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Signal Processing 87 (2007) 2535–2543
SIGNAL PROCESSING
www.elsevier.com/locate/sigpro

Frequency invariant beamforming for two-dimensional and three-dimensional arrays

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Received 15 August 2006; received in revised form 8 March 2007; accepted 31 March 2007
Available online 14 April 2007

Abstract

A novel method for the design of two-dimensional (2-D) and three-dimensional (3-D) arrays with frequency invariant beam patterns is proposed. By suitable substitutions, the beam pattern of a 2-D or 3-D arrays can be regarded as the 3-D or 4-D Fourier transform of its spatial and temporal parameters. Since frequency invariance can be easily imposed in the Fourier domain, a simple design method is derived. Design examples for the 2-D case are provided.

Keywords: Frequency invariant beamforming; Linear arrays; Planar arrays; 3-D arrays; Fourier transform

1. Introduction

In the past, broadband beamformers have been studied extensively due to their wide applications to sonar, radar and communications [1–3]. Amongst them is a class of arrays with frequency invariant beam patterns [4–10], which aim to overcome the fact that for fixed aperture, the spatial resolution is proportional to the signal frequency.

In order to achieve a frequency invariant response, we may optimise the array's coefficients directly with respect to the desired response. However, this is not a practical approach for large arrays. Moreover, for two or three-dimensional arrays, even for a moderate size, the number of coefficients to be optimised can be extremely large and it is almost impossible to optimise them directly. In the planar array design example provided later, the number of coefficients is $24 \times 24 \times 24 = 13\,824$. Another approach is harmonic nesting, where for a number of frequency bands, different subarrays with appropriate aperture and sensor spacing are operated [1,4,5,11]. This method can be based on frequency bin processing [1,4,5] or a decomposition into octave bands by means of filter banks [11]. Subsequently, each octave band or group of frequency bins lying within one octave draws their inputs from one specific subarray. While the resulting beam pattern is octave-independent, the spatial resolution within an octave band is still dependent on frequency.

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0165-1684/$ - see front matter
doi:10.1016/j.sigpro.2007.03.018

To achieve invariance within octaves, [8] combines harmonic nesting and filter-and-sum beamforming together, whereby each element of a subarray is followed by an FIR filter whose response is determined by the desired beam pattern. Frequency bin-dependent windowing of the array elements can lead to a constant beamwidth for the main beam [1], whereby the same method can be applied in the time domain with suitably designed lowpass filters following each sensor [7]. In both approaches, sensor elements close to the array's end positions are disemphasised at higher frequencies, thus yielding a constant beam width and near frequency invariant beam pattern. Different from the above methods, an approach employing the asymptotic theory of unequally spaced arrays has been suggested in [6], where the relationship between the beam pattern properties and array properties is derived and exploited for the broadband linear array design. More recently, a systematic method has been proposed in [9,10], which can be applied to one-dimensional (1-D), two-dimensional (2-D) and three-dimensional (3-D) arrays. In this new method, each element in the array is followed by its own primary filter and the outputs of these primary filters share a common secondary filter to form the final output. Although the design for a 1-D array is relatively simple because of the dilation property of the primary filters, for higher-dimensional arrays this property is not guaranteed, which makes the general design case too complicated and no design examples for a 2-D array were provided there.

In [12], a novel method was proposed to design a frequency invariant beamformer with specific beam direction for a broadband linear array. In this work, we extend the idea in [12] to the problem of how to achieve a general desired frequency invariant response with arbitrary beam pattern for a linear array and then focus on the design problem of 2-D and 3-D arrays. Starting from the desired frequency invariant beam pattern of an n-D array, the proposed method uses a series of substitutions and an $(n+1)$-D inverse Fourier transform to achieve a frequency invariant beamformer design.

This paper is organised as follows. In Section 2, we will introduce the idea for the linear array case by both reviewing and generalising the method proposed in [12]. In order to understand the 2-D and 3-D cases easily, a detailed discussion about the linear array case will be provided. In Section 3 we will propose the design procedure for the case of a 2-D array. As its extension to the 3-D case is straightforward, details for the 3-D design are omitted. A design example is given in Section 4 and conclusions are drawn in Section 5.

Note in the derivations presented in this paper we always assume that all of the array elements are omni-directional and have a constant response over the required frequency range. If the array elements have a frequency-dependent response, then under the reasonable assumption that all of them have the same response, the frequency-dependent response can be compensated using a single filter at the output of the frequency invariant beamformer obtained by the proposed method.

2. 1-D array

An equally spaced linear array with a sensor spacing of d_x is shown in Fig. 1. The received signal by the mth sensor is sampled with a sampling period of T and then processed by a digital filter with coefficients $d[m,n]$, $n=-\infty,\ldots,-1,0,1,\ldots,+\infty$. The response of the array is given by

$$P(\omega,\theta)=\sum_{m,n=-\infty}^{\infty} d[m,n]\,\mathrm{e}^{-\mathrm{j}m(\omega\sin\theta/c)d_x}\,\mathrm{e}^{-\mathrm{j}n\omega T}. \tag{1}$$

To avoid aliasing in both the spatial and temporal domains, T should be less than half of the period of the maximum frequency $\omega_{\max}$ of interest and d_x should be less than half of the wavelength $\lambda_{\max}$ corresponding to $\omega_{\max}$. If we assume the limit case for alias-free sampling, we have $d_x=\lambda_{\max}/2=cT$ and $\omega_{\max}T=\pi$. We denote the normalised angular frequency by $\Omega=\omega T$. Then, (1) can be rewritten as

$$P(\Omega,\theta)=\sum_{m,n=-\infty}^{\infty} d[m,n]\,\mathrm{e}^{-\mathrm{j}m\Omega\,\sin\theta}\,\mathrm{e}^{-\mathrm{j}n\Omega}. \tag{2}$$

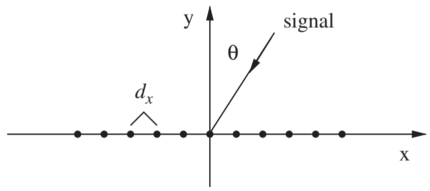

Fig. 1. An equally spaced linear array with a sensor spacing of d_x, where the signal impinges from the direction θ.

2.1. Design idea

By substituting $\Omega_1 = \Omega \sin\theta$ and $\Omega_2 = \Omega$, (2) yields

$$P(\Omega_1, \Omega_2) = \sum_{m,n=-\infty}^{\infty} d[m,n]\,\mathrm{e}^{-\mathrm{j}m\Omega_1}\,\mathrm{e}^{-\mathrm{j}n\Omega_2}. \tag{3}$$

It is clear that the beam pattern of a linear array can be obtained by first applying a 2-D Fourier transform to $d[m,n]$ and then re-substituting $\Omega_1 = \Omega \sin\theta$ and $\Omega_2 = \Omega$. Now the spatio-temporal spectrum of the impinging signal lies on the line $\Omega_1 = \Omega_2 \sin\theta$, which for a variable angle of arrival θ covers the area between the two lines $\Omega_1 = \Omega_2$ and $\Omega_1 = -\Omega_2$ as shown in Fig 2.

For the beam pattern to be frequency invariant, the 2-D Fourier transform $P(\Omega_1, \Omega_2)$ must be a function of only θ, or more precisely $\sin\theta$. Let $F(\sin\theta)$ be such a frequency invariant beam pattern. In order to match this desired beam pattern, the function $P(\Omega_1, \Omega_2)$ must, after re-substituting $\Omega_1 = \Omega\sin\theta$ and $\Omega_2 = \Omega$, be identical to $F(\sin\theta)$. In order to achieve this, the variables Ω_1 and Ω_2 must obey a specific dependency in the expression of $P(\Omega_1, \Omega_2)$ for Ω to disappear. Note that if $P(\Omega_1, \Omega_2)$ depends on Ω_1 and Ω_2, such that it can be written as $P(\Omega_1/\Omega_2)$, then after the re-substitutions, Ω_1/Ω_2 will change to

$$\frac{\Omega_1}{\Omega_2} = \frac{\Omega \sin\theta}{\Omega} = \sin\theta, \tag{4}$$

thus eliminating any dependency on Ω.

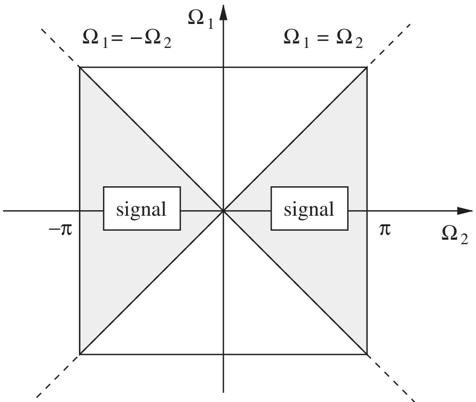

Fig. 2. The possible location of the spatio-temporal spectrum of the impinging signal on the (Ω_1, Ω_2) plane.

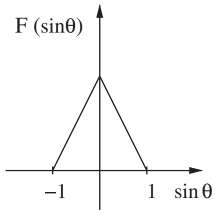

Fig. 3. An example for a frequency invariant beam pattern.

2.2. Design procedure

Given a desired frequency invariant beam pattern $F(\sin\theta)$ as shown in Fig. 3, which may be obtained by a narrowband beamformer design method, we can obtain the 2-D response $P(\Omega_1, \Omega_2)$ by the substitution $P(\Omega_1, \Omega_2) = F(\Omega_1/\Omega_2)$. Thereafter, an inverse 2-D Fourier transform is applied to $P(\Omega_1, \Omega_2)$, which yields the time domain parameters of the desired beamformer. The precise steps are outlined below.

Step 1: Let the desired beam pattern be $F(\sin\theta)$. With the substitution $\sin\theta = \Omega_1/\Omega_2$, for $(\Omega_1, \Omega_2) \in [-\pi; \pi)$ we have

$$P(\Omega_1, \Omega_2) = \begin{cases} F(\Omega_1/\Omega_2) & \left|\dfrac{\Omega_1}{\Omega_2}\right| \leqslant 1 \cap \Omega_2 \neq 0, \\ A(\Omega_1, \Omega_2) & \text{otherwise}. \end{cases} \tag{5}$$

$A(\Omega_1, \Omega_2)$ is an arbitrary function to define values for $P(\Omega_1, \Omega_2)$ and it will not affect the beam pattern because no signal exists in this area according to Fig. 2. Note that $P(\Omega_1, \Omega_2)$ is a function with a period of 2π. The response of $P(\Omega_1, \Omega_2)$ for the example of Fig. 3 is shown in Fig. 4 with $A(\Omega_1, \Omega_2) = 0$.

Step 2: Applying a 2-D inverse Fourier transform to $P(\Omega_1, \Omega_2)$ results in an infinite support of $d[m,n]$. It is difficult to obtain the result analytically; therefore we can apply the 2-D inverse discrete Fourier transform (DFT) as an approximation by sampling $P(\Omega_1, \Omega_2)$. After applying an inverse DFT, the resulting $d[m,n]$ may need to be delayed along the discrete time index n for causality and be truncated according to the number of sensors and the digital filter length attached to each sensor.

3. 2-D and 3-D arrays

3.1. 2-D arrays

Fig. 5 shows the structure of a planar array with sensor spacings of d_x and d_y, respectively. With

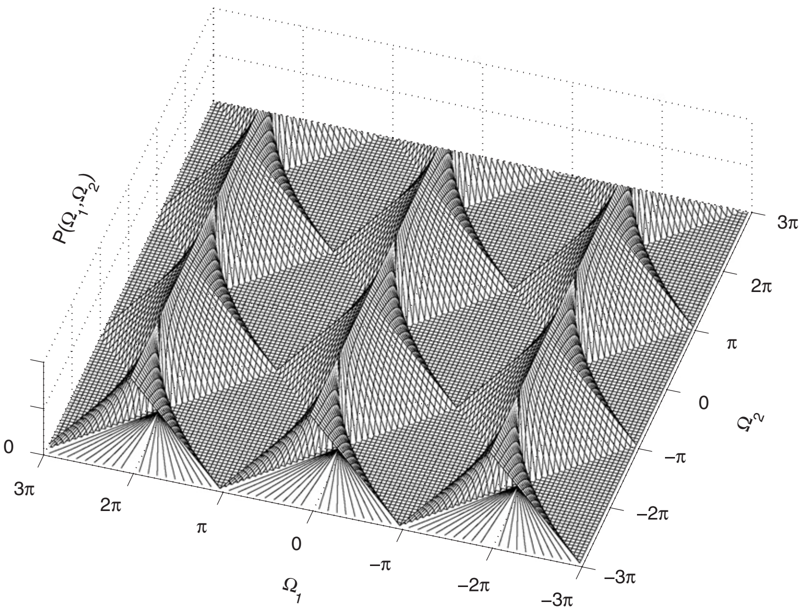

Fig. 4. The response of $P(\Omega_1, \Omega_2)$ for the example of Fig 3.

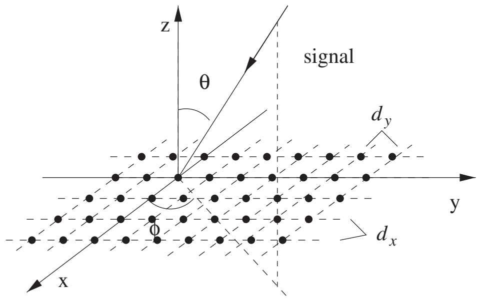

Fig. 5. An equally spaced planar array with sensor spacings of d_x and d_y, respectively, where the signal impinges from the direction (θ, ϕ).

spatial indices l and m and the time index n, the 2-D array response is given by

$$\begin{aligned} P(\omega,\theta,\phi) &= \sum_{l,m,n=-\infty}^{\infty} d[l,m,n] \\ &\quad \times \mathrm{e}^{-\mathrm{j}l(\omega/c)d_x \sin\theta\cos\phi}\,\mathrm{e}^{-\mathrm{j}m(\omega/c)d_y \sin\theta\sin\phi}\,\mathrm{e}^{-\mathrm{j}n\omega T}. \end{aligned} \tag{6}$$

Assuming the same setup for spatial and temporal samplings as in Section 2 such that $d_x = d_y = \lambda_{\max}/2 = cT$, we have

$$\begin{aligned} P(\Omega,\theta,\phi) = &\sum_{l,m,n=-\infty}^{\infty} d[l,m,n] \\ &\times \mathrm{e}^{-\mathrm{j}l\Omega\sin\theta\cos\phi}\,\mathrm{e}^{-\mathrm{j}m\Omega\sin\theta\sin\phi}\,\mathrm{e}^{-\mathrm{j}n\Omega}. \end{aligned} \tag{7}$$

Substituting $\Omega_1 = \Omega\,\sin\theta\cos\phi$, $\Omega_2 = \Omega\,\sin\theta\sin\phi$ and $\Omega_3 = \Omega$ into (7) gives

$$P(\Omega_1,\Omega_2,\Omega_3) = \sum_{l,m,n=-\infty}^{\infty} d[l,m,n]\,\mathrm{e}^{-\mathrm{j}l\Omega_1}\,\mathrm{e}^{-\mathrm{j}m\Omega_2}\,\mathrm{e}^{-\mathrm{j}n\Omega_3}. \tag{8}$$

The spatio-temporal spectrum of the impinging signal in the 2-D case lies on the lines $\Omega_1/\Omega_3 = \sin\theta\cos\phi$ and $\Omega_2/\Omega_3 = \sin\theta\sin\phi$, respectively. Based on the analysis and design of the 1-D case, we perform the 2-D design as follows:

Step 1: Suppose $F(\sin\theta\cos\phi, \sin\theta\sin\phi)$ is the desired beam pattern. With the substitutions $\sin\theta\cos\phi = \Omega_1/\Omega_3$ and $\sin\theta\sin\phi = \Omega_2/\Omega_3$, we obtain $P(\Omega_1,\Omega_2,\Omega_3)$ defined over an interval of one period $\Omega_1,\Omega_2,\Omega_3 \in [-\pi;\pi)$ as

$$P(\Omega_1,\Omega_2,\Omega_3) = \begin{cases} F\left(\dfrac{\Omega_1}{\Omega_3},\dfrac{\Omega_2}{\Omega_3}\right), & \sqrt{\Omega_1^2+\Omega_2^2} \leqslant |\Omega_3| \cap \Omega_3 \neq 0, \\ A(\Omega_1,\Omega_2,\Omega_3) & \text{otherwise}. \end{cases} \tag{9}$$

$A(\Omega_1,\Omega_2,\Omega_3)$ is an arbitrary function to define values for $P(\Omega_1,\Omega_2,\Omega_3)$ and it will not affect the beam pattern because no signal exists in this area.

Step 2: Applying a 3-D inverse Fourier transform (or DFT as an approximation) to $P(\Omega_1,\Omega_2,\Omega_3)$ returns the desired response $d[l,m,n]$. For a causal and practical result, a truncation in the spatial l and m domains and the temporal n domain is necessary with a possible shift in time n prior to truncation in order to ensure a good approximation with a causal response.

3.2. 3-D array

With spatial indices k, l, m, and the temporal index n, the response of a 3-D array is given by

$$P(\omega,\theta,\phi)=\sum_{k,l,m,n=-\infty}^{\infty} d[k,l,m,n]\,\mathrm{e}^{-\mathrm{j}k(\omega\sin\theta\cos\phi/c)d_x} \times \mathrm{e}^{-\mathrm{j}l(\omega\sin\theta\sin\phi/c)d_y}\,\mathrm{e}^{-ml(\omega\cos\theta/c)d_z}\,\mathrm{e}^{-\mathrm{j}n\omega T}. \quad (10)$$

By selecting the array parameters as in the 1-D and 2-D cases, we obtain $d_x=d_y=d_z=cT$, $\Omega=\omega T$. The substitutions $\Omega_1=\Omega\sin\theta\cos\phi$, $\Omega_2=\Omega\sin\theta\sin\phi$, $\Omega_3=\Omega\cos\theta$ and $\Omega_4=\Omega$ applied to (10) yield

$$P(\Omega_1,\Omega_2,\Omega_3,\Omega_4)=\sum_{k,l,m,n=-\infty}^{\infty} d[k,l,m,n] \times \mathrm{e}^{-\mathrm{j}k\Omega_1}\,\mathrm{e}^{-\mathrm{j}l\Omega_2}\,\mathrm{e}^{-\mathrm{j}m\Omega_3}\,\mathrm{e}^{-\mathrm{j}n\Omega_4}. \quad (11)$$

Suppose $F(\sin\theta\cos\phi,\sin\theta\sin\phi,\cos\theta)$ is the desired frequency invariant beam pattern, we can use the substitutions $\sin\theta\cos\phi=\Omega_1/\Omega_4$, $\sin\theta\sin\phi=\Omega_2/\Omega_4$, and $\cos\theta=\Omega_3/\Omega_4$ to get $P(\Omega_1,\Omega_2,\Omega_3,\Omega_4)$. Then by a 4-D inverse Fourier transform, $d[k,l,m,n]$ can be obtained. Analogously to the 2-D array case, a design procedure can be developed for 3-D arrays, for which we omit the details here.

However, there are some other factors to be considered in the design. For a 3-D array, it is very likely that the sensors within the 3-D array will not be able to receive the signals properly from some specific directions as the view of those sensors has been blocked by the sensors at the outer layer. Similarly reflection and diffraction can be a serious problem in the design for a real 3-D array. Simply ignoring these problems in the design may cause unaccounted degradation in the performance. But, when the size of each sensor is very small compared to their spacings, the 3-D design result for the ideal situation may provide a good approximation to the real problem or some useful guidance when we design a 3-D array.

4. Design example and discussions

4.1. Design example

To show the effectiveness of the proposed method, we give a design example for a planar array with 24×24 sensors, each followed by an FIR filter with a length of 24. It comprises of equally spaced sensors with $d_x=d_y=cT=\lambda_{\max}/2$ and has its main beam directed towards broadside. The desired beam pattern is given by

$$F_{2\mathrm{D}}(\sin\theta\cos\phi,\sin\theta\sin\phi) = \frac{1}{49}\sum_{l=-3}^{3}\sum_{m=-3}^{3}\mathrm{e}^{-\mathrm{j}l\pi\sin\theta\cos\phi}\,\mathrm{e}^{-\mathrm{j}m\pi\sin\theta\sin\phi}. \quad (12)$$

It is easy to see that $F_{2\mathrm{D}}(\sin\theta\cos\phi,\sin\theta\sin\phi)$ is the response of a narrowband equispaced planar array with uniform weighting.

According to (9), we first obtain the response $P(\Omega_1,\Omega_2,\Omega_3)$, and then set $A(\Omega_1,\Omega_2,\Omega_3)=0$ for the area outside the beam pattern's domain. A $32\times 32\times 32$ 3-D inverse DFT is employed and the $32\times 32\times 32$ temporal coefficients are subsequently truncated to the required array dimension $24\times 24\times 24$.

The beam pattern resulting from the proposed design are 4-D and it is impossible to show this 4-D pattern in one figure. Instead, some 3-D slices of this response are given below. Figs. 6 and 7 are the planar array's response to the frequencies $\Omega=0.4\pi$ and 0.9π, respectively. The frequency invariant property can be shown by slices of its beam pattern at different values of ϕ, i.e. for each value of ϕ, we draw its response to signals varying over the parameters Ω and θ. Three representative slices are given in Figs. 8, 9 and 10 with $\phi=60^\circ$, 120°, and 180°, respectively, where for $\Omega\geqslant 0.3\pi$ the beam pattern is nearly frequency invariant.

Although the above example is for a response with a broadside main beam, the proposed method can be applied to any desired responses, including the case with either a broadside main beam or an off-broadside main beam, or even no specific beams in any directions. To design a beamformer with an off-broadside main beam, we can simply use a narrowband beamformer design method to obtain such a desired response with the required off-broadside main beam, and then apply the proposed technique exactly in the same way as in the example provided. However, to obtain such an off-broadside main beam, we may need more filter taps because of some discontinuity problem when we apply the inverse Fourier transform. This problem has been addressed in [13] for the 1-D array design case, which is applicable to the 2-D and 3-D cases too.

From the examples shown above, it can be seen that this method cannot achieve a good frequency

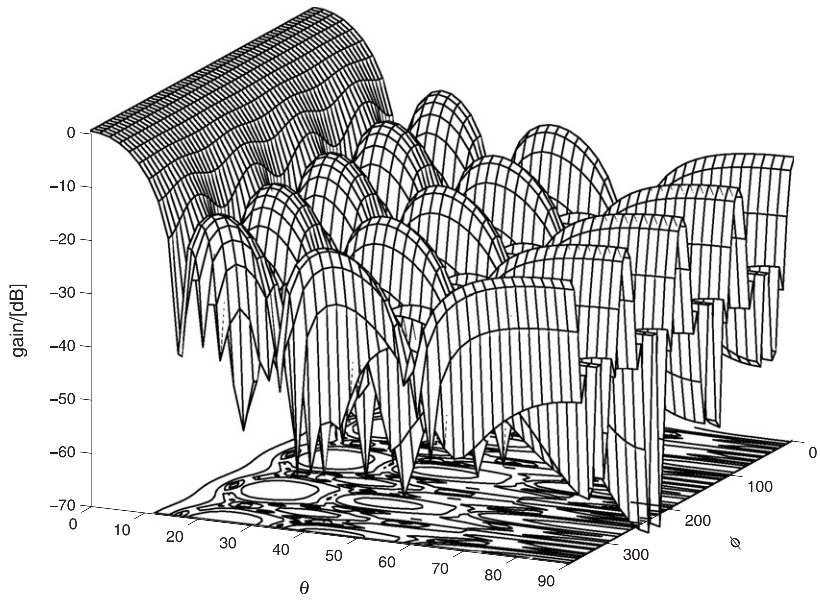

Fig. 6. The resultant beam pattern of the planar array at $\Omega = 0.4\pi$.

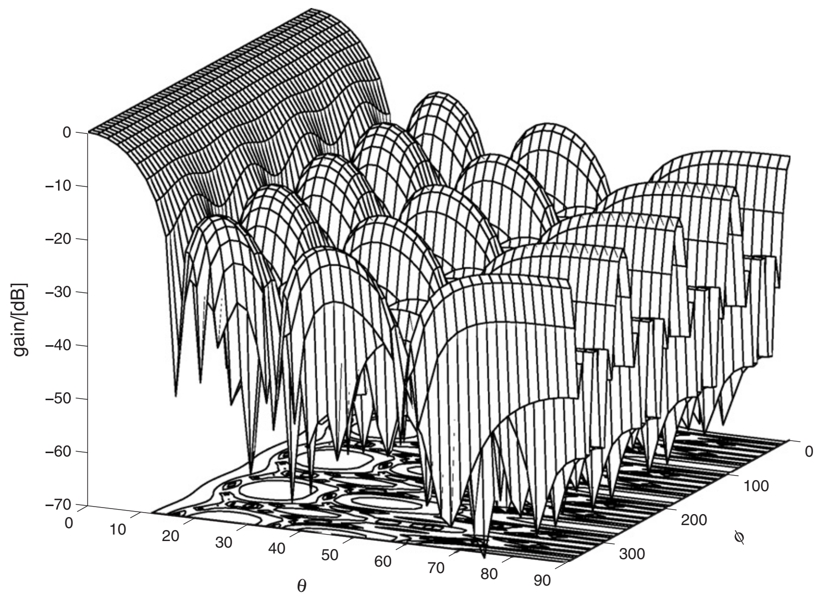

Fig. 7. The resultant beam pattern of the planar array at $\Omega = 0.9\pi$.

invariant property for the lower frequencies. The reason for this phenomenon lies in the fact that the array aperture dimension in terms of the signal wavelength gets smaller as the frequency is reduced. Mathematically, the density of the spatio-temporal spectrum of the input signal is much higher at small values of Ω_3. The low frequency region cannot be represented as sufficiently as the higher frequency part when performing the inverse DFT. To show this, we consider the linear array as an example. According to Fig. 2, the incoming signals for a fixed angle θ lie on one of the dashed lines in Fig. 11. When the inverse DFT is executed in order to calculate the temporal coefficients, we sample the (Ω_1, Ω_2) plane on the grid points given in Fig. 11. Clearly, in the whole shaded

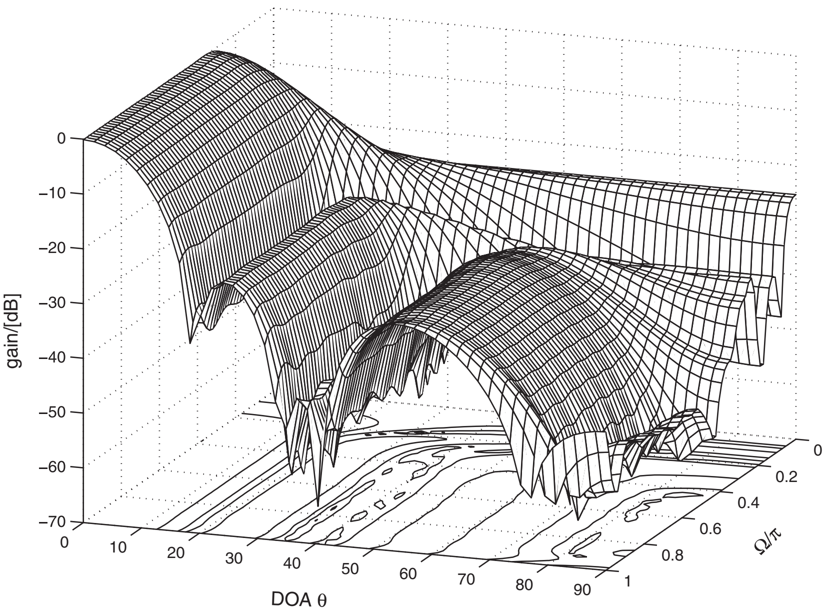

Fig. 8. A slice of the beam pattern at $\phi = 60^{\circ}$.

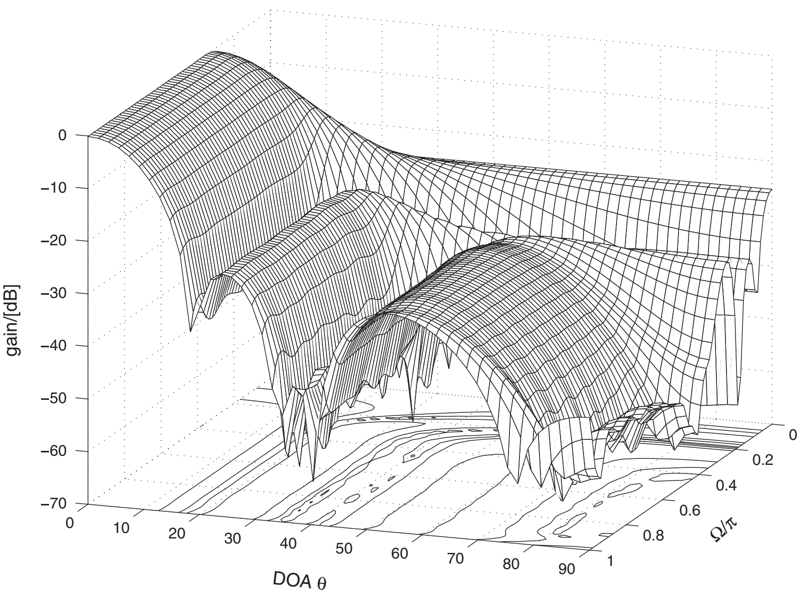

Fig. 9. A slice of the beam pattern at $\phi = 120^{\circ}$.

area, where signals may exist, for low values of Ω_2, the number of acquired samples is much lower than that for large values of Ω_2, which means the sampling resolution decreases with decreasing Ω_2. Therefore, with respect to frequency invariance, at low frequencies the array resulting from the proposed design cannot perform as well as at high frequencies. The DC component corresponds to the single point $(0, 0)$ on the (Ω_1, Ω_2) plane and hence the sampling density for the DC component is always the same and cannot be increased, that is why we cannot achieve a meaningful beam at DC, no matter how large the array dimension is.

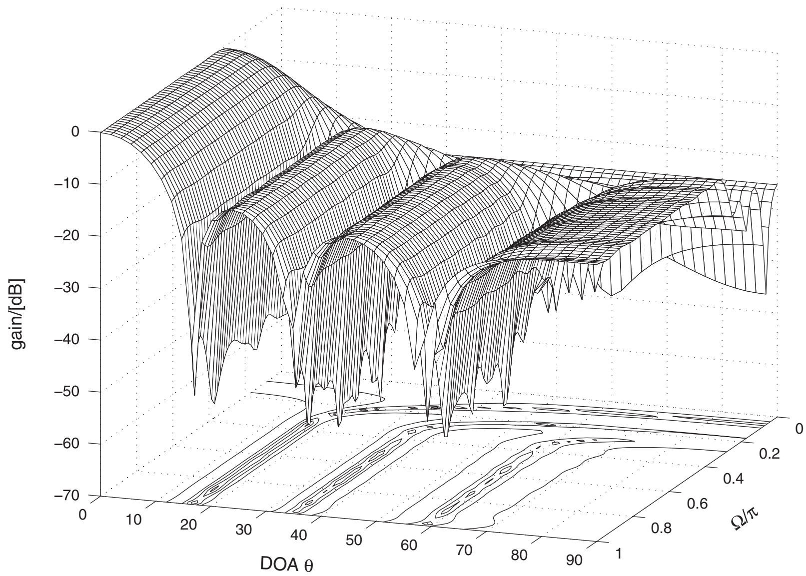

Fig. 10. A slice of the beam pattern at $\phi = 180°$.

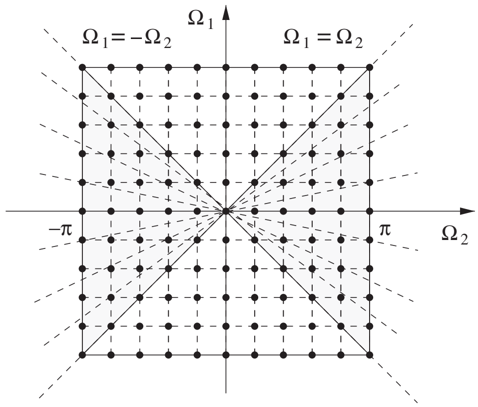

Fig. 11. Sampling of the spatio-temporal spectrum of the signal for a linear array.

Note that the design example provided is based on the normalised frequency, depending on the specific application, they can be transformed into the original frequencies directly using the specified sampling frequency. For example, for a microphone array system [2], suppose the maximum frequency of interest is 4 KHz and the sampling frequency is 8 KHz, then the spacing $d_x = d_y = 34\,000/8000 = 4.25$ cm and this microphone array system will have a good frequency response between $0.3 \times 4 = 1.2$ KHz and 4 KHz, as it has a good frequency invariant response over the bandwidth $[0.3\pi; \pi]$. To extend the frequency invariant range further, e.g. lower than 1.2 KHz, the dimension of the array should be increased accordingly.

4.2. Discussions

4.2.1. Design using Fourier transform

The key to the proposed method is to transform the desired frequency invariant beam pattern $F(\sin\theta)$ (for the case of equally spaced linear array, as an example) into the frequency response $P(\Omega_1, \Omega_2)$ of a normal 2-D digital filter. Thus the design problem is simplified into the design of a 2-D FIR filter for 1-D arrays, 3-D FIR filter for 2-D arrays, and 4-D FIR filter for 3-D arrays. We can employ any available generic FIR design method to obtain the array coefficients and using Fourier transform is just one example. However, it seems that the Fourier transform is the only available generic design method for higher-dimensional FIR filter designs. We use the word "generic", because the desired 2-D, 3-D and 4-D FIR filter response in our proposed method is not a lowpass, highpass or bandpass response, but with a very special shape, as shown in Fig. 4 as an example. We can see from Fig. 4, that the FIR design problem cannot be specified by normal FIR filter design parameters such as the passband attenuation, stopband attenuation and cutoff frequencies, etc.

4.2.2. Choice of the window function for truncation

With the Fourier transform method, it is unavoidable to truncate the obtained 3-D or 4-D FIR filter coefficients to fit the dimension of the 2-D and 3-D arrays. In the example, we used the rectangular window and as shown the result is satisfactory. We tried to use some other window functions, like the hamming window, but we find that it is not as good as the rectangular window. The reason for this is again that the shape of the desired FIR response is not a normal lowpass one, but with a special shape.

4.2.3. Optimum sensor number and tap number for a desired response

Another question is, given the desired response, how many sensors and taps we need to match this response. This is a difficult question and there is no analytical solution to it. By experiment, according to [12], the dimension of the array should be at least three times the dimension of the narrowband beamformer used to provide the desired frequency invariant response. We followed this rule in our design example, where the dimension of the desired response given in (12) is 7×7, and the design result has a dimension of $24 \times 24 \times 24$, which is a little larger than $21 \times 21 \times 21$ ($7 \times 3 = 21$). However, we can see in Figs. 8 and 9 that for lower frequencies $\Omega < 0.3\pi$, the frequency invariant property is not satisfactory. One solution to this problem is that, we sample the desired response in a much larger number and after the inverse Fourier transform, the resultant beamformer will have the same dimension as the samples. We then truncate the results more and more until reaching the point where the desired response cannot be matched within a specified error for the specified frequency range.

5. Conclusions

We have proposed a novel method for the design of 2-D and 3-D frequency invariant beamformers. For completeness, the design of the 1-D case was also included with a generalisation of the previously proposed frequency invariant linear arrays. A design example was provided for the 2-D case, which shows a satisfactory frequency invariant property for a large frequency band.

Acknowledgement

The work undertaken has been sponsored through the U.K. MoD's Corporate Research Programme.

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