# A NOTE ON THE FILTERING EQUALIZATION IN LARGE MULTIACTUATOR PANELS

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## ABSTRACT

Multiactuator Panels (MAPs) are flat loudspeakers that can be used to build arrays for Wave Field Synthesis (WFS), alternatively to classical dynamic loudspeakers. The sound radiation of these loudspeakers is a complex superposition of excited modes that vary strongly with frequency. Therefore, the characteristic MAP spectrum is very irregular and needs to be equalized to obtain a flat, colorless response. In this paper, an efficient digital equalization method is applied successfully to the problem of spectral equalization of MAPs. It is based on a chain of second-order sections of infinite impulse response parametric filters with very low computational cost. The method compensates for the measured MAP response in order to achieve a desired frequency response. The equalization process has been applied to a large MAP prototype, comprising a large number of exciters within a panel. A dedicated exciter equalization and its consequences on the filtered responses have been addressed.

## 1. INTRODUCTION

WFS is a spatial sound rendering technique that generates a true sound field using loudspeaker arrays [1]. Wave fields are synthesized based on virtual sound sources at some position in the sound stage behind the loudspeakers or even inside the listening area.

Most WFS prototypes employ arrays of dynamic loudspeaker drivers. However, a main disadvantage of these drivers is that they need a housing with a relatively large volume to avoid the additional stiffness of the back-volume. Also, due to the piston-like motion of their diaphragms, they exhibit strong and narrow lobes around their main axis at high frequencies.

Therefore, efforts have been made to find alternatives that are as omnidirectional as possible, and easy to mount directly onto the wall surface. Multi Actuator Panels (MAPs) represent a new technology in which the emission of sound is based on the stimulation of bending waves on a flat, hard and light panel material through special dynamical transducers, so called exciters [2],[3]. Such a panel can be easily mounted to the wall and due to the variable size and shape, MAPs enable integration into a living-room because they serve as a projection screen.

The MAP radiation produces an excitation of modal frequencies, which are sufficiently dense and evenly spaced that an illusion of continuous spectrum is created for the listener. However, the vibration pattern of the panel causes a complex superposition of the bending wave excitations which vary strongly with frequency. This uneven response, with large dips and peaks, degrades the perceived sound quality and makes the use of MAPs less attractive compared with the use of conventional speakers. Therefore, the irregular MAP response needs to be equalized for a natural, uncolored response.

In this paper, an efficient algorithm for spectral equalization is addressed aiming at compensating the unnatural colored response of a large MAP prototype intended for WFS. Large panels have added benefits for WFS-based immersive environments since sound and image stimuli are generated by a single element, being the panel the video screen and loudspeaker array simultaneously. The equalization method is oriented towards using a very low computational cost by means of an advantageous IIR scheme, which was proposed by two of the authors in [4] and later tested in medium-sized MAPs in [5]. The aim is to explore the potential advantages of the symmetry of the exciter positioning in such a large panel to reduced the number of measurements and filter computations required. The presented method can be employed for compensating the absolute value of the spectrum with application to spectrum equalization of the panel.

## 2. MULTIACTUATOR PANELS

In this section, an overview of Distributed Mode Loudspeakers (DML) as single-channel systems is presented, together with their evolution on increasing the number of channels to convert them into MAPs. DML consist of a flat panel of a light and stiff material to which one or more dynamic exciters are attached. The exciter mechanical vibration forces the panel material to vibrate, which in turn radiates a sound field [6],[7]. In contrast to the piston-like surface movement of cone loudspeakers, the radiation mechanism of DML panels is produced by means of bending waves that travel across the surface of the panel [8],[9].

The typical impulse response of an exciter in a DML panel consists of a first direct pulse, just as with dynamic loudspeakers, followed by a long tail related to the distributed modes of the panel. In contrast to the random behavior of the panel, this initial part of the impulse response is deterministic and creates an adequate spatial impression [2], [3].

MAPs are an extension of the distributed mode technology to be used in WFS applications. At first, an array of small DML panels that substitute dynamic transducers was tested to be conceptually valid [2]. However, this solution was not optimal because of the poor response in low frequency and the edge effects that a set of adjacent panels would cause. Then, a larger panel was needed but without increasing the

 

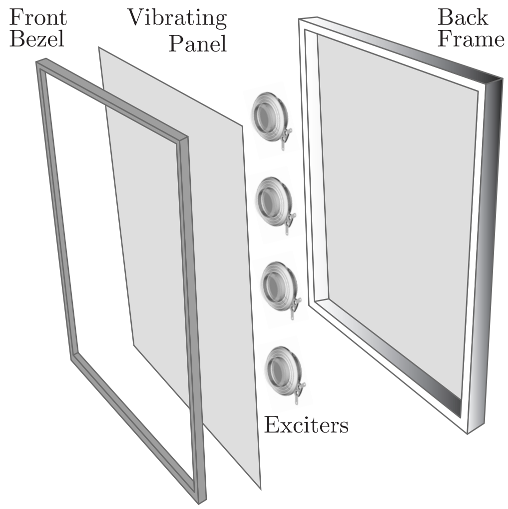


Figure 1: Illustration of a 4-exciter MAP. Electric wiring is omitted for simplicity.

distance between exciters if a certain aliasing frequency has to be maintained. Then, a single panel driven by multiple exciters, in which each exciter reproduces a given "driving signal", is a configuration called MAP [10],[11]. Figure 1 shows an illustration of a MAP with four exciters.

One of the practical advantages of MAPs is their ease to mount directly on the wall surface. Besides, they are lightweight loudspeakers with a small back housing that is unnoticed as part of the decoration. Since the panel surface can be large and the vibration is low enough to be imperceptible to the human eye, they can be integrated into a room interior and simultaneously used as projection screens [12],[11]. In this way, image and sound are fully integrated for multimedia applications. Furthermore, the cost of MAPs is generally lower than that of dynamic loudspeakers on baffles. These features makes MAPs very suitable for WFS reproduction.

Furthermore, the diffuse radiation of MAPs produces room reflections that are less correlated to the direct sound than those radiated from piston sources and thus, constructive and destructive interference of sound is minimized [13]. This property is beneficial for WFS since the individual components can merge correctly into the desired wavefront for a wider area. Also, the sensation of spaciousness is better than that of dynamic loudspeakers, which enhances the recreation of sources far behind the panels [14]. Moreover, for extending the synthesized wave field, the pressure level decay for MAPs as planar transducers is less pronounced with distance in comparison with conventional loudspeakers.

However, the downside of these loudspeakers is in the inherent irregular frequency response: the radiation of a MAP is generated by means of an excitation of modal frequencies, aiming at creating a smooth modal density and the perception of continuous spectrum. Nevertheless, this results in a complex superposition of bending wave excitations, which vary strongly as a function of frequency, with dips and peaks that are noticed by listeners as unnatural and colored. The next section will deal with a methodology to equalize such frequency responses and obtain responses as appealing as with typical cone-based loudspeakers.

## 3. EQUALIZATION

In the context of WFS reproduction, different filtering strategies have been proposed. On the one hand, the algorithms compensating for unwanted reflections are called room compensation or reflection compensation algorithms [15, 16, 17]. On the other hand, spectral equalization aims at compensating the irregularities in the frequency response of a loudspeaker [18]. Studies reported experimentally that, in general, the measured frequency response after applying the equalization filter in MAPs was still not flat but it exhibited a more uniform shape [19].

In WFS applications, a large number of channels are used and the computational cost of the compensation filters increases linearly with channel number. Furthermore, in a MAP, the behaviour of the exciters is a function of their position on the panel, so a filtering process must be considered for each exciter. For only compensating the spectrum absolute value, i.e. the coloration, rather than a FIR scheme, an IIR scheme can be employed, which requires a lower computational cost to achieve a proper equalization.

In this paper, a successful technique of IIR design applied to audio systems that has been proposed recently in [4] and validated for MAPs in [5] is used. This filter design method, differing from other IIR design methods, is characterized by the fact that the equalization structure is planned from the beginning as a chain of second order sections (SOS). A SOS chain is usually the way in which IIR filters are implemented later on a digital signal processor (DSP). The point of this filtering structure is that instead of defining each SOS by its five coefficients, each section is set as a parametric peak filter, high-pass filter or low-pass filter defined by three parameters of acoustic significance: central frequency, gain and $Q$-factor. Therefore the parameters to be optimized fulfill a desired frequency response range from five coefficients per filter to only three.

## 4. EXPERIMENTAL SETUP

Individual impulse responses are measured on axis of every exciter on the panel. Based on the response of these measurements, the corresponding filters are calculated and applied to the exciters. The on-axis frequency responses have been obtained by placing a 1/2" electrostatic microphone 1 meter in front of every individual exciter, obtaining a total of 13 impulse responses. In all measurements, Maximum Length Sequences (MLS) were used as excitation signals in order to improve signal-to-noise ratio and minimize unwanted reflections [20].

The effect that the filtering has on the synthetic wavefront created by means of WFS is obtained by sampling the MAP radiation both in time and space. Then, two Fourier transformations are made to obtain spatial and temporal frequencies, and the results are plotted in a graphical representation for analysis purposes. The analysis technique can be found in [21].

The sampling of the wave field along a 2 m line in front of the panel is performed by a microphone array. For implementing such an array, a single 1/2" electrostatic microphone is driven at 1 cm steps by a linear sliding table. This spacing is sufficiently below the theoretical Nyquist sampling limit of 9 cm. Details of the geometric description of the experimental setup are given in Figure 2.

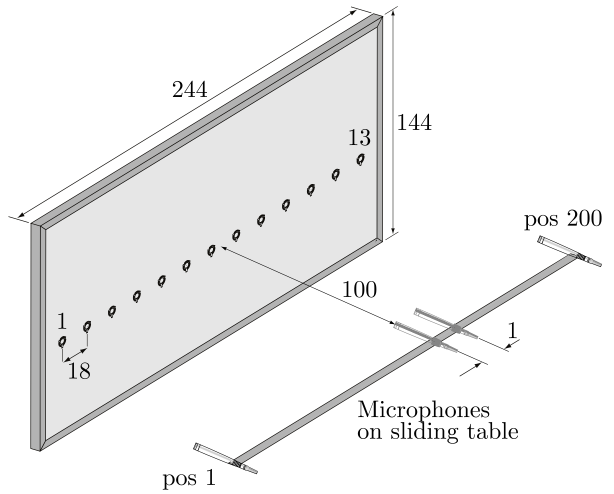


Figure 2: Experimental setup geometry for the wavenumber domain analysis. Dimensions in centimeters.

## 5. RESULTS

### 5.1 Filtering

The spectrum equalization process was applied to the large MAP in order to flatten the exciter responses up to a point where the physical constraints of the panel would counteract the filter process.

To that end, all individual exciters were measured on their respective axis. The results showed an interesting trend that was also observed in the smaller MAPs of [5], consisting of the similarity in the frequency response of symmetrically positioned exciters onto the panel. This feature can be exploited to reduce the number of measurements and filtering processes. In fact, since the behavior of the exciters positioned in symmetrical places at both ends of the panel is almost identical, it is feasible to measure for the first to the sixth exciter and assume that the thirteenth to the eighth exciter will behave similarly. Therefore, filters will be designed the for central exciter (seventh) and for exciters 1 to 6 and will be replicated accordingly to account for the rest of the exciters.

Figures 3(a) and (b) depict two selected measured responses corresponding to exciter 1, closest to the edge, and exciter 7, which is positioned in the center of the panel, respectively. Due to the enlarged size of the panel, the low cut-off frequency is lower than that of the smaller panels of the technical literature. Exciter 1 reaches the mid frequency band at 150 Hz and shows an extended output level from 200 to 800 Hz and from 10 to 20 kHz, the latter being in the region above coincidence frequency. In between, three moderate dips can also be observed. The response of exciter 7 is slightly extended in the low frequency region to 130 Hz and also presents the same two large peaks, whereas dips are more smoothed. As observed in the figures, the influence of the fact that the exciter is mounted close to the edge is negligible for this large MAP.

Considering these representative responses, all filters are designed with a second-order high-pass Butterworth target at 175 Hz, shown in gray line in Figure 3. Due to the amount of irregularities to compensate for, the filters are designed with 12 SOS sections. In all cases, from the total number of SOS,

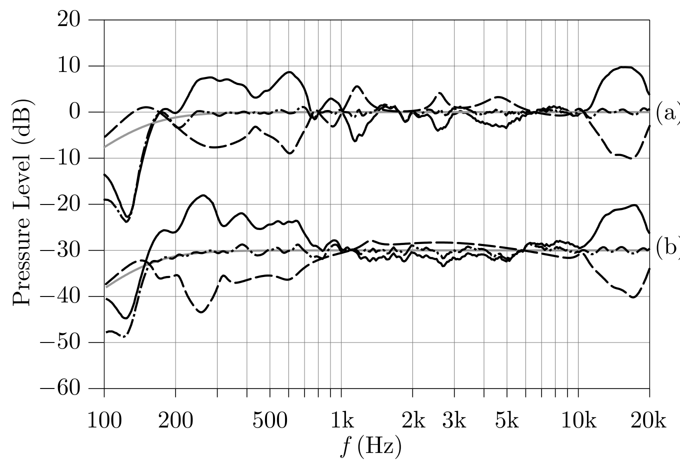


Figure 3: Frequency responses of the exciter equalization process for the large MAP. Exciter measurement (solid), filter target (gray), filter response (dashed), simulated exciter filtered response (dashed-dotted). (a) Exciter 1. (b) Exciter 7. Representations are shifted 30 dB in order to maintain clarity.

one is employed for the second-order high-pass filter. Regarding the control in the high frequency region, the nature of the MAP loudspeaker causes a roll-off at high frequencies, so no additional low-pass filtering is necessary. The responses of these filters and the simulated exciter filtered responses are given in Figure 3 in dashed and dashed-dotted respectively. The rest of the filters are computed in a similar way.

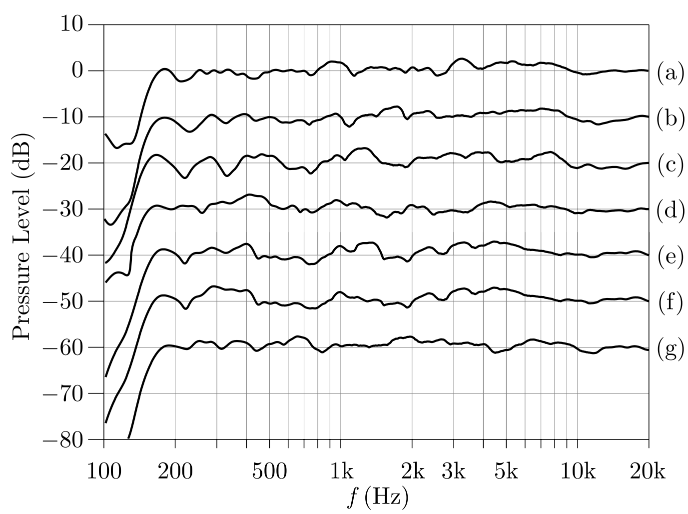


Figure 4: Exciter filtered responses of the large MAP. (a) Exciter 1 to (g) Exciter 7. Representations are shifted 10 dB in order to maintain clarity.

By taking advantage of the exciter position symmetry, filters were only calculated for exciters 1 to 6 and for exciter 7. Then, the resulting filters 1 to 6 were applied to the symmetric exciters 13 to 8, respectively. These filters were employed in the large MAP and the equalized frequency responses were measured in the laboratory. The resulting frequency responses are depicted in Figure 4.

From the above results, the filtering process has equalized the responses considerably, being now almost flat. As expected, some irregularities in the form of partially filtered peaks and dips appear in all the exciters. The physical constraints of the panel played a role by counteracting the filtering process to an extent. However, the filtering process has considerably improved the original uneven responses.

In order to give an objective quality measurement of the flatness of responses, the logarithmic standard deviation can be used. For a discrete Fourier spectrum $P[k]$ with length $L$ the logarithmic standard deviation is given by:

$$\sigma_{\log} = \sqrt{\frac{1}{L-1}\sum_{k=1}^{L}\{20\log|P[k]| - \overline{P_{\log}}\}^2} \qquad (1)$$

where $P_{\log}$ is the mean value of $20\log|P[k]|$ for $1 \leq k \leq L$. The logarithmic standard deviation allows one to compare multiple responses with different amplitudes since scale factors in the data are cancelled out. This measurement is also used by other authors for assessing the quality of MAPs [19]. For a perfectly flat frequency response, the logarithmic standard deviation will be equal to 0 dB, which is the lower limit. The higher the value, the more irregular is the frequency response.

For the filtered responses of Figure 4 in the range of the target frequency response (175 Hz to 20 kHz), the resulting logarithmic standard deviation values result in values between 1 and 2 dB.

### 5.2 WFS Performance

This section will assess the performance of the large MAP by analyzing the sound wavefield created by synthesizing an axial plane wave source in WFS. Details on the technique used can be found in [21].

Figure 5 depicts the experimental setup and the responses of the MAP on emitting an axial plane wave ($\theta = 0^{\circ}$). Multitrace impulse response shown in Figure 5(b) is achieved by representing the impulse responses of each microphone and gives information on the wave front propagating through space. In Figure 5(c), a space-time wavenumber domain representation is plotted.

As expected, the multitrace impulse response illustrates some artifacts that can be seen as two edge events, weaker and delayed, that follow the planar wavefront. In the wavenumber domain representation, there are components at zero $k_x$ which are repeated in the spatial axis with periods that depend on the spacing of the transducers in the array [22]. For such a MAP radiating an axial plane wave and considering 18 cm as the distance between exciters, the maximum spatial frequency without aliasing is $k_x$=34.9 m$^{-1}$. In an infinite length loudspeaker array, where radiation occurs at angles reaching $\pm 90^{\circ}$, the associated temporal wavenumber would be directly $k$=34.9 m$^{-1}$ (1.9 kHz). However, the finite size of the MAP causes truncation effects which modify the maximum incoming angle. For the geometry setup, a centered listening point has an angle of $\pm 50^{\circ}$ with respect to the array aperture (2.44 m). Then, the aliasing spatial frequency of 34.9 m$^{-1}$ matches to $k$=45 m$^{-1}$, so the aliasing frequency for axial plane waves in such a particular geometry setup is 2.4 kHz. If other angles are considered, when reaching the maximum plane wave that the MAP is capable

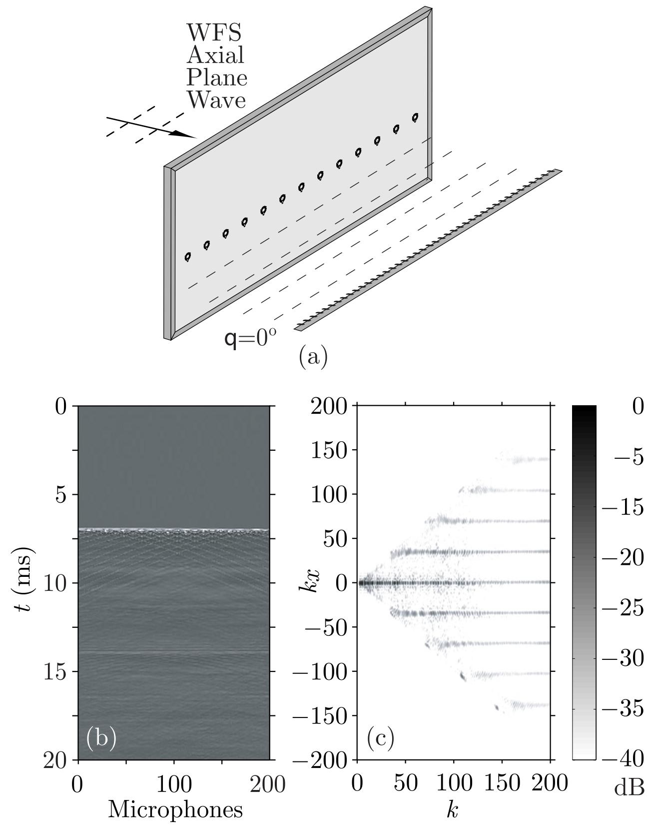


Figure 5: Axial plane wave generated by the MAP prototype. (a) Setup. (b) Multitrace impulse response. (c) Wavenumber domain representation.

of emitting, the aliasing frequency decreases to a minimum of 1.2 kHz. Note that the theoretical aliasing frequency of $c/2\Delta x$, where $\Delta x$ is the spacing between transducers, is calculated for loudspeaker arrays of infinite length.

### 5.3 Conclusion

A large MAP prototype, designed and built to fulfill the demands of WFS audio applications, has been equalized with an efficient equalization method that requires a very low computational requirements.

Due to the low visual profile and the ability of integrating video and audio stimuli, a large size screen is a proper solution for true audiovisual immersion for a large audience. However, MAPs exhibit an irregular frequency response due to the modal behaviour that must be compensated for every exciter. To that end, individual equalization of the drivers for the entire frequency band has been tested with satisfactory results. Although not completely flat, the improvement of the frequency responses with low computational cost is remarkable.

For a MAP with a large amount of excitation points, similarities in the response of symmetrical exciters on a MAP can be used to considerably reduce the number of measurement points. Additionally, if several MAPs are manufactured maintaining exciter type and position, panel material and housing, their responses will be similar. Hence, for a complete reproduction setup with several MAPs, only a small set of equalization filters must be designed, provided that exciters are symmetrically placed on the panels. Also, results in the form of multitrace impulse responses and wavenumber domain analysis showed that the sound field is properly generated for a wide listening area.

By taking advantage of this efficient equalization method and the similarities in the symmetrical positioned exciters on a panel, a very large WFS system driven by many MAPs can be equalized individually with low hardware requirements.

## REFERENCES


[1] A. Berkhout, D. de Vries, and P. Vogel, “Acoustic control by Wave Field Synthesis,” *J. Acoust. Soc. America*, vol. 93, no. 5, pp. 2764–2778, May 1993.

[2] M. Boone and W. de Brujin, “On the applicability of distributed mode loudspeaker panels for Wave Field Synthesis based sound reproduction,” in *108th Conv. Audio Eng. Soc.*, Conv. Paper 5165, Paris, France, Feb. 2000.

[3] M. Boone, “Multi-actuator panels (MAPs) as loudspeaker arrays for Wave Field Synthesis,” *J. Audio Eng. Soc.*, vol. 52, no. 7–8, pp. 712–723, Jul. 2004.

[4] G. Ramos and J. J. López, “Filter design method for loudspeaker equalization based on IIR parametric filters,” *J. Audio Eng. Soc.*, vol. 54, no. 12, pp. 1162–1178, Dec. 2006.

[5] B. Pueo, J. J. López, G. Ramos, and J. Escolano, “Efficient equalization of multi-exciter distributed mode loudspeakers,” *Appl Acoustic*, vol. 70, no. 5, pp. 737–746, may 2009.

[6] N. Harris and J. Hawksford, “Introduction to distributed mode loudspeakers (DML) with first order behavioural modelling,” *IEE Proc. Circuits Devices Syst.*, vol. 147, no. 3, pp. 153–157, Jun. 2000.

[7] J. Borwick, *Loudspeaker and Headphone Handbook*, 3rd ed. Focal Press, 2001.

[8] J. W. Panzer and N. Harris, “Distributed-mode loudspeaker simulation model,” in *104th Conv. Audio Eng. Soc.*, Conv. Paper 4739, Amsterdam, The Netherlands, May 1998.

[9] J. A. Angus, “Distributed mode loudspeaker radiation mechanisms,” in *108th Conv. Audio Eng. Soc.*, Conv. Paper 5164, Paris, France, Sep. 2000.

[10] U. Horbach, E. Corteel, and D. de Vries, “Spatial audio reproduction using distributed mode loudspeaker arrays,” in *AES 21st Int. Conf. on Architectural Acoustics & Sound Reinforcement*, no. 99, St. Petersburg, Russia, Jun. 2002.

[11] W. de Brujin and M. Boone, “Application of Wave Field Synthesis in life-size videoconferencing,” in *114th Conv. Audio Eng. Soc.*, Conv. Paper 5801, Amsterdam, The Netherlands, Mar. 2003.

[12] M. Boone, W. de Bruijn, and W. van Rooijen, “Recent developments on wfs for high quality spatial sound reproduction,” in *110th Conv. Audio Eng. Soc.*, Conv. Paper 5370, Amsterdam, The Netherlands, May 2001.

[13] N. Harris, S. Flanagan, and M. Hawksford, “Stereophonic localization in the presence of boundary reflections, comparing specular and diffuse acoustic radiators,” in *104th Conv. Audio Eng. Soc.*, Conv. Paper 4684, Amsterdam, The Netherlands, May 1998.

[14] E. Corteel, K.-V. NGuyen, O. Warusfel, T. Caulkins, and R. Pellegrini, “Objective and subjective comparison of electrodynamic and map loudspeakers for wave field synthesis,” in *AES 30th Int. Conf.*, no. 15, Saariselkä, Finland, Mar. 2007.

[15] S. Spors, H. Buchner, R. Rabenstein, and W. Herbordt, “Active listening room compensation for massive multichannel sound reproduction systems using wave-domain adaptive filtering,” *J. Acoust. Soc. Am.*, vol. 122, no. 1, pp. 354–369, Jul. 2007.

[16] P.-A. Gauthier and A. Berry, “Objective evaluation of room effects on wave field synthesis,” *Acta Acust. united Ac.*, vol. 93, no. 5, pp. 824–836, Sep/Oct 2007.

[17] J. J. López, A. González, and L. Fuster, “Room compensation in wave field synthesis by means of multichannel inversion,” in *IEEE Workshop on Appl. of Sig. Proc. to. Audio and Acoust. (WASPAA'05)*, New Paltz, NY, USA, Oct. 2005.

[18] E. Corteel, U. Horbach, and R. Pellegrini, “Multichannel inverse filtering of multiexciter distributed mode loudspeakers for wave field synthesis,” in *112th Conv. Audio Eng. Soc.*, Conv. Paper 5611, Munich, Germany, May 2002.

[19] J. van Dorp and D. de Vries, “Wave field synthesis using multi-actuator panel: Further steps to optimal performance,” in *Proceedings of the AES 28th Int. Conf. on Multichannel Audio*, no. 4-3, Piteå, Sweden, Jun. 2006.

[20] J. Vanderkooy, “Aspects of MLS measuring systems,” *J. Audio Eng. Soc.*, vol. 42, no. 4, pp. 219–231, Apr. 1994.

[21] B. Pueo, J. J. López, J. Escolano, and S. Bleda, “Analysis of multiactuator panels in the space-time wavenumber domain,” *J. Audio Eng. Soc.*, vol. 55, no. 12, pp. 1092–1106, Dec. 2007.

[22] S. Spors and R. Rabenstein, “Spatial aliasing artifacts produced by linear and circular loudspeaker arrays used for Wave Field Synthesis,” in *120th Conv. Audio Eng. Soc.*, Conv. Paper 6711, Paris, France, May 2006.