are Fourier transformed to the frequency domain and processed by a frequency domain beamformer. The output of that process is then inverse-Fourier transformed back to time domain for an audio output. Fig. 2 shows the schematic presentation of the time-frequency domain beamformer.

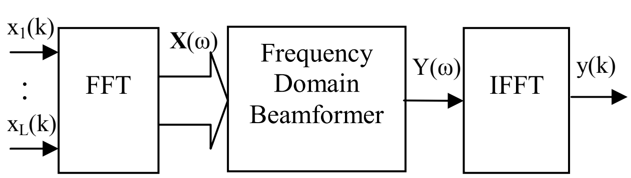

Figure 2. Schematic presentation of the time-frequency domain beamformer.

The beamformer used for the frequency domain processing is the Robust Capon Beamformer (RCB) presented in [2], a type of diagonally loaded MVDR (Minimum Variance Distortionless Response) beamformer. It should be noted that the original purpose of the RCB method is to make the algorithm robust to errors in the signal model, and the appropriate value for the loading is determined by the anticipated signal mismatch. For the audio application, however, loading is being introduced for another purpose [3]. Some degree of loading is required in audio processing to limit the so-called white noise gain (WNG) [4] of the beamformer, even in the situation of without any signal mismatch. The WNG must not be too high because the weights of the frequency domain beamformer are obtained through averaging N blocks of data whereas audio signals are calculated using individual blocks of data (i.e. no averaging). Within each block of data the noise may be stronger in different bearings than those of an average. With a high WNG, this noise might be amplified and interfere with the signal of interest [3].

III. Simulation System

The aim of the simulations is to compare the ability of the TDABF and the TFDABF to produce high fidelity audio output in the presence of strong interferences and background noise. The performance is evaluated in terms of the coherence between the beamformer output and the signal of interest, as well as the computational complexity. As defined in [3], the coherence γ is based on the correlation coefficient, defined in a way that accounts for a time delay between the signal and the time-series output from the beamformer,

$$\gamma = \max_{\tau} \frac{\left|COV\{x(t)y(t-\tau)\}\right|}{\sqrt{COV\{x^2(t)\}COV\{y^2(t-\tau)\}}} \qquad (5)$$

Here $COV\{x(t)y(t-\tau)\}$ denotes the covariance between signal $x(t)$ and the delayed beamformer output $y(t-\tau)$.

Note that γ has a value between 0 and 1, with value 0 indicating statistical independence between the signal and beamformer output, and value 1 indicating that the two waveforms are proportional when an appropriate time-delay is included. It was determined experimentally that a γ of at least 0.950 is required for the audio to be of high fidelity from the viewpoint of the listener. This subjective criterion is used to judge the beamforming performance as successful or not.

The sensor array considered in the simulations is a uniform linear array of 16 elements with an inter-element spacing of 0.17 meters. The speed of sound in air is taken to be 340 m/s and a sampling frequency of 4000 Hz is used. The design frequency of the array is 1000 Hz.

The FIR filters in the TDABF have 64 taps. A larger number of taps will always result in a higher coherence but is accompanied by a greater computational cost. Therefore, a trade-off between these two is made, with 64 taps giving high coherences while maintaining a reasonable computational cost.

The TFDABF uses a FFT (fast Fourier transform) of length 512, which gives a binwidth (frequency resolution) of 7.8125 Hz. This size was experimentally found to give the lowest permissible frequency resolution capable of adequately detecting all frequencies in the FFT frequencies, without excessive latency in audio processing. If the frequency resolution is too low, signal frequencies between the FFT frequencies are not adequately detected, resulting in a falsified perception of the incoming signal frequencies and a lower coherence. On the other hand, a finer frequency resolution would achieve better results but at the expense of a much larger latency.

The signal and interferences in the simulations are sinusoidal waveforms having four tonal components, each varying around 200 Hz, 400 Hz, 600 Hz and 800 Hz. Frequencies of these orders were selected to simulate a broad range of low, mid and high frequencies, relative to the design frequency of the array.

IV. Simulation

The simulations investigate the following limitations on the TDABF and TFDABF algorithms.

i) The minimum separation angle $\Delta\theta$ between the signal's direction-of-arrival (DOA) and the DOA of the interferences, when
   a) The signal arrives from broadside, or 0° (perpendicular to the sensor array), and the two interferences equally spaced on either side of the signal.
   b) One interference arrives from endfire, or 90°, and the signal and the other interference equally spaced away from it.

ii) The minimum separation of the signal and interference frequencies in multiples of the binwidth of the FFT.

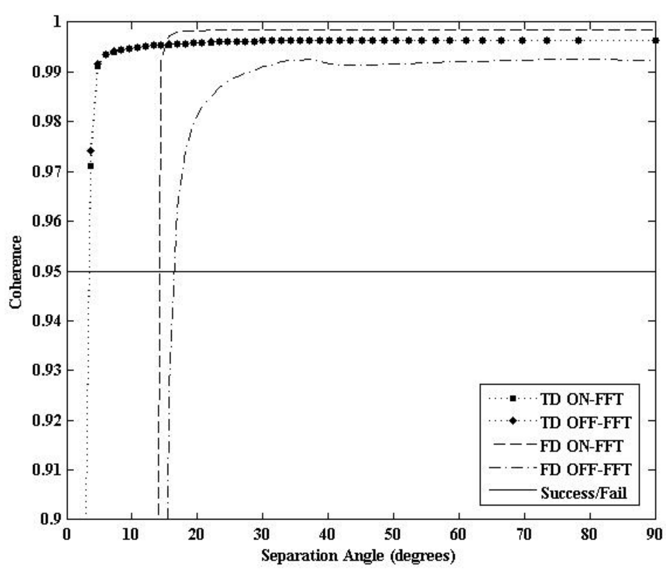

Figure 3. Scenario 1 – signal arriving from 0°.

The above cases are tested for signal and interference frequencies that lie i) on, and ii) between the FFT frequencies. The results are summarised in Figs. 3-5. These results are obtained by computing the coherence γ by averaging over 100 independent runs of the simulation process, where the phase relations among the tonal components are randomly chosen for each run.

A. Scenario 1

The first scenario has the acoustic signal arriving from 0° and consisting of four tonal components. The frequency separations between the signal and interferences are four binwidths of FFT. Both interferences have a power of 10 dB relative to the signal power. White noise is also added with a relative power of 0 dB to the signal power. The results of this scenario are shown in Fig. 3. It should be noted that those results represent the best performance achievable by both approaches, as the narrower beamwidth associated with around the broadside direction produces the minimal overlap of the signal and interference power patterns.

The simulation shows that the TDABF generally performs better than the TFDABF. Firstly, the performance of the TDABF is not affected by the condition of whether the signal and interference frequencies lie on or off the FFT frequencies, whereas the performance of the TFDABF is affected by this condition. Secondly, the minimum separation angle $\Delta\theta$ of the TDABF can be as small as 3°, whereas that of the TFDABF is 14° for the on-FFT frequency cases or 16° for the off-FFT frequency cases. Thirdly, the coherence achieved by the TDABF is higher than that of the TFDABF in most cases, except for those cases of on-FFT frequencies with large separation angles.

As for the computational complexity, the ratio of the TDABF to the TFDABF is 9:1.

B. Scenario 2

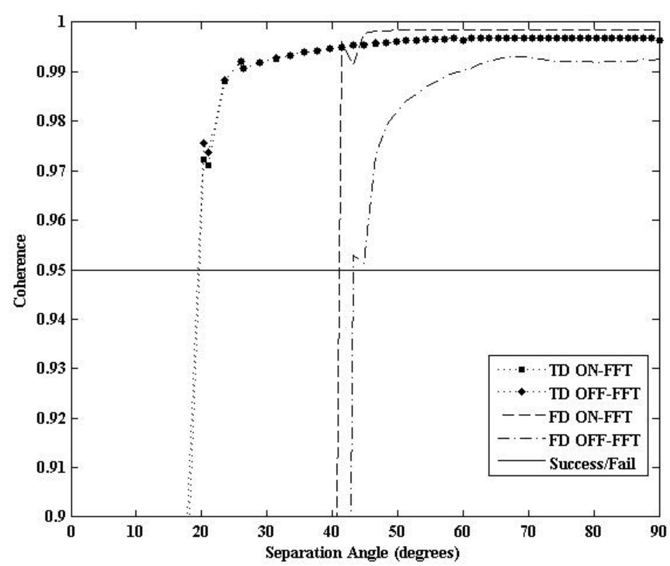

Figure 4. Scenario 2 – interference arriving from 90°.

We now place one interference at 90° – the direction that results in the widest beamwidth. This scenario is run to investigate the effect of wider beamwidths on the proximity of the interferences to the signal. All the other parameters are identical to those in Scenario 1. The results are shown in Fig. 4.

Again we see that the TDABF generally performs better and can handle smaller separation angle than the TFDABF, and the same effects of on and off FFT frequencies is shown. However, all four cases feature a larger minimum separation angle $\Delta\theta$. This is indeed due to the wider beamwidth of the signal and interferences around the endfire direction which produces more overlap of the signal and interference power patterns.

C. Scenario 3

We are now interested in investigating the effect of reducing the separation of signal and interference frequencies. Here the signal and interferences only contain frequencies that lie on the FFT frequencies since simulations showed that the TFDABF fails in the off-FFT case for all $\Delta\theta$ while the frequency separation is less than four binwidths. The simulations are run for a frequency separation of one and two binwidths and using the same parameters as in scenario 1. The results are shown in Fig. 5.

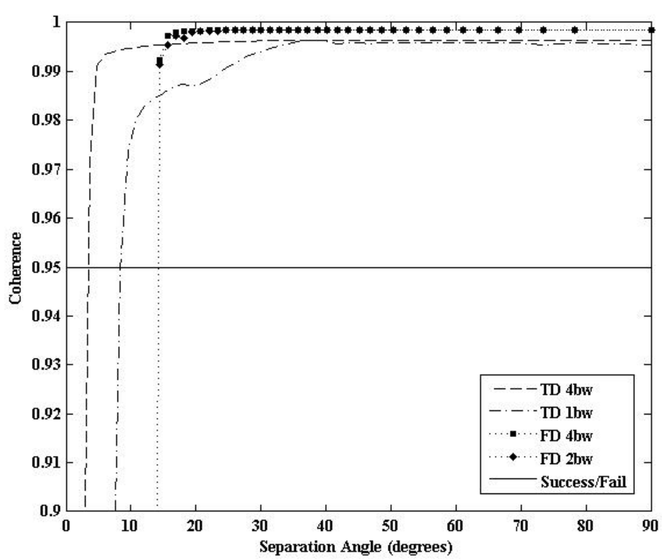

Figure 5. Scenario 3 – signal arriving from 0°, different frequency separations, all on-FFT frequencies.

The TFDABF does not display much of a change yet is limited by a frequency separation of two binwidths, as it fails for all $\Delta\theta$ when the separation is one binwidth. The TDABF, however, is capable of handling the frequencies separated by only one binwidth but its performance is affected by the closer interference frequencies. Upon inspection of curves 'TD 4bw' and 'TD 1bw' in Fig. 5 it can be concluded that the performance of the TDABF deteriorates as the frequency separation is reduced. This can be explained by the increased correlation between the signal and interference that comes with smaller frequency separations.

It is known that the application of these adaptive beamforming algorithms is limited by the requirement that the non-look direction noise and interferences on the taps are uncorrelated with the look-direction signal [1]. In the case of this investigation, this effect is revealed in the cross spectral matrix R, where the signal and interferences have some degree of correlation. It is possible, however, to effectively de-correlate coherent sources using spatial smoothing techniques [6].

V. CONCLUSION

The time domain and time-frequency domain adaptive beamforming algorithms have been compared in terms of the signal fidelity and the computational complexity. Simulations show that the TDABF generally performs better in terms of the signal fidelity. The TDABF is able to handle situations of very small separations in bearing as well as in frequency between the signal and interferences, while the TFDABF is prone of failure in cases of small separations. However, in some cases of large separations (bearing and frequency), particularly when the signal frequencies lie on the FFT frequencies, the TFDABF can achieve higher fidelity than the TDABF. The TFDABF algorithm also has the advantage of much less computational cost and therefore is suitable to use in these cases.

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