ON THE NOISE REDUCTION PERFORMANCE OF THE MVDR BEAMFORMER IN NOISY AND REVERBERANT ENVIRONMENTS

Chao Pan[1], *Jingdong Chen*[1], *and Jacob Benesty*[2]

[1]: Northwestern Polytechnical University
127 Youyi West Road
Xi'an, Shaanxi 710072, China

[2]: INRS-EMT, University of Quebec
800 de la Gauchetiere Ouest, Suite 6900
Montreal, QC H5A 1K6, Canada

ABSTRACT

The minimum variance distortionless response (MVDR) beamformer has been widely studied for extraction of desired speech signals in noisy acoustic environments. The performance of this beamformer, however, depends on many factors such as the array geometry, the source incidence angle, the noise field characteristics, the reverberation conditions, etc. In this paper, we study the performance of the MVDR beamformer in different noise and reverberation conditions with a linear microphone array. Using the gain in signal-to-noise ratio (SNR) as the performance metric, we show that the optimal performance of the MVDR beamformer generally occurs when the source is in the endfire directions in different types of noise, which indicates that, as long as a linear array is used, we should configure it in such a way that the endfire direction is pointed to the desired source. Simulations in reverberant environments also verified this result, though the performance difference between endfire and broadside directions reduces as the degree of reverberation increases.

Index Terms—Beamforming, microphone arrays, minimum variance distortionless response (MVDR) beamformer, noise reduction.

1. MVDR BEAMFORMER

The MVDR beamformer can be derived in several different ways, generally based on an ideal signal model without considering the multipath or reverberation effect.

1.1. Signal Model

Let us first consider a simple signal model where a desired speech source (plane wave) propagates in an anechoic acoustic environment and impinges on a uniform linear array consisting of M omnidirectional microphones, as shown in Fig. 1. Let us choose the first microphone as the reference point, the signal received by the mth microphone ($m = 1, 2, \dots, M$) can then be written as [1]

$$y_m(t) = x_m(t) + v_m(t) = x(t - \tau_m) + v_m(t), \qquad (1)$$

where $y_m(t)$, $x_m(t)$, and $v_m(t)$ are the noisy, clean speech, and noise signals, respectively, captured by the mth microphone at time t, $\tau_m = (m-1)\tau_0 \cos\theta_\mathrm{d}$ is the relative time delay between the mth microphone and the reference sensor, $\tau_0 = \delta/c$ with δ being the spacing between two neighboring sensors and c being the speed of sound in air, i.e., $c = 340$ m/s, θ_d is the incidence angle of the desired sound source, and $x(t) = x_1(t)$ is the clean signal received at the reference microphone. All signals are considered to be zero-mean, real, and broadband. Furthermore, the noise signals $v_m(t),\ m = 1, 2, \dots, M$, are assumed to be uncorrelated with the clean signals $x_m(t),\ m = 1, 2, \dots, M$.

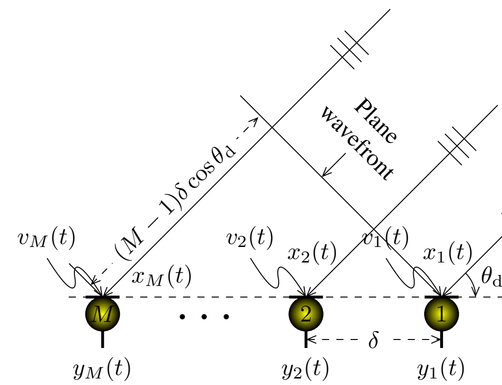

Fig. 1. Illustration of a uniform linear microphone array system, where M is the number of microphones, δ is the microphone spacing, and θ_d is the incidence angle of the desired source which is located in the far field.

To make the processing efficient, we study the beamforming problem in the frequency domain. In this domain, the signal model given in (1) is written as

$$\begin{aligned} Y_m(\omega) &= X_m(\omega) + V_m(\omega) \\ &= e^{-\jmath(m-1)\omega\tau_0\cos\theta_\mathrm{d}} X(\omega) + V_m(\omega), \end{aligned} \qquad (2)$$

where $Y_m(\omega)$, $X_m(\omega)$, $V_m(\omega)$, and $X(\omega)$ are the Fourier transforms of $y_m(t)$, $x_m(t)$, $v_m(t)$, and $x(t)$, respectively, $\jmath$ is the imaginary unit, i.e., $\jmath^2 = -1$, $\omega = 2\pi f$ is the angular frequency, and f (> 0) denotes the temporal frequency. We can rearrange (2) into the following vector form:

$$\begin{aligned} \mathbf{y}(\omega) &\triangleq \begin{bmatrix} Y_1(\omega) & Y_2(\omega) & \cdots & Y_M(\omega) \end{bmatrix}^T \\ &= \mathbf{d}_{\theta_\mathrm{d}}(\omega) X(\omega) + \mathbf{v}(\omega), \end{aligned} \qquad (3)$$

where the superscript ^{T} is the transpose operator,

$$\mathbf{d}_{\theta_\mathrm{d}}(\omega) \triangleq \begin{bmatrix} 1 & e^{-\jmath\omega\tau_0\cos\theta_\mathrm{d}} & \cdots & e^{-\jmath(M-1)\omega\tau_0\cos\theta_\mathrm{d}} \end{bmatrix}^T \qquad (4)$$

is the steering vector, and the noise signal vector, $\mathbf{v}(\omega)$, is defined in a similar manner to $\mathbf{y}(\omega)$.

1.2. MVDR Beamformer

The objective of beamforming is to extract the desired source signal, $X_1(\omega)$, from the observations by applying a linear filter, $\mathbf{h}(\omega)$, to $\mathbf{y}(\omega)$, i.e.,

$$\begin{aligned} Z(\omega) &= \mathbf{h}^H(\omega)\mathbf{y}(\omega) = \mathbf{h}^H(\omega)\mathbf{x}(\omega) + \mathbf{h}^H(\omega)\mathbf{v}(\omega) \\ &= \mathbf{h}^H(\omega)\mathbf{d}_{\theta_\mathrm{d}}(\omega) X_1(\omega) + \mathbf{h}^H(\omega)\mathbf{v}(\omega), \end{aligned} \qquad (5)$$

where $Z(\omega)$ is the output of the beamformer, $\mathbf{h}^H(\omega)\mathbf{x}(\omega)$ is the filtered speech signal, and $\mathbf{h}^H(\omega)\mathbf{v}(\omega)$ is the residual noise.

The MVDR beamformer can be derived by minimizing the variance of either the beamformer's output, i.e., $Z(\omega)$, or the residual noise, i.e., $\mathbf{h}^H(\omega)\mathbf{v}(\omega)$, with the constraint that the signal from the desired look direction is passed through without any distortion. Let us consider, in this paper, the minimization of the variance of the residual noise. The problem is then written as

$$\min_{\mathbf{h}(\omega)} E\left[\left|\mathbf{h}^H(\omega)\mathbf{v}(\omega)\right|^2\right] \text{ subject to } \mathbf{h}^H(\omega)\mathbf{d}_{\theta_\mathrm{d}}(\omega) = 1, \quad (6)$$

$E[\cdot]$ denotes mathematical expectation. Using a Lagrange multiplier to adjoin the constraint to the objective function, then differentiating with respect to $\mathbf{h}(\omega)$, and equating the result to zero, we deduce the solution to (6) as

$$\mathbf{h}_{\theta_\mathrm{d}}(\omega) = \frac{\mathbf{\Gamma}_\mathbf{v}^{-1}(\omega)\,\mathbf{d}_{\theta_\mathrm{d}}(\omega)}{\mathbf{d}_{\theta_\mathrm{d}}^H(\omega)\,\mathbf{\Gamma}_\mathbf{v}^{-1}(\omega)\,\mathbf{d}_{\theta_\mathrm{d}}(\omega)}, \quad (7)$$

where $\mathbf{\Gamma}_\mathbf{v}(\omega) = \mathbf{\Phi}_\mathbf{v}(\omega)/\phi_{V_1}(\omega)$ is the pseudo-coherence matrix of the noise, with $\mathbf{\Phi}_\mathbf{v}(\omega) = E\left[\mathbf{v}(\omega)\mathbf{v}^H(\omega)\right]$ and $\phi_{V_1}(\omega) = E\left[|V_1(\omega)|^2\right]$

It is seen that the MVDR beamformer is a function of two terms. One is the steering vector corresponding to the desired signal, which is in turn a function of the incidence angle of the desired signal, and the other is the pseudo-coherence matrix of the noise. Consequently, we should expect that the performance of this beamformer would heavily depend on the source incidence angle and the noise characteristics, which will be discussed in the next section.

2. PERFORMANCE OF THE MVDR BEAMFORMER IN DIFFERENT NOISE ENVIRONMENTS

In this section, we analyze the performance of the MVDR beamformer by examining the SNR gain in four typical noise environments: spatially white, diffuse, diffuse-plus-white, and point-source-plus-white noise.

2.1. SNR Gain

The input SNR of an array is defined as the SNR at the reference sensor. With the signal model in (2), the input SNR is written as

$$\mathrm{iSNR}(\omega) \triangleq \frac{\phi_X(\omega)}{\phi_{V_1}(\omega)}, \quad (8)$$

where $\phi_X(\omega) = E\left[|X(\omega)|^2\right]$ is the variance of $X(\omega)$.

With the beamformer's output given in (5), the output SNR is defined as

$$\begin{aligned}\mathrm{oSNR}[\mathbf{h}(\omega)] &= \frac{E\left[\left|\mathbf{h}^H(\omega)\mathbf{d}_{\theta_\mathrm{d}}(\omega)X(\omega)\right|^2\right]}{E\left[\left|\mathbf{h}^H(\omega)\mathbf{v}(\omega)\right|^2\right]} \\ &= \mathrm{iSNR}(\omega) \times \frac{\left|\mathbf{h}^H(\omega)\,\mathbf{d}_{\theta_\mathrm{d}}(\omega)\right|^2}{\mathbf{h}^H(\omega)\,\mathbf{\Gamma}_\mathbf{v}(\omega)\,\mathbf{h}(\omega)}, \quad (9)\end{aligned}$$

which depends on the input SNR, the signal incidence angle, the beamforming filter, as well as the pseudo-coherence matrix of the noise.

The definition of the SNR gain is easily derived from (8) and (9), i.e.,

$$\mathcal{G}[\mathbf{h}(\omega)] = \frac{\mathrm{oSNR}[\mathbf{h}(\omega)]}{\mathrm{iSNR}(\omega)} = \frac{\left|\mathbf{h}^H(\omega)\,\mathbf{d}_{\theta_\mathrm{d}}(\omega)\right|^2}{\mathbf{h}^H(\omega)\,\mathbf{\Gamma}_\mathbf{v}(\omega)\,\mathbf{h}(\omega)}. \quad (10)$$

Substituting the MVDR filter, $\mathbf{h}_{\theta_\mathrm{d}}(\omega)$, given in (7) into (10), we deduce that

$$\begin{aligned}\mathcal{G}[\mathbf{h}_{\theta_\mathrm{d}}(\omega)] &= \frac{\left|\mathbf{h}^H(\omega)\,\mathbf{d}_{\theta_\mathrm{d}}(\omega)\right|^2}{\mathbf{h}^H(\omega)\,\mathbf{\Gamma}_\mathbf{v}(\omega)\,\mathbf{h}(\omega)} \\ &= \mathbf{d}_{\theta_\mathrm{d}}^H(\omega)\,\mathbf{\Gamma}_\mathbf{v}^{-1}(\omega)\,\mathbf{d}_{\theta_\mathrm{d}}(\omega), \quad (11)\end{aligned}$$

which is now explicitly dependent on the signal incidence angle and the pseudo-coherence matrix of the noise.

2.2. SNR Gain in Spatially White Noise

If the noise is spatially white, the pseudo-coherence matrix can be written as

$$\mathbf{\Gamma}_\mathbf{v}(\omega) = \mathbf{\Gamma}_\mathrm{wn}(\omega) = \mathbf{I}_M, \quad (12)$$

which is just the $M \times M$ identity matrix. The corresponding SNR gain of the MVDR beamformer is then

$$\mathcal{G}[\mathbf{h}_{\theta_\mathrm{d}}(\omega)] = M, \quad (13)$$

which is a constant and independent of the incidence angle of the desired source.

2.3. SNR Gain in Diffuse Noise

In an acoustic enclosure such as a room, the noise may have an energy flow of equal probability in all directions due to the multipath effect and reverberation, leading to a diffuse noise field [2]–[4]. In this scenario, we have $\mathbf{\Gamma}_\mathbf{v}(\omega) = \mathbf{\Gamma}_\mathrm{dn}(\omega)$, with

$$[\mathbf{\Gamma}_\mathrm{dn}(\omega)]_{ij} = \mathrm{sinc}[\omega\tau_0(j-i)] = \frac{\sin[\omega\tau_0(j-i)]}{\omega\tau_0(j-i)}, \quad (14)$$

where $[\mathbf{\Gamma}_\mathrm{dn}(\omega)]_{ij}$ is the (i, j)th element of the matrix $\mathbf{\Gamma}_\mathrm{dn}(\omega)$. There are two extreme cases: 1) if $\omega\tau_0$ is very large, e.g., high frequencies or large spacing, the noise signals observed by two sensors tend to be uncorrelated, and then the diffuse noise field is close to the spatially white noise field; 2) if $\omega\tau_0$ is very small, e.g., low frequencies or small spacing, the noise signals observed by two sensors tend to be coherent.

Substituting (14) into (11), one can obtain the SNR gain of the MVDR beamformer in diffuse noise, which is very difficult to write into an analytic form. However, if we plot the SNR gain as a function of the source incidence angle and frequency, one can easily see that the SNR gain reaches its maximum in the endfire directions.

2.4. SNR Gain in Diffuse-plus-White Noise

In most of the practical acoustic environments, there might be presence of both diffuse and uncorrelated white noise. In this case, the noise pseudo-coherence matrix can be written as

$$\mathbf{\Gamma}_\mathbf{v}(\omega) = \mathbf{\Gamma}_\mathrm{dwn}(\omega) = (1-\alpha_\mathrm{dn})\mathbf{I}_M + \alpha_\mathrm{dn}\mathbf{\Gamma}_\mathrm{dn}(\omega), \quad (15)$$

where α_dn $(0 \le \alpha_\mathrm{dn} \le 1)$ is a constant that specifies the level of the diffuse noise relative to the spatially white noise. The corresponding SNR gain also reaches its maximum in endfire directions.

2.5. SNR Gain in Point-Source-plus-White Noise

In many application scenarios, there may be competing sources. In this subsection, we consider the case where there is a point noise source in addition to the spatially white noise. Assuming that the incidence angle of the point noise source is θ_n, the corresponding pseudo-coherence matrix can be written as

$$\mathbf{\Gamma}_\mathrm{psn}(\omega) = \mathbf{d}_{\theta_\mathrm{n}}(\omega)\,\mathbf{d}_{\theta_\mathrm{n}}^H(\omega), \quad (16)$$

where $\mathbf{d}_{\theta_{\mathrm{n}}}(\omega)$ is the steering vector of the point noise source, which is defined in a similar way to $\mathbf{d}_{\theta}(\omega)$. Then, the pseudo-coherence matrix of the point-source-plus-white noise is

$$\mathbf{\Gamma}_{\mathrm{pswn}}(\omega) = (1-\alpha_{\mathrm{psn}})\mathbf{I}_M + \alpha_{\mathrm{psn}}\mathbf{\Gamma}_{\mathrm{psn}}(\omega). \quad (17)$$

where α_{psn} ($0 \le \alpha_{\mathrm{psn}} < 1$) is a parameter that controls the level of the point source noise relative to that of the spatially white noise. By utilizing the Woodbury's identity, we can write the inverse of the pseudo-coherence matrix as

$$\mathbf{\Gamma}_{\mathrm{pswn}}^{-1}(\omega) = \frac{1}{1-\alpha_{\mathrm{psn}}}\left[\mathbf{I}_M - \frac{\mathbf{d}_{\theta_{\mathrm{n}}}(\omega)\,\mathbf{d}_{\theta_{\mathrm{n}}}^H(\omega)}{(1-\alpha_{\mathrm{psn}})/\alpha_{\mathrm{psn}} + M}\right]. \quad (18)$$

Substituting (18) into (11), we can derive the SNR gain:

$$\mathcal{G}\left[\mathbf{h}_{\theta_{\mathrm{d}}}(\omega)\right] = \frac{1}{1-\alpha_{\mathrm{psn}}}\left[M - \frac{\left|\mathbf{d}_{\theta_{\mathrm{n}}}^H(\omega)\,\mathbf{d}_{\theta_{\mathrm{d}}}(\omega)\right|^2}{(1-\alpha_{\mathrm{psn}})/\alpha_{\mathrm{psn}} + M}\right]. \quad (19)$$

If the value of α_{psn} is fixed, the minimum of the gain in (19) occurs when the desired signal and the point source noise come from the same direction, i.e., $\theta_{\mathrm{n}} = \theta_d$. In this case, only the white noise is reduced while the point noise is not changed and the corresponding SNR gain is

$$\mathcal{G}\left[\mathbf{h}_{\theta_{\mathrm{d}}}(\omega)\right] = \frac{M}{1+(M-1)\,\alpha_{\mathrm{psn}}}. \quad (20)$$

The maximum of the SNR gain occurs if the steering vectors of the desired and point noise sources are orthogonal, i.e., $\mathbf{d}_{\theta_{\mathrm{n}}}^H(\omega)\,\mathbf{d}_{\theta_{\mathrm{d}}}(\omega) = 0$. In this situation, the point noise is completely removed and the corresponding SNR gain is

$$\mathcal{G}\left[\mathbf{h}_{\theta_{\mathrm{d}}}(\omega)\right] = \frac{M}{1-\alpha_{\mathrm{psn}}}. \quad (21)$$

We have discussed the SNR gain of the MVDR beamformer in four typical noise environments. Figure 2 plots the SNR gain as a function of the source incidence angle θ_{d} for a microphone array with $M = 10$ and $\delta = 2$ cm. As seen, the highest SNR gain appears at the endfire directions.

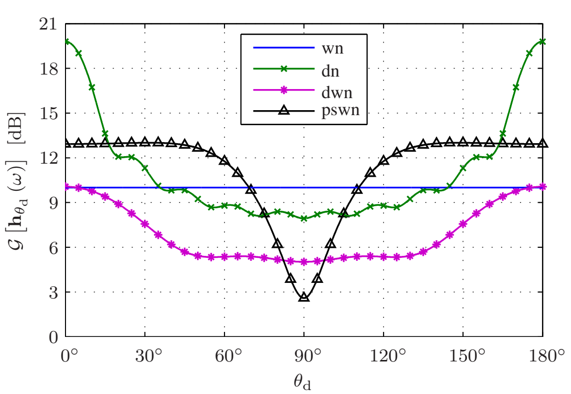

Fig. 2. The SNR gain of the MVDR beamformer in different noise environments, where $M = 10$, $\delta = 2$ cm, $f = 2$ kHz, $\alpha_{\mathrm{dn}} = 0.9$, $\alpha_{\mathrm{psn}} = 0.5$, and $\theta_{\mathrm{n}} = 90^\circ$.

3. PERFORMANCE STUDY IN NOISY AND REVERBERANT ENVIRONMENTS

It was shown in (11) that the SNR gain of the MVDR beamformer is an explicit function of the source incidence angle and the noise characteristics. There is another very important factor that would dramatically affect the performance of this beamformer but is not explicitly shown in (11). That is the reverberation. When there is reverberation, the microphone array outputs are no longer in the simple form as in (1), but in a more complicated form as [1]

$$\begin{aligned} y_m(t) &= g_m(t) * s(t) + v_m(t) \\ &= x_m(t) + v_m(t),\ m = 1, 2, \ldots, M, \end{aligned} \quad (22)$$

where $g_m(t)$ is the impulse response from the desired source, $s(t)$, to the mth microphone and $*$ denotes linear convolution. The corresponding frequency-domain counterpart is written as

$$\begin{aligned} Y_m(\omega) &= G_m(\omega)S(\omega) + V_m(\omega) \\ &= X_m(\omega) + V_m(\omega),\ m = 1, 2, \ldots, M, \end{aligned} \quad (23)$$

where $G_m(\omega)$ and $S(\omega)$ are the Fourier transforms of $g_m(t)$ and $s(t)$, respectively. Reverberation affects the MVDR beamformer by changing the form of the source steering vector, which is no longer a simple function of the incidence angle θ_{d}. However, a theoretical analysis of the reverberation effect on the SNR gain of the MVDR beamformer can be very difficult if not impossible. In this section, we investigate the SNR gain of the MVDR beamformer in reverberant environments through simulations.

3.1. Simulation Setup

A 10-element linear microphone array is placed in the center of a reverberant room of size 3 m$\times$3 m$\times$3 m. A loudspeaker is placed at $(2.5, 1.5, 1.5)$, playing back a pre-recorded speech signal to simulate desired speech source. The speech signal was recorded in a quiet room with a sampling rate of 8 kHz. The length of this signal is 25 s. Both the microphone array and the loudspeaker are on the horizontal plane at $z = 1.5$ m. The spacing between two neighboring sensors is 2 cm. The microphone array outputs are generated by convolving the source signal with the impulse responses from the loudspeaker to the microphone sensors. The impulse responses are generated with the well-known image-model method [5], [6]. Noise is then added to the convolved speech to control the input SNR (10 dB in this paper). The reflection coefficients of all the six walls are set to be identical. We vary these coefficients from 0 to 1 to control the reverberation time, T_{60}. Fixing the source position, and rotating the array clockwise with respect to the array center, we evaluate the fullband SNR gain of the MVDR beamformer as a function of the source incidence angle in different reverberation conditions.

3.2. Fullband SNR Gain of the MVDR Beamformer

In the previous section, the SNR gain is defined and evaluated on a narrowband basis. In this section, we analyze the fullband SNR gain in the time domain, which is defined as

$$\mathcal{G}_{\theta_{\mathrm{d}}} \triangleq \frac{\mathrm{oSNR}}{\mathrm{iSNR}}, \quad (24)$$

where $\mathrm{iSNR} \triangleq E\left[x_1^2(t)\right]/E\left[v_1^2(t)\right]$ is the fullband input SNR and $\mathrm{oSNR} \triangleq E\left[x_{\mathrm{fd}}^2(t)\right]/E\left[v_{\mathrm{rn}}^2(t)\right]$ is the fullband output SNR with $x_{\mathrm{fd}}(t)$ and $v_{\mathrm{rn}}(t)$ being, respectively, the time-domain filtered desired signal and residual noise reconstructed from $\mathbf{h}_{\theta_{\mathrm{d}}}^H(\omega)\mathbf{x}(\omega)$ and $\mathbf{h}_{\theta_{\mathrm{d}}}^H(\omega)\mathbf{v}(\omega)$, respectively.

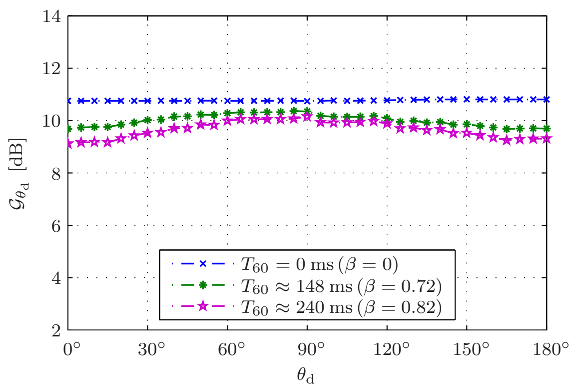

Fig. 3. Fullband SNR gain in spatially white noise.

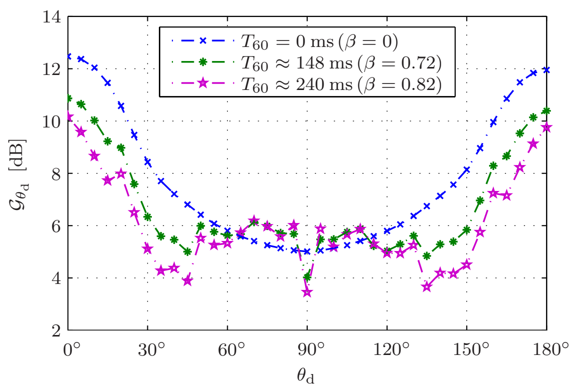

Fig. 4. Fullband SNR gain in diffuse noise.

3.2.1. Performance in Reverberation and Spatially White Noise

The fullband SNR gain as a function of the source incidence angle in spatially white noise with three different reverberation conditions is plotted in Fig. 3 where the spatially white noise is generated using the Matlab randn function. One can see that SNR gain slightly decreases as the environment becomes more reverberant (i.e., reverberation time increases) given an incidence angle. In each reverberation condition, the gain does not change much with the source incidence angle. We notice that when there is reverberation (e.g., ($T_{60} \approx 148$ ms and $T_{60} \approx 240$ ms), the MVDR beamformer has a slightly smaller SNR gain at the endfire directions. This is mainly due to the fact the filtered signal has a smaller variance since more reflected signals are rejected in endfire directions.

3.2.2. Performance in Reverberation and Diffuse Noise

The fullband SNR gain as a function of the source incidence angle in diffuse noise with three different reverberation conditions is sketched in Fig. 4, where the diffuse noise is generated by the method presented in [7], which sums 100 point sources (each source is a white noise) uniformly distributed on the surface of a sphere around the array. It is seen that the maximum SNR gain appears in the endfire directions regardless of the degree of reverberation. But the performance difference between endfire and broadside directions reduces as the degree of reverberation increases.

3.2.3. Performance in Reverberation and Diffuse-plus-White Noise

The SNR gain as a function of the source incidence angle in three different reverberation and a diffuse-plus-white noise (with $\alpha_{\rm dn} = 0.9$) conditions is plotted in Fig. 5. Again, the maximum SNR gain occurs in the endfire directions. It is also observed that the SNR gain decreases as the reverberation time increases.

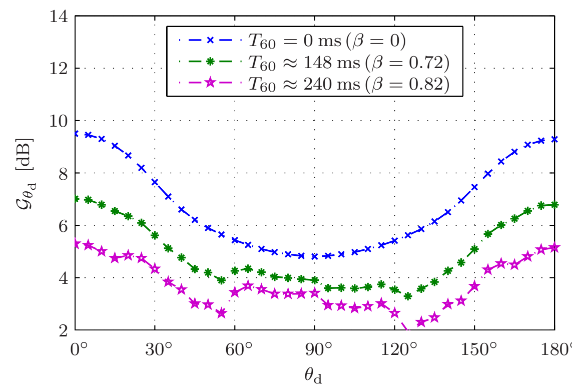

Fig. 5. Fullband SNR gain in diffuse-plus-white noise.

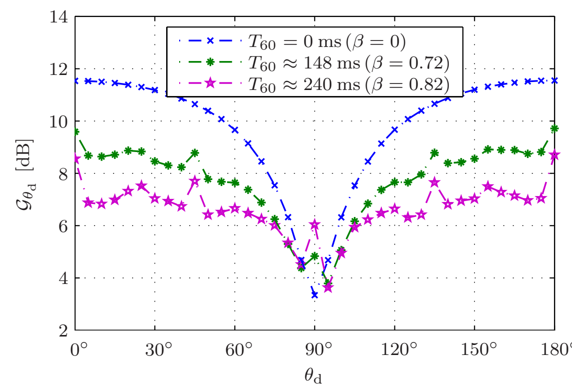

Fig. 6. Fullband SNR gain in point-source-plus-white noise.

3.2.4. Performance in Reverberation and Point-Source-plus-White Noise

The last experiment is concerned with the SNR gain in reverberant environments with a point-source-plus-white noise ($\theta_{\rm n} = 90^\circ$ and $\alpha_{\rm psn} = 0.5$). The result is shown in Fig. 6. Now the SNR gain depends on the separation between the source and point noise incidence angles. One can notice that reverberation affects the performance of the MVDR beamformer significantly.

4. CONCLUSIONS

This paper investigated the noise reduction performance of the MVDR beamformer in different noise and reverberation conditions. With a linear microphone array, we made the following observations. 1) The SNR improvement may change significantly with the noise characteristics. 2) The SNR gain is generally a function of the source incidence angle; as far as a linear array is concerned, the best SNR gain generally occurs in the endfire directions. 3) Reverberation does not only decrease the SNR gain of the MVDR beamformer, but also reduces the performance differences from endfire to broadside directions.

5. RELATION TO PRIOR WORK

The MVDR beamformer, which was first proposed by Capon [8], has been widely studied over the past decades to extract the signal of interest in noisy environments [1] [9]– [13]. In acoustic environments, it was found that reverberation plays a significant role on the performance of the MVDR beamformer [14], [15]. Recently, we showed that the performance of the MVDR beamformer in noise is also a function of the source incidence angle [16]. This paper presents our continued efforts in studying the MVDR beamformer for noise reduction. The focus is on the effect of the source incidence angle, noise, and reverberation on the MVDR beamformer's performance.

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