State-of-the-art Analysis of Beamforming using Microphone Arrays with Near-Field Broadbanding for Acoustic Applications

Erik Hagenaars (4272404) and Wouther Bons (4092716)

***Abstract*—This literature review concerns the application of near-field beamforming techniques using microphone arrays to tackle acoustic problems. An overview and state-of-the-art analysis of beamforming are presented.**

Recently, researchers have developed robust beamformers using region-based algorithms, using constrained optimisation and by performing a maximum likelihood estimation. Because robustness is a very practical concern in beamformer design, the measured directivity of practical microphones is of interest. Microphone responses can not be neglected when designing beamformers.

As an alternative to linear microphone arrays spherical microphone setups can be used for near-field beamforming.

State-of-the-art design techniques include a total-least squares approach (TLS) that uses eigenfilters, subspace-based methods for use with spherical microphone arrays and time-space complex envelope techniques for use in acoustics.

Lastly, it is noted that real-world performance of beamforming algorithms can drastically vary from their theoretical performance and even from tests in an an-echoic chamber.

In conclusion state-of-the-art research focuses on spherical microphone array setups and a variety of algorithms to design robust beamformers for near-field acoustic applications. New algorithms and techniques should be designed for robustness and this should be verified using real-world experiments.

I. Introduction

At conferences and big meetings, when someone in a big room asks a question, it is often hard to hear this person speak without equipping them with a microphone first. To tackle this annoying and time consuming problem, researchers are trying to implement a configuration for microphone arrays with different algorithms to only listen to a specific direction, a technique referred to as beamforming. This paper will look into fundamental and state-of-the-art research in beamforming with a focus on the Near-Field.

In order to understand the difficult techniques, the basic knowledge of beamforming and its types is discussed first (sec. II), consisting of the basis for beamforming, the simplified problem in the Far-Field, the complex problem in the Near-Field and at last a different configuration of spherical microphone arrays. This is followed by the state-of-the-art in the field of research of algorithms, configurations and approaches to specific problems (sec. III).

Finally, from the most recent papers it is concluded that almost no papers evaluate their claims using practical tests, instead relying fully on simulated models using random generated noise and interference (sec. IV). However, in order to legitimately evaluate the designed algorithms and approaches to design these systems, real-time tests need to be conducted.

II. Supporting Methodology

Before the state-of-the-art analysis is done, the basic knowledge and supporting methodology concerning beamforming is discussed. Firstly, the Far-Field configuration is summarised since this configuration lowers the complexity of the beamforming problem [1]. This is followed by the Near-Field configuration. Lastly, a spherical configuration of the microphone array is considered as an alternative to the more common linear array configuration.

A. Beamforming methodology

Beamforming is a spatial processing technique where an array of sensors only 'listen' to one particular direction. In this case, the sensors are microphones and the desired signal is an acoustic sound wave coming from that direction. The input signal of a microphone k can be described using equation 1 [2, p. 1],

$$x_k = s_k h_{s_k} + \sum_{i=1}^{I} v_{i_k} h_{v,i_k} + n_k \tag{1}$$

Where the summation is a convolution of the desired signal v_k convoluted with the steering channel $h_{v,k}$, s_k are interference signals with other steering channels h_{s_k} and n_k is measurement noise.

With this equation, the standard way of filtering the noise and interference is to create a weight vector w_k which will be convolved with vector x_k in order to retrieve the desired signals and to filter the noise and interference. To find this vector w_k, a least squares optimisation algorithm can be used in combination with the correlation matrices of the desired signal. This technique can not be applied if the correlation matrices are unknown, which is often the case. An important piece of information in the received signal can be found however: the time delay at which the signal arrives at each microphone. If the source position is known, the desired signal can be extracted. Problems that could still occur include reflections and line of sight interference.

In this paper, the two types of field considered are Far-Field and Near-Field, defined as respectively the space far away and close to the source. Each field has its own advantages and disadvantages.

B. Far-Field beamforming

The Far-Field beamforming configuration lowers the complexity of the beamforming problem [1]. The curvature of the sound waves' wavefront is assumed neglegible and consequently the wavefront is approximated as a straight line. For this configuration, a linear microphone array is used. When assuming this linearisation, the delay of the receiving signal at every microphone can be a simple equation as shown in equation 2 where τ_n is the delay at microphone n as a function of the angle θ in which the signal arrives at the microphone array. d_n is the distance to the centre microphone and c and f_s are the speed of light and sample frequency respectively.

$$\tau_n(\theta) = \frac{d_n cos\theta}{c} f_s \tag{2}$$

With this equation, the steering matrix and thus the spatial directivity spectrum can be constructed. These two matrices are important for the design of the filter which needs to be optimised to fit the spatial directivity pattern. To evaluate this filter, three cost functions can be used: (i) a least-squares cost function (ii) a maximum energy array cost function (iii) a non-linear cost function. Each of these cost functions have their own linear constraints and problems.

C. Near-Field beamforming

Near field beamforming is much more complex since the curvature of the signal is relevant. This curvature becomes valid when the source is closer to the microphones. Far-Field characteristics are only valid under the following condition:

$$r < \frac{d_{tot}^2 f_s}{c} \tag{3}$$

Where the r equals the distance of the source to the central microphone, d_{tot} the total length of the microphone array, f_s and c the sample frequency and speed of light respectively. The cost functions used for the Far-Field are the same as for the Near-Field. The only difference is that the steering matrix and other quantities are defined differently. As expected this steering matrix becomes more complex where the distance of the source becomes important. The following function describes the delay of the signal τ_n for every microphone n. Where $r_n(\theta, r)$ is the distance from the source to the microphone n:

$$\tau_n(\theta, r) = \frac{r_n(\theta, r) - r}{c} f_s \tag{4}$$

$$r_n(\theta, r) = \sqrt{(rsin\theta)^2 + (d_n + rcos\theta)^2} \tag{5}$$

Since one more variable, r, is used, the computations become much more complex. Since this operation requires a lot of hard computations, the beamformers are often designed for a fixed amount of distances.

D. Spherical microphone array configuration

Instead of using a linear microphone array configuration, the configuration could be spherical. Some research has been done for this to find out if this configuration brings more benefits to the beamforming problems. An advantage of the spherical configuration is the three dimensional symmetry.

III. State of the art analysis

A. Robustness

Most existing algorithms do not account for reverberation explicitly. State-of-the-art research has been done in this area by the Circuits and Systems group at the TU Delft. In one of their papers [3] the researchers propose to account for this reverberation using a region-based algorithm. This results in a robust beamformer for near-field applications.

Another recent approach was to formulate the beamformer design problem as a constrained optimisation problem, also resulting in a robust near-field beamformer [4].

Furthermore, an adaptive method was proposed [5] specifically for the problem of estimating speech signals from multiple acoustic sources using a microphone array. The method, which has experimentally been confirmed to be robust, uses a rank-deficient correlation matrix to do a maximum likelihood estimation.

Complementary to the study of robustness in beamforming are the imperfections that result in the need for robust algorithms. Recent research has therefore looked at the directivity of microphones in smartphones and found that the results of beamforming techniques can be ”degraded considerably if the microphone responses are not considered in the beamformer design” [6, p. 97]

In the general studies of noise reduction for microphone arrays, an algorithm has been developed that minimises distortion [7]. The suggested algorithms provides many appealing properties for practical uses in comparison with existing techniques, but has not yet been tested on near-field beamformers.

A different paper did recently experimentally test interference suppression in near-field beamformers for acoustic applications [8], but this was using a spherical microphone array.

B. Spherical microphone arrays

Other research looked into the robustness of spherical microphone arrays when used for beamforming [9]. The research concludes that these kinds of arrays can indeed be used effectively in a design, as they show how an adaptive design algorithm can be used to converge to the optimal solution quickly without losing robustness.

In recent years, advances are being made in the field of using spherical microphone arrays for near-field beamforming. Traditionally, spherical arrays have mostly found their use in far-field applications because of how easy they are to process, but these methods have recently been translated to a model for near-field setups [8].

C. Design techniques

Lots of techniques can be used to design beamformers. One approach (for the far- and near-field) is to use eigenfilters [10]. This technique, that’s based on a total-least squares (TLS) approach, circumvents the problem of needing a frequency-angle point reference to design the beamformer.

In recent years several algorithms have been developed and experimentally tested for spherical microphone array setups [8]. These are subspace-based methods.

Specifically for the acoustic field, a recent study [11] has shown time-space complex envelope techniques to be effective. It shows numerical simulation results that show sensitivities of three Green’s functions: the time-domain Green’s function is sensitive to the distance and number of measurements, the space-domain Green’s function to the distance only and the wavenumber domain Green’s function is sensitive to neither of those two.

Performance of these techniques is more often than not evaluated only by simulations and comparison to simulations of competing techniques, rather than using real-world testing. This is a problem because the ”simulations or even tests in controlled environments like an anechoic chamber can lead to non-realistic conclusions" [3, p. 2497]. Real-world tests are therefore required to evaluate the performance of the algorithms and techniques. Papers from recent years have often confirmed their claims about the performance of their techniques, but plenty of older results have not yet been evaluated using simulations.

IV. CONCLUSION

This paper presented a state-of-the-art analysis of near-field beamforming using microphone arrays as a way to tackle problems in acoustics. One such problem is that of picking up a person's speech at meetings amidst interference. For those situations, the beamformer should be designed for robustness, which can be done using region-based algorithms, by reformulating the design problem as a constrained optimisation problem and by using an adaptive method that effectively performs a maximum likelihood estimation. Since robustness and the connected topic of interference are very practical aspects that need to be considered when doing beamforming, researchers have tested the directivity of practical microphones (i.e. of smartphones) and found that microphones' responses can not be neglected when designing beamformers.

After that an alternative to the linear microphone array was discussed. Research using this spherical setup to do near-field beamforming is promising, even though they have traditionally mostly been used for far-field beamforming. Several papers present their viability for near-field setups.

Another topic of interest was state-of-the-art design techniques. These are a total-least squares (TLS) approach that uses eigenfilters, subspace-based methods for use with spherical microphone arrays and time-space complex envelope techniques for use in acoustics.

Lastly, it was noted that recent research indicates real-world performance of beamforming algorithms can drastically vary from their theoretical performance and even from tests in an anechoic chamber. Claims from papers on the performance of beamforming techniques that have not been verified in *real-world* experiments should therefore be confirmed before blindly utilising those results as a basis for future research.

In conclusion, state-of-the-art research focuses on spherical microphone array setups and a variety of algorithms to design robust beamformers for near-field acoustic applications. New algorithms and techniques should be designed for robustness and this should be verified using real-world experiments.

REFERENCES

[1] J. G. Ryan and R. A. Goubran, "Near-field beamforming for microphone arrays," in *1997 IEEE International Conference on Acoustics, Speech, and Signal Processing*, vol. 1, Apr 1997, pp. 363–366 vol.1.

[2] C. A. Anderson, P. D. Teal, and M. A. Poletti, "Spatially robust far-field beamforming using the von mises(-fisher) distribution," *IEEE/ACM Transactions on Audio, Speech, and Language Processing*, vol. 23, no. 12, pp. 2189–2197, Dec 2015.

[3] J. Martinez, N. Gaubitch, and W. B. Kleijn, "A robust region-based near-field beamformer," in *2015 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, April 2015, pp. 2494–2498.

[4] F. Borra, L. Bianchi, F. Antonacci, S. Tubaro, and A. Sarti, "A robust data-independent near-field beamformer for linear microphone arrays," in *2016 IEEE International Workshop on Acoustic Signal Enhancement (IWAENC)*, Sept 2016, pp. 1–5.

[5] A. S. Ivanenkov and A. A. Rodionov, "Robust adaptive method for speech signal waveform estimation using microphone array," in *2015 23rd European Signal Processing Conference (EUSIPCO)*, Aug 2015, pp. 1162–1166.

[6] N. D. Gaubitch, J. Martinez, W. B. Kleijn, and R. Heusdens, "On near-field beamforming with smartphone-based ad-hoc microphone arrays," in *2014 14th International Workshop on Acoustic Signal Enhancement (IWAENC)*, Sept 2014, pp. 94–98.

[7] J. Chen, J. Benesty, and Y. Huang, "A minimum distortion noise reduction algorithm with multiple microphones," *IEEE Transactions on Audio, Speech, and Language Processing*, vol. 16, no. 3, pp. 481–493, March 2008.

[8] L. Kumar and R. M. Hegde, "Near-field acoustic source localization and beamforming in spherical harmonics domain," *IEEE Transactions on Signal Processing*, vol. 64, no. 13, pp. 3351–3361, July 2016.

[9] Z. Li and R. Duraiswami, "Flexible and optimal design of spherical microphone arrays for beamforming," *IEEE Transactions on Audio, Speech, and Language Processing*, vol. 15, no. 2, pp. 702–714, Feb 2007.

[10] S. Doclo and M. Moonen, "Design of far-field and near-field broadband beamformers using eigenfilters," *Signal Processing*, vol. 83, no. 12, pp. 2641–2673, 2003, cited By 60.

[11] X. Guo, X. Wang, C. Lan, and C. Han, "Application of time - space complex envelope technique in time-domain near-field acoustic holography," in *2017 2nd Asia-Pacific Conference on Intelligent Robot Systems (ACIRS)*, June 2017, pp. 125–128.