FILTER-AND-SUM BEAMFORMER WITH ADJUSTABLE FILTER CHARACTERISTICS

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ABSTRACT

In this paper we introduce a polynomial filter structure for filter-and-sum beamforming applied to microphone array application. The structure is a multi-dimensional extension of well-known Farrow structure, which has mainly been used for fractional delay filtering and interpolation of 1-D signals. The proposed method enables an easy, smooth, and efficient control of beamforming filter characteristic by adjusting only a single control variable e.g. for dynamic beam steering. The optimization method for polynomial beamforming filter design is presented and illustrated with simulations of beamforming filter characteristics. The design example is given for a linear array of four omni-directional microphones and a polynomial FIR filter with 20-tap delay lines.

1. INTRODUCTION

Beamforming systems are commonly used for improving the quality of a received signal. Typical applications can be found in radio communications, radar signal processing, underwater acoustics, and speech acquisition for teleconferencing and hands-free systems. Currently, the most frequently studied beamforming methods are based on the Griffiths-Jim generalized sidelobe canceller (GSC) [1] that is derived from the Frost's linearly constrained adaptive beamformer [2]. The fundamental problem of these adaptive beamformers is that they may cause unpredictable distortion of the desired signal [3]. This is usually caused by multi-path propagation of the desired signal, misaligned look direction, hardware impairments, or variations in the propagation medium.

Various improvements to minimize the desired signal distortion have been proposed [4, 5, 6, 7], but the robustness is typically achieved at the expense of the interference cancellation performance. In highly reverberant environments and under the influence of diffuse noise field, like inside a car cabin, the interfering signals propagate towards the array in many directions and the adaptive interference canceller cannot perform better than an optimized constant beamformer [8, 9, 10]. Thus, in order to alleviate the desired signal distortion in the reverberant environments, it is more favorable to design an optimal filter-and-sum beamformer for a given application.

The advantage of constant beamforming is that filtering performance is deterministic and the filter coefficients can be optimized for given performance criteria utilizing all available information about the signal sources and the acoustic environment. We have previously shown [11] that several parameters like transducer positions, filter length and filter coefficients of a filter-and-sum beamformer can jointly be optimized for various predefined filter characteristics such as look direction, beam shape, signal bandwidth, etc. Since the directional sensitivity of the resulting beamformer is invariant in time, it may be necessary to optimize the beamforming filters in advance for several different filter characteristics, e.g. for different look directions. Thus, the collection of optimal filters would have to be stored in a memory and continuously updated by retrieving filter coefficients from the memory, e.g. to support steering of the look direction towards a moving signal source. In adaptive beamforming, separate beam steering delays have been proposed for the beamformer front-end to align the input signals in such a way that the target signal enters the beamforming filter in phase [12]. However, in case of an optimized constant filter-and-sum beamformer, the usage of steering delays in front of the beamforming filter leads to a suboptimal performance, since the steering delays distort the spatio-temporal sampling grid.

In order to reduce the amount of memory required for different filter characteristics and to simplify the beam steering procedure, we introduce a polynomial beamforming filter based on the well-known Farrow structure [13, 14, 15]. The objective of the proposed method is to offer an efficient adaptation to certain predetermined changes in the environment by using a single control variable.

This paper is organized as follows. In Section 2 we derive the polynomial model of a filter-and-sum beamformer and in Section 3, we provide simulation results for a beamforming filter design with a dynamic look direction control. Finally we summarize our major findings and outline our future work.

2. FILTER-AND-SUM BEAMFORMER

2.1. Input signals

In this work, we study the performance of the proposed polynomial beamforming filter using a linear array of omni-directional microphones as an example (see Fig. 1). Assuming that signal sources are point sources pulsating harmonic spherical waves [11] in a lossless medium a single point source ψ is uniquely defined by a pair (S_ψ, f_ψ), where S_ψ is the source location and f_ψ is the signal frequency. Let us approximate the sound field at position M_i with a discrete set Ψ of point sources $\psi \in \Psi$. We can now write the output of an ideal omni-directional microphone at location M_i as

$$x_i(t) = \sum_{\psi \in \Psi} \frac{V}{r_\psi} \cos\left(2\pi f_\psi \left(t - \frac{r_\psi}{c}\right)\right), \tag{1}$$

where c is the speed of sound, Ψ is a set of source signals, r_ψ is the Euclidean distance from the source location S_ψ to the microphone position M_i. In this paper, we use a 1-D linear array as an example. Thus, a source location S in 3-D space can be defined using the distance r and direction ϕ. More complex 2-D and 3-D array

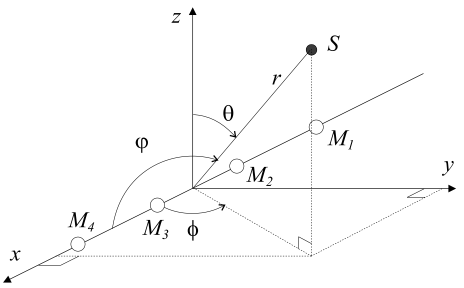

Fig. 1. Linear array of four microphones M_i and a single point source S.

structures even with directional microphones may be used for better localization of sound sources. The array geometry is illustrated in Fig. 1. Synchronous sampling of the microphone signals $x_i(t)$ in (1) on a sampling frequency Fs produces a spatial sample vector $[x_1(n), \ldots, x_M(n)]^T$ at each time instant $n = t \cdot \text{Fs}$.

2.2. Filter-and-sum beamforming filter structure

We have previously developed a method to optimize the directional sensitivity of a filter-and-sum beamformer [11] consisting of M FIR filters of length L. In Fig. 2, we show the filter-and-sum beam-

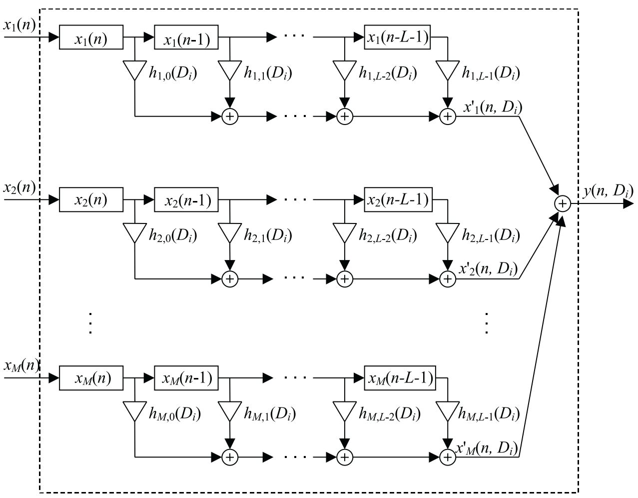

Fig. 2. Filter-and-sum beamforming filter structure.

forming filter structure in which the coefficients $h_{j,k}(D_i)$ and the output signal

$$y(n, D_i) = \sum_{j=1}^{M} \sum_{k=0}^{L-1} h_{j,k}(D_i) x_j(n-k), \tag{2}$$

are represented as a function of the control variable D_i. For example, if D_i denotes a desired look direction, the filter coefficients $h_{j,k}(D_i)$ can be separately optimized for N different look directions D_i, $i = 1, 2, ..., N$. Therefore, we have to save $N{\cdot}M{\cdot}L$ filter coefficients in memory. Also, processing the output signal (2) for a look direction D_i requires $M{\cdot}L$ multiply-and-add operations per sampling interval 1/ Fs.

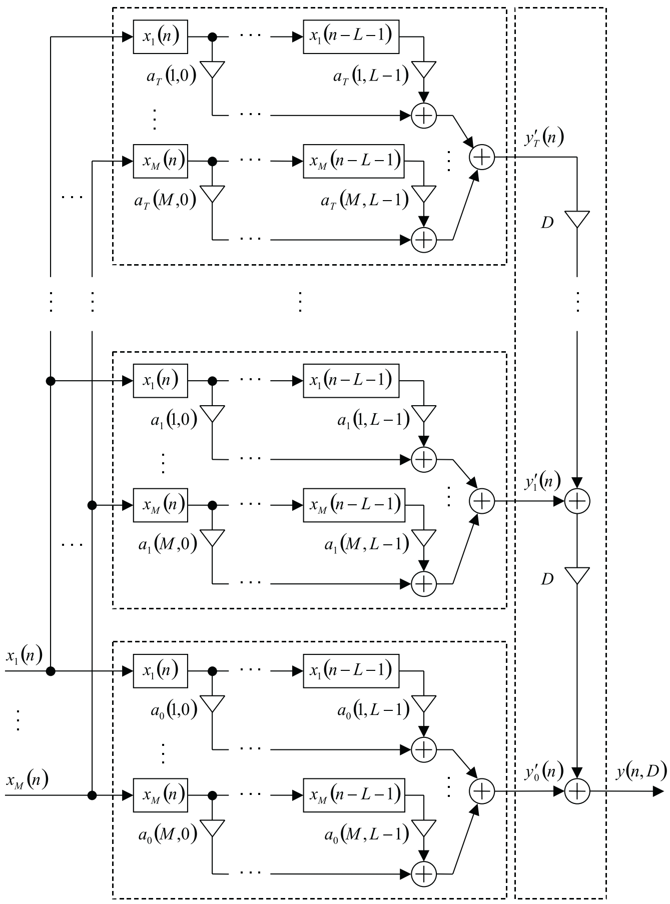

Fig. 3. Polynomial beamforming filter structure.

2.3. Polynomial beamforming filter structure

Let us now introduce a polynomial FIR filter representation

$$h_{j,k}(D) = a_0(j,k) + a_1(j,k)D + \cdots + a_T(j,k)D^T. \tag{3}$$

It is also possible to generalize (3) by replacing $\{D^t\}$ with an alternative function basis $\{F_t(D)\}$. The polynomial beamforming filter structure (Fig. 3) can be derived from the conventional structure (Fig. 2) as follows. Firstly, by replacing $h_{j,k}(D_i)$ in (2) with $h_{j,k}(D)$ in (3), we get

$$y(n, D) = \sum_{j=1}^{M} \sum_{k=0}^{L-1} \sum_{t=0}^{T} a_t(j,k) D^t x_j(n-k). \tag{4}$$

Changing the order of the terms in (4) we can write

$$y(n, D) = \sum_{t=0}^{T} D^t \sum_{j=1}^{M} \sum_{k=0}^{L-1} a_t(j,k) x_j(n-k). \tag{5}$$

Finally, by defining the intermediate output signals

$$y'_t = \sum_{j=1}^{M} \sum_{k=0}^{L-1} a_t(j,k) x_j(n-k), \tag{6}$$

the beamformer output in (5) can be written as

$$y(n, D) = \sum_{t=0}^{T} D^t y_t'(n). \tag{7}$$

Thus, the proposed beamforming filter structure in Fig. 3 consists of two parts: $T + 1$ fixed prefilters (6) and a polynomial postfilter (7). Since we use a continuous control variable D to parameterize the output of the polynomial filter (7), we achieve optimal filter characteristics that can be dynamically adjusted in a range $D_{min} \leq D \leq D_{max}$.

The polynomial filter structure is easily expanded to provide P parallel output signals (7) simply by processing P copies of the polynomial postfilter. Kellermann has proposed the usage of multiple parallel beamforming filters for robust speaker tracking [16]. Such a system would be easy to implement using the polynomial filter structure.

The memory requirement to save a priori optimized fixed filter parameters $a_t(j, k)$ is $(T + 1) \cdot M \cdot L + P$ for P parallel output signals. In practice, if $N > T + 1$, the proposed structure requires less memory than the conventional filter-and-sum approach using several parallel constant beamformers. The computational load of processing P parallel output signals (7) is in the order of $(T + 1) \cdot M \cdot L + P \cdot T$ multiply-and-accumulate operations per sampling interval, whereas the conventional filter-and-sum beamformer (2) requires an order of $P \cdot M \cdot L$ operations. Therefore, the proposed structure is computationally more efficient, when $P > (T + 1)/\left(1 - \frac{T}{M \cdot L}\right)$.

3. SIMULATION

To illustrate the theory, we simulate a 4-element linear microphone array of omni-directional transducers for the beamforming filter structure shown in Fig. 3. We select 5 cm sensor spacing in order to avoid spatial aliasing on the highest frequency of interest, i.e. 3.4 kHz for a narrowband speech signal.

3.1. Design criteria

We have developed a method to optimize the directional sensitivity of a polynomial filter-and-sum beamformer. The directivity of the broadband microphone array is optimized by adjusting the polynomial filter coefficients to minimize the mean square error (MSE) between the desired and the actual response of the beamformer.

Let $\Omega_{S_d} \subseteq \mathbb{R}^3$ denote the set of desired signal source positions and let $\Omega_{S_n} \subseteq \mathbb{R}^3$ be the noise source locations. Given the desired beamformer output magnitude response $|Y_d(z, D)|$ for the source points in $\Omega_S = \Omega_{S_d} \bigcup \Omega_{S_n}$, frequencies Ω_f, and look direction Ω_ϕ we define the MSE as

$$MSE = \sum_{S \in \Omega_S} \sum_{f \in \Omega_f} \sum_{D \in \Omega_\phi} \frac{(|Y(z, D)| - |Y_d(z, D)|)^2}{|\Omega_S| \cdot |\Omega_f| \cdot |\Omega_\phi|}, \tag{8}$$

where $Y(z, D) = \sum_{j=1}^{M} H_j(z, D) X_j(z)$ is the output of the beamformer in the frequency domain.

One of the design targets is to optimize the fixed filter parameters $a_t(j, k)$ to provide a flat magnitude response in a look direction. Another criterion is to have a flat spectrum in any other direction as well. The third aim is to enable steering of the desired look direction ϕ_{des} in any direction $0° \leq \phi_{des} \leq 180°$. Hence, we define $D_{des} = (\phi_{des} - 90°)/90°$ to keep the values of D_{des} in a practical range from -1.0 to $+1.0$. To meet all these design targets, we optimize the fixed filter parameters $a_t(j, k)$ for $M = 4$, $L = 20$ and $T = 3$ (see Fig. 3 and (5)).

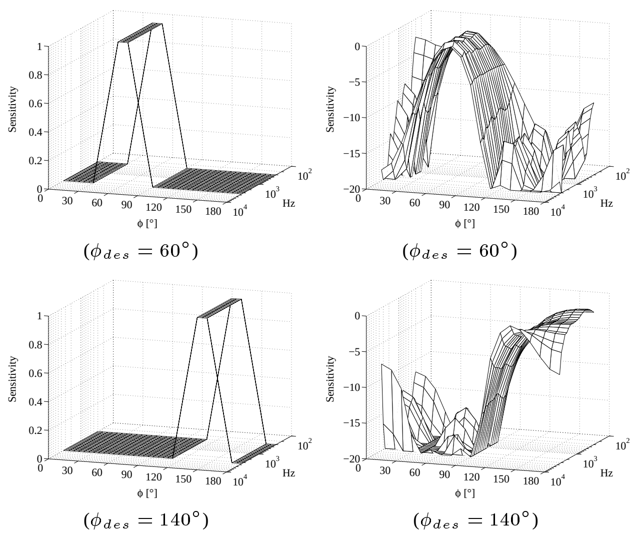

Fig. 4. *Design target vs. actual filter response for $\phi_{des} = 60°$ (top) and $\phi_{des} = 140°$ (bottom).*

3.2. Design example

In the following example, we show the results of steering an optimal polynomial beamforming filter. The following three figures illustrate the performance of the polynomial filter for different look directions and for signal sources located at the distance $r = 0.4$ m.

Fig. 4 illustrates the target response $|Y_d(z)|$ used for filter optimization and the corresponding directional sensitivity of the polynomial beamforming filter for the look directions $60°$ and $140°$. In Fig. 5 the performance of the polynomial beamformer is compared to traditional constant filter-and-sum beamformers for three look directions $45°$, $90°$, and $115°$. The polynomial beamformer is jointly optimized to five look directions $[35°, 55°, 90°, 125°, 145°]$ and the directions in between these angles are interpolated by the polynomial filtering. On the left column the corresponding constant beamformer represents the minimum MSE solution for the given look direction ϕ_{des}. The small visual differences between the parallel figures in Fig. 5 can be confirmed from Fig. 6 where MSE of the two beamformer types is shown.

4. SUMMARY

In this paper, we have introduced a new polynomial filter implementation of a conventional filter-and-sum beamformer. The proposed filter structure provides an efficient beamforming filter implementation, which supports dynamic adjustment of beamforming filter characteristics. Additionally, polynomial beamforming filters offer the same robustness as conventional filter-and-sum beamforming techniques.

We have demonstrated the new method by simulating the operation of a linear array of four omni-directional microphones. In future, we will apply the proposed method to 2-D and 3-D array ge-

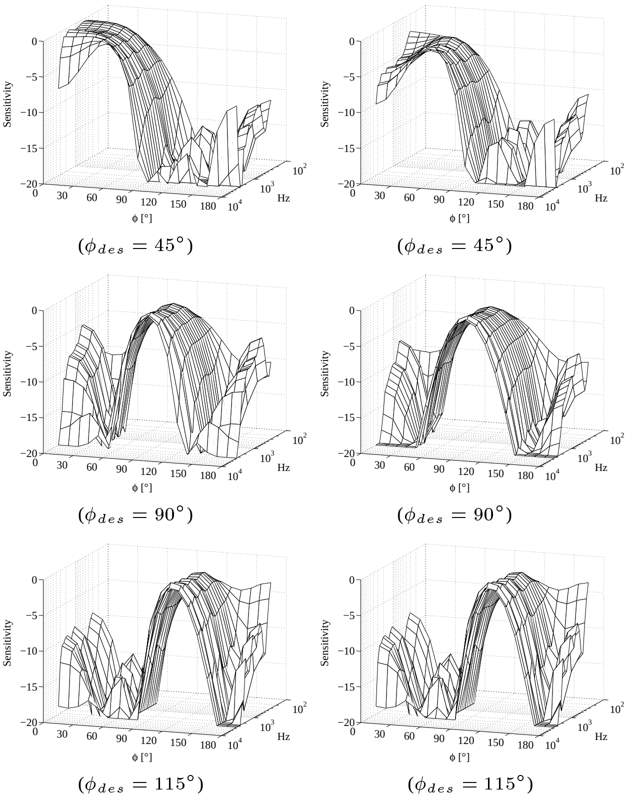

Fig. 5. *Traditional constant filter-and-sum beamformers (left column) and a polynomial beamformer (right column) in three look directions* $45°$, $90°$, *and* $115°$.

ometries as well as to directional microphones. We will also study the integration of source tracking algorithms with beam steering to look for potential applications for future communication systems.

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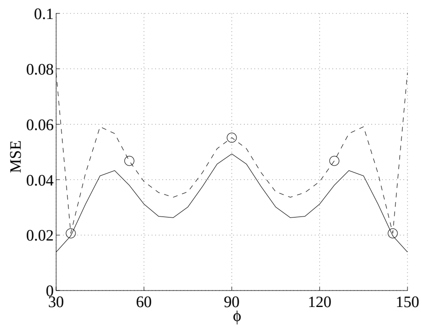

Fig. 6. MSE of the traditional filter-and-sum beamformers optimized for one look direction at the time (solid line) and the proposed polynomial beamformer (dashed line), respectively. Optimization angles for polynomial beamformer are marked with "o".

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