SPHERICAL MICROPHONE ARRAY ACOUSTIC RAKE RECEIVERS

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ABSTRACT

Several signal independent acoustic rake receivers are proposed for speech dereverberation using spherical microphone arrays. The proposed rake designs take advantage of multipaths, by separately capturing and combining early reflections with the direct path. We investigate several approaches in combining reflections with the direct path source signal, including the development of beam patterns that point nulls at all preceding reflections. The proposed designs are tested in experimental simulations and their dereverberation performances evaluated using objective measures. For the tested configuration, the proposed designs achieve higher levels of dereverberation compared to conventional signal independent beamforming systems; achieving up to 3.6 dB improvement in the direct-to-reverberant ratio over the plane-wave decomposition beamformer.

Index Terms— Acoustic rake receiver, spherical microphone array, spherical harmonic domain, dereverberation.

1. INTRODUCTION

Speech signals produced in offices, conference halls and other enclosed environments, undergo multipath propagation such that the observed signals consist of many delayed and summed copies of the desired source. This reverberation can reduce the perceived quality of the speech and in extreme cases, can even damage intelligibility. Moreover, reverberation substantially reduces the accuracy of automatic speech recognition [1]. Dereverberation, that aims to mitigate the detrimental effects of reverberation, has consequently received tremendous interest from the research community, with intended applications of this research ranging from hearing aids to speech recognition systems [2] [3].

Proposed dereverberation techniques to date have tended to focus on either spatial filtering approaches [4], or methods that estimate and suppress the reverberant signal component according to a model of the reverberation process. Of the latter methods, spectral subtraction [5, 6] is perhaps the simplest, employing a statistical model of late reverberation, but also tends to suffer from musical noise. At the other extreme multichannel equalisation methods [7, 8] require an estimate of the acoustic impulse responses (AIRs) between the source and each microphone. Since these are hard to obtain in practice, it is desirable to develop methods which are robust to channel estimation errors [9–11]. In a recent dereverberation challenge [12], the system of [13] performed particularly well. It was based on multichannel linear prediction (MCLP) [14, 15] which requires an iterative estimate of a subband autoregressive process.

In contrast to the above methods, spatial filtering (beamforming), requires only an estimate of the direction of arrival of the desired source. The spatial diversity provided by a microphone array is exploited to selectively amplify sounds originating from a particular 'look' direction. Since diffuse reverberation arrives with equal probability from all directions, attenuating sound incident from all directions except in the look direction, as defined by the main lobe of the spatial response of the beamformer, can achieve a significant level of dereverberation [16].

By considering only the direct propagation path between the source and array, conventional beamformers implictly ignore the contributions of strong early reflections. The motivation behind rake receivers [17] is that reflections can be used constructively to reinforce the direct path signal. The term acoustic rake receiver was first used in [18] but the concept can be traced back to [19, 20]. Using knowledge of the image source locations up to 3rd order, independent beams (fingers) were directed at each of the reflections and summed, having first time aligned the image source signals with the direct path signal. Further, it was shown that this operation is conceptually similar to applying a matched filter to each channel of the array, given prior knowledge of the AIRs. In [21] a more practical, though simulated, implementation was presented where the dominant directions of arrival were identified using a steered response power map. Beams were steered in these directions and interbeam delays determined using a modified adaptive eigenvalue decomposition method. In [22] the concept of an acoustic rake receiver is presented in some detail with alternative formulations being shown to optimise different performance metrics. It should be noted that their delay-sum formulation, 'Rake-DS', is different to that presented in this paper.

One of the potential problems with rake receivers is that the output of each finger contains reflections which arrive before the target reflection, albeit at a lower amplitude, contributing to a pre-echo. The contributions of this paper are (1) formulation of the acoustic rake receivers in the spherical harmonic domain, (2) a novel acoustic rake formulation which suppresses pre-echo and (3) simulation results which demonstrate the advantage of including only a specific subset of possible fingers in the rake.

The remainder of this paper is organised as follows. Section 2 provides an overview of the theoretical framework that underpins spherical microphone array processing and beamforming. Rake designs are proposed and described in Section 3. In Section 4, the experimental setup and simulations used to test these designs are presented, with results highlighted in Section 5. Finally, conclusions are drawn and future work discussed, in Section 6.

2. BACKGROUND THEORY

The acoustic rake receivers proposed in this work are implemented for a spherical microphone array. Whilst the designs can be generalised to any arbitrary array, spherical microphone arrays, where microphones are placed in a spherical configuration either in free space or on a rigid baffle, present several intrinsic advantages over linear and planar array setups. These include an elegant mathemati-

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cal framework from which to operate, the ability to analyse the sound field in three dimensions, and most importantly for our application, being able to produce three dimensional beam patterns independent of the look direction [23]. In this section the theory of spherical microphone arrays is briefly reviewed, with a focus on the decomposition of a sound field for processing in the spherical harmonic domain (SHD). The signal model and beamformers considered in this work are then formulated and presented in the spherical harmonic domain.

2.1. Spherical Microphone Arrays

A sound pressure field impinging a sphere, $p(k, \mathbf{r})$, can be decomposed into spherical harmonics using the Spherical Fourier Transform (SFT)

$$p_{nm}(k,r) = \int_0^{2\pi}\int_0^{\pi} p(k,\mathbf{r})\, Y^*_{nm}(\theta,\phi)\, sin(\theta)\,\mathrm{d}\theta\,\mathrm{d}\phi, \quad (1)$$

where k is the wavenumber, $\mathbf{r}$=(r, θ, ϕ) the position on the sphere in spherical coordinates and Y^*_{nm} the complex conjugate of a spherical harmonic, of order n and degree m. The produced coefficients p_{nm}, generally referred to as eigenbeams, are the respective spherical harmonic weights required to reproduce the sound field via the Inverse Spherical Fourier Transform (ISFT),

$$p(k,\mathbf{r}) = \sum_{n=0}^{\infty}\sum_{m=-n}^{n} p_{nm}(k,r)\, Y_{nm}(\theta,\phi). \quad (2)$$

In practice, with a finite number of microphones, Q, placed on a spherical array, the SFT can be approximated by

$$p_{nm}(k,r) \approx \sum_{q=1}^{Q} a_q(k)\, p(k,\mathbf{r}_q)\, Y^*_{nm}(\theta_q,\phi_q), \quad (3)$$

where $\mathbf{r_q}$=(r_q, θ_q, ϕ_q), represents the position of the q-th microphone in spherical coordinates, and the quadrature weights $a_q(k)$ are chosen to ensure the expression is a good approximation for a given spatial sampling configuration [24].

The number of microphones also limits the maximum sound field order, N, that can be decomposed by the spherical array without spatial aliasing. Practically, this means that a wideband signal captured by a spherical array must be sufficiently band limited, such that $k_{max}\, r < N$, where the wavenumber k_{max} corresponds to the upper operating frequency [24].

2.2. Spherical Harmonic Domain Signal Model

A sound signal of interest produced in a reverberant environment and captured by a spherical microphone array, can be thought of as a finite number of planewaves arriving at the array from different directions. A unit-amplitude planewave arriving from the direction $\Omega_l = (\theta_l, \phi_l)$, can be mathematically described in the spherical harmonic domain as,

$$v_{l,nm}(kr) = b_n(kr) Y^*_{nm}(\Omega_l), \quad (4)$$

where the mode strength, b_n, is a function of the array configuration.

It follows that the sound pressure field can be expressed in the spherical harmonic domain, as a summation of planewaves, each with their own respective amplitude and phase [25]. Namely,

$$x_{nm}(kr) = \sum_{l=0}^{L} \alpha_l(k)\; v_{l,nm}(kr)\; s(k), \quad (5)$$

where $s(k)$ is the desired source signal, and the term $\alpha_l(k)$ represents the amplitude and phase of the l-th planewave. Although to accurately model a reverberant signal L can be large (or even infinite), in our work we consider only first order reflections, under the assumption that they are the most dominant. For the purposes of raking or combining these paths, they are deemed sufficient.

2.3. Spherical Harmonic Domain Beamforming

The output of a spherical harmonic domain beamformer can be expressed as a weighted sum of eigenbeams,

$$y(k,r) = \sum_{n=0}^{N}\sum_{m=-n}^{n} w^*_{nm}(k)\; p_{nm}(k,r), \quad (6)$$

where w_{nm} represent the weight of the associated spherical harmonic. The beamformer output can be more conveniently expressed using vector notation,

$$y(k,r) = \mathbf{w^H p}, \quad (7)$$

where $(\cdot)^H$ denotes the Hermitian transpose,

$$\mathbf{w} = [w_{00}, w_{1(-1)}, w_{10}, w_{11}, ..., w_{NN}]^T \quad (8)$$

and

$$\mathbf{p} = [p_{00}, p_{1(-1)}, p_{10}, p_{11}, ..., p_{NN}]^T \quad (9)$$

are the $(N+1)^2 \times 1$ vectors of the spherical harmonic array weights and sound pressure fields respectively. The dependencies on wavenumber and array radius are omitted for brevity.

A beamformer can be designed in the spherical harmonic domain by selecting array weights to modify the directivity pattern in a particular way, with the weight selection either signal dependent or independent. In the latter, the beamformer weights depend only on the direction of arrival of the source of interest. In the former statistical properties of the desired signal, and or the noise field, are also taken into account to achieve statistically optimal noise reduction and minimal speech distortion. Similar to beamforming in the spatial domain, the weights can also be chosen to maximise or minimise additional design criteria such as robustness, sidelobe level, white noise gain and so on.

A well known approach in selecting beamformer weights is known as the plane wave decomposition (PWD) beamformer [26]. This is a signal independent beamformer in which a maximum directivity beam is steered at a specified look direction Ω_l. The beamformer, or eigenbeam, weights are chosen by compensating for the array mode strength:

$$w^*_{l,nm}(k) = \frac{Y_{nm}(\Omega_l)}{b_n(kr)}. \quad (10)$$

For the purposes of our proposed rake designs, which aim to capture individual multipaths, the realisation of a beamformer capable of pointing nulls is an important objective. As a result we consider the approach of [27], in which weights are chosen to satisfy a distortionless response as well as multiple null constraints in specified look directions. The constraints can be expressed as,

$$w^*_{l,nm}(k)\; v_{l,nm}(kr) = \delta_{\,i\,l}, \quad (11)$$

where $\delta_{(\cdot)}$ denotes the Kronecker delta and $v_{l,nm}$ is the array response to a planewave from direction Ω_l, as described in (4). By setting $i = l$, where $l = 0, ..., L$, a distortionless response constraint in the l-th direction is established, along with L number of null constraints in the remaining specified directions. A least squares solution can be formulated to select array weights which satisfy the above constraints.

3. SPHERICAL RAKE DESIGNS

In the context of signal acquisition in reverberant environments, an acoustic rake receiver that captures and combines individual multipaths constructively, introduces dual benefits. The first is reinforcement of the direct path component. The second is the attenuation of the reverberant decay tail, for which suppression under their summation is due to the incoherent nature of late reverberation. In this section we present several rake design formulations in the spherical harmonic domain. The designs assume the directions and time of arrival of the direct path and all first order reflections are known. In practice, this information can be obtained using numerous methods, for example through the use of a spherical microphone steered response power map [21]. This approach would allow any significant reflections, if present and whatever their order, to be identified and localised. The reflections could then be time aligned and raked, by computing the relative time delays between them, using well known time delay estimation techniques [28]. Alternatively, in some scenarios knowledge of room geometry can be exploited in combination with source localisation methods, to determine image positions and time delays using geometric rules [29]. The experimental simulations conducted in this work allow us to exactly determine source and image positions and the relative time lags between them.

3.1. Spherical Delay-and-Sum Rake

The first rake receiver design proposed is the spherical delay-and-sum rake, or Rake-DS. In this intuitive implementation a series of beamformers, or rake fingers, are constructed to point a single PWD beam in the direction of individual sources - the direct path (true source) and a chosen number of early reflections (image sources). Knowledge of the relative time lags between multipaths is then utilised to apply the appropriate phase correction, such that the output of the beamformers are time aligned and combined together constructively. The Rake-DS output is expressed as

$$\begin{aligned} z_{DS}(k,r) &= \sum_{l=0}^{L} y_l(k,r) \cdot e^{f(\tau_l)} \\ &= \sum_{l=0}^{L} \mathbf{w}_{l,\mathbf{pwd}}^{\mathbf{H}} \, \mathbf{p} \cdot e^{f(\tau_l)}, \end{aligned} \tag{12}$$

where $\mathbf{w}_{l,\mathbf{pwd}}$ denotes the PWD array weight vector that points a beam towards Ω_l, the direction of the l-th multipath. The weights are computed according to (10). To constructively combine the captured multipaths, individual rake fingers are multiplied by complex exponentials in order to time align them with the L-th path. The phase correction, $f(\cdot)$, is a function of the time of arrival of the l-th multipath, τ_l. It is described as

$$f(\tau_l) = j\,k\,c\,(\tau_L - \tau_l), \tag{13}$$

where $j = \sqrt{-1}$ and c denotes the speed of sound. In our experiments in Section 5, all first order reflections for a rectangular room are considered, L = 6.

3.2. Pre-echo Null Rake

The second rake design proposed is the pre-echo null rake, or Rake-PN, which differs from the Rake-DS in that the rake fingers designed to point beams at individual early reflections, also steer nulls in the direction of any preceding multipaths, up to and including the direct path. The incorporation of nulls in this design is motivated purely by perceptual concerns; to avoid audible pre-echoes that would degrade signal quality. The Rake-PN can be expressed as

$$z_{PN}(k,r) = \sum_{l=0}^{L} \mathbf{w}_{l,\mathbf{null}}^{\mathbf{H}} \, \mathbf{p} \cdot e^{f(\tau_l)}, \tag{14}$$

where $\mathbf{w}_{l,\mathbf{null}}$ denotes the null steering beamformer array weights, which point a beam towards the l-th multipath, and nulls at $0, ..., (l-1)$-th paths. The weights are the least square solution to (11).

3.3. Path Selective Raking

Variations of the Rake-DS and Rake-PN designs, which are path selective, are also considered. In these designs, instead of combining in the rake structure all six first order reflections (which would be produced in a rectangular room), we investigate the performances of the previous two rakes as a function of the number and combination of multipaths that are raked with the direct path.

4. EXPERIMENTAL SETUP AND EVALUATION

The dereverberation performances of the acoustic rake receivers proposed in this work were evaluated in a simulated room of dimensions 5 x 4 x 3 m^3, the size of a typical office, with a reverberation time of 500 ms. The corresponding room impulse responses between a source and a rigid 32 element spherical microphone array, of radius 4.2 cm, were computed using the spherical microphone array impulse response generator [30]. The array was placed in the centre of the room, at a height of 1 m, whilst the source was positioned at an azimuth of 305°and an elevation of 15°, at a distance of 1.5 m from the array. The choice of source and microphone array positions was selected in order to achieve a low direct-to-reverberant ratio (DRR), as well as distinctly localisable early reflections so as to effectively illustrate the operation of the methods.

The spherical array beamformers implemented are of order N = 3, and the frequency of operation 100-4000 Hz. The Rake-DS and Rake-PN designs use all six first order reflections (L = 6), whilst the path selective rake designs consider all possible combinations of first order reflections. Baseline comparisons are made with the response of a virtual omnidirectional microphone placed at the centre of the spherical array and regular PWD beamforming. As an optimal signal independent beamformer, in the maximum directivity sense, the PWD is an appropriate baseline to compare against the signal independent rake designs proposed.

In order to assess dereverberation performances, we consider the well known DRR metric. Additionally, given the coherent summing of beamformer outputs through time alignment, we also consider the relative pre-echo energy an important evaluation criterion. In our work we define the direct to pre-echo ratio (DPER) as the energy ratio of the rake output reinforced direct path, to the total energy of the taps preceding it. Namely,

$$DPER = 10\log_{10}\left(\frac{z^2(n_L)}{\sum_{n=0}^{n_L} z^2(n)}\right), \tag{15}$$

where $z(n)$ is the rake output and n_L the sample index corresponding to the time of arrival of the L-th multipath being raked. This is in contrast to the DRR, in which the direct path energy is compared to the proceeding taps. The choice of the DPER metric is motivated by the fact that pre-echoes have a perceptually damaging effect, and therefore reverberation reduction, in the form of improvements in DRR, can not come at the cost of perceptible pre-echoes.

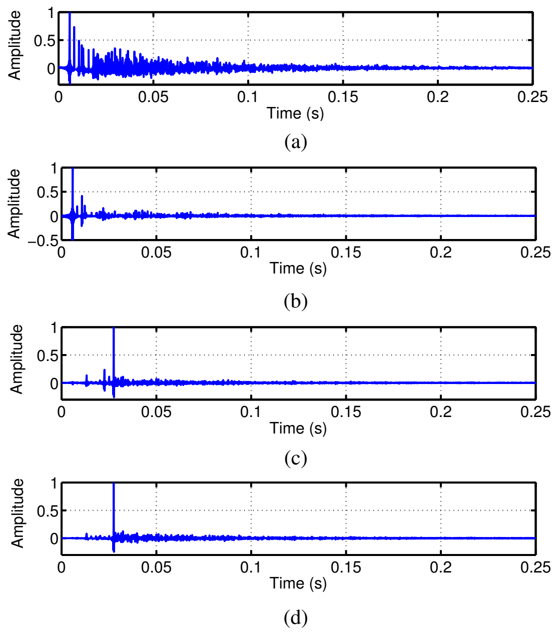

Fig. 1: Array response to a Dirac delta input in a 5 x 4 x 3 m^3 room, T_{60} = 500ms, (a) virtual omnidirectional microphone response, (b) PWD beamformer output, (c) spherical delay-and-sum rake and (d) pre-echo null rake.

Table 1: DRR and DPER performance of rake receiver designs compared to virtual and PWD baselines.

	DRR(dB)	DPER(dB)
Omni-mic	-7.06	-
PWD	1.88	-
Rake-DS	5.52	9.21
Rake-PN	3.98	16.56

5. RESULTS AND DISCUSSION

The results obtained for the various rake receiver designs are presented and compared in this section. The output of the rake designs to a room impulse response input are illustrated in figure 1. The plots help illustrate that in low DRR cases, rake designs that incorporate first order reflections can lead to substantial reverberation reduction over regular PWD beamforming.

Of the two rake designs proposed, it can clearly be seen that although the Rake-DS design results in higher direct path energy over the reverberant decay tail, the level of pre-echo energy is high. By incorporating null steering, the Rake-PN on the other hand results in DRR improvements whilst not suffering from the same level of pre-echo. These results are numerically summarised in Table 1.

Further investigation has shown that the performance of the proposed rake receivers is highly dependent on the number and combination of multipaths being raked. Figure 2 shows the distribution of DRR scores for all possible rake combinations for both rake designs. The plots make clear that whilst on average an increasing number of rake fingers results in DRR improvement over regular PWD beamforming, there is a large variation. Dependent on the multipaths selected, the rake designs can lead to lower DRR scores than using a PWD beamformer. This is especially true for the Rake-PN. This can be attributed to the difficulty in synthesising beam patterns that require a null constraint close to the direction of a beam.

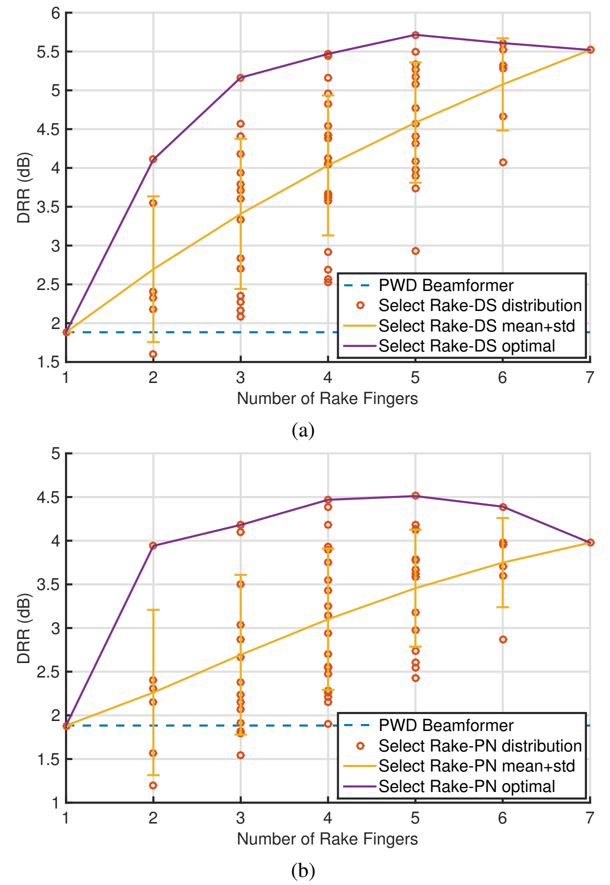

Fig. 2: DRR performance of path selective (a) spherical delay-and-sum raking and (b) pre-echo null raking.

6. CONCLUSION AND FUTURE WORK

In this paper we have presented acoustic rake receiver designs that utilise multipaths to improve signal acquisition in reverberant environments. Experimental simulations show that the designs lead to improved levels of dereverberation over regular PWD beamforming. Furthermore, by incorporating null steering into the Rake-PN design, we were able to suppress the level of pre-echoes present at the output by 7.35 dB compared to the Rake-DS design, whilst still maintaining DRR improvements over PWD beamforming. Our investigation has also demonstrated that the performance of rake designs is dependent on the number and combinations of multipaths raked, depending on how similar their direction of arrival is. Future work would include development of designs that would select the paths to rake, depending on their directions of arrival. Robustness to errors in source and reflection angles of arrival will also be investigated.

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