1st EAA – EuroRegio 2010
Congress on Sound and Vibration
15 - 18 September 2010, Ljubljana, Slovenia
With Summer School for Young Researchers from 13 - 15 September 2010

Multi-band low-frequency room correction with chameleon subwoofer arrays

Adam J. HILL [a] and Malcolm O. J. HAWKSFORD [b]

Audio Research Laboratory, School of Computer Science and Electronic Engineering,
University of Essex, Colchester CO4 3SQ, UNITED KINGDOM
[a] ajhilla@essex.ac.uk [b] mjh@essex.ac.uk

ABSTRACT

Variations in the acoustic pressure response across a closed-space listening area result in inconsistent listening experiences dominant within the low-frequency band. An emerging solution proposed as the next evolutionary state in room correction systems involves an array of generalized subwoofers where each loudspeaker has a three-dimensional, frequency-dependent polar response operating over multiple low-frequency bands; a configuration termed a chameleon subwoofer array. System optimization is performed by measuring pressure responses at strategic room coordinates from which a set of orthogonal transfer functions is derived and applied to each degree of freedom within the array. The correction procedure benefits not only the set of discrete measurement points, but all points within the defined listening area. Performance is validated by finite-difference time-domain acoustic simulation taken over a random walk within a virtual acoustic listening space.

1. INTRODUCTION

Small listening rooms naturally exhibit highly position-dependent low-frequency responses. This is due to the buildup of standing waves between one, two or all three of the room's primary dimensions where integer multiples of the half-wavelength of certain frequencies fit perfectly. These are known as room modes which are distinctly perceivable below the Schroeder frequency which is calculated using the room volume and reverberation time (RT_{60}) [1]. A small-sized listening room will typically exhibit noticeable room modes below 120 Hz. Modes are present above this limit but are sufficiently dense in the spatial and frequency domains to not be strongly perceived due to modal overlapping.

There exist a number of well-known methods to address the problem of low-frequency spatial variance, ranging from simple passive techniques to highly complex active techniques. Each correction approach can improve certain aspects of the low-frequency response, but none has demonstrated full error suppression in terms of time and frequency domain variations.

This paper presents a hardware and software approach to full low-frequency error correction which aims to eliminate time and frequency domain variations across a large listening area while still maintaining system efficiency and room character. Existing low-frequency correction mechanisms will be analyzed to provide a benchmark for analysis followed by results from the new correction method using a three-dimensional virtual closed space.

2. COMMON CORRECTION TECHNIQUES

Numerous correction procedures are regularly used in practice. One of the more typical of these is increasing the walls' absorption levels to reduce the amount of acoustical reflections. This technique works well for high frequencies, but requires large absorbers for efficient low-frequency performance. In addition, added absorption will limit the reinforcement provided by room acoustics, resulting in greater demands on the system and also removing the room's acoustical characteristics, which may result in an unnatural quality [2].

Another simple passive method for limiting low-frequency spatial variance is strategic single/multiple subwoofer placement. This technique has been thoroughly discussed in [3, 4] concluding that subwoofer placement at wall midpoints on the floor gives the least spatial variance in the subwoofer operating band. This is due to the subwoofer(s) being located at nodes of one or more room mode giving the lowest source-to-room coupling factor thanks to destructive interference from wall reflections. The downside to this technique is the loss of system efficiency from the modal suppression via destructive acoustical interference [3].

The most common active low-frequency error correction approach creates an inverse equalization filter based on the measured response at a single listening location. This method has been proven to provide a maximally flat frequency response at the single target location, but often will not benefit any other location in the room since this technique does not affect the source-to-listener coupling factor due to the subwoofer's single degree of freedom [5]. This single point technique has been extended to take multiple listening location measurements into account with varying success [5-7]. Drawbacks associated with these systems are that some require continuous measurements for the adaptive filters and/or the systems can be highly listener-position sensitive (correction only applied to small local listening areas). Critically, time domain performance is rarely addressed explicitly within these low-frequency correction systems.

3. CHAMELEON SUBWOOFER ARRAYS

Historically, subwoofers contain a single drive unit, resulting in roughly an omnidirectional radiation pattern. This severely limits correction possibilities as each subwoofer in the system contributes only a single degree of freedom towards system correction. The next logical step in subwoofer design is to increase the degrees of freedom within each unit, thus allowing for more precise correction schemes.

A Chameleon Subwoofer Array (CSA), first described in [8] and motivated by the work in [9], consists of one or more hybrid subwoofer units where each controls four orthogonal, spherical harmonic source components; one omnidirectional and three dipole components. The correction scheme operates using a direct computational and calibration approach. Each individual source component within the system is activated sequentially and transfer function measurements obtained using maximum length sequences (MLS) for all target points within the defined listening area. For each measurement location these complex frequency response measurements are then used along with user-defined target responses to generate a set of correction filters via Eq. (1).

$$H = X^{-1}Y \tag{1}$$

where, H is an S x 1 matrix containing the correction filter coefficients, X is an L x S matrix containing the measured frequency responses and Y is an L x 1 matrix containing the desired frequency responses for each target location with S being the total number of source components and L the number of target points (L must equal S). The advantage of the CSA correction filter generation technique is that it operates using a direct calculation unlike other multi-point correction techniques which utilize time-consuming optimization routines [5-7].

The advantages of this single-equation direct calculation technique is that the method allows the target responses to be adjusted in real-time to meet the performance needs of the listeners. Specifically, this enables listeners to tailor the target response to their liking. The default response for the CSA is the measured room average response across all target points since this maintains room character and can achieve a more natural sound; in addition it eases the system requirements by avoiding compensation for any sharp nulls created by room modes. The conservation of the natural room sound endows CSA its “chameleon” descriptor, enabling the system to adopt the natural acoustic surroundings.

4. MULTI-BAND CORRECTION PROCEDURE

A few key considerations must be made in order to guarantee system stability when performing correction. First, it is necessary to bind the upper correction frequency to either the Schroeder frequency or the upper limit of the subwoofer band, whichever is lower. This ensures that correction is not applied within the diffuse frequency range, where the sound field is more random in nature. Applying a CSA to the diffuse field has revealed filter instability that can result in severe and even non-convergent time-domain ringing [8].

It is well understood that dipole sources lose radiation efficiency as frequency decreases [10, 11]; therefore, it is beneficial to split the CSA frequency range into two or more bands. With this in mind, limiting the amount of very low-frequencies sent to the dipoles will avoid over-excursion and even allow all drive units to contribute to room pressurization. The crossover point for a two-band correction system can be calculated using Eq. (2) based on the target point spacing. In addition, below a certain frequency there should be minimal sound pressure variations within the listening area, allowing for those frequencies to be reproduced without correction. This lower correction limit is calculated using Eq. (3).

$$f_X = {}^{c}\!/_{4D_L} \tag{2}$$

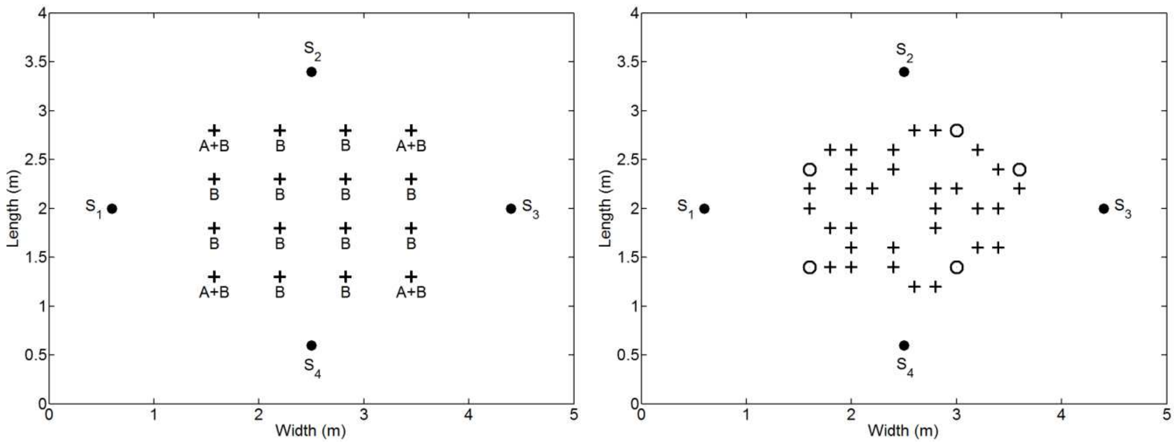
$$f_L = {}^{cD_H}\!\big/_{3D_L^2} \tag{3}$$

where, f_x is the omni-to-dipole crossover point (Hz), f_L is the lower correction limit (Hz), D_L is the mean spacing (m) of the omni-specific target points (maximally spaced from one another), D_H is the mean spacing (m) of all target points and c is the speed of sound in air (m/s). The crossover frequency is calculated to ensure that correction is applied only to frequencies where at least one quarter-cycle fits within the listening area using all available spherical harmonic components. Frequencies with longer wavelengths are corrected using just the omnidirectional components for improved efficiency and room pressurization.

5. SIMULATED CORRECTION RESULTS

The CSA low-frequency error correction method was tested within a finite-difference time-domain (FDTD) acoustical simulation toolbox developed within the University of Essex Audio Research Laboratory, as described in [12, 13]. To best judge the performance over the entire listening area (instead of restricting evaluation to the target points), a virtual walking path was designed to demonstrate correction benefits at both target and non-target points.

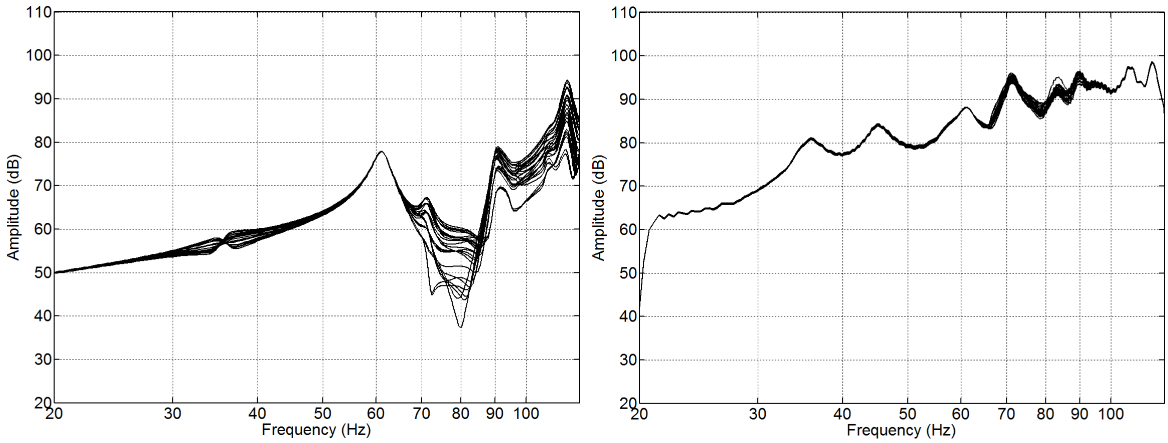
The four-subwoofer system with wall midpoint placement was chosen for the CSA layout as it provides the lowest spatial variance among the passive correction techniques. Since this CSA increases the degrees of freedom from four to sixteen, spatial variance should be significantly reduced. Both systems were corrected and tested as shown in Figures 1a and 1b, respectively. The resulting frequency responses are displayed in Figures 2a and 2b.



(a) (b)

Figure 1. Four-unit layout for (a) CSA correction filter calculation (A = omni, B = omni + dipole) and (b) 36-point virtual walking test (circles = target points, crosses = non-target points)

The spatial variance was measured based on a simulation of a single omnidirectional subwoofer in the room corner for reference (5.18 dB). The four omnidirectional subwoofer configuration gave a 2.16 dB spatial variance (58.3% improvement) while the four-unit CSA showed 0.31 dB (94.0% improvement). Clearly, the CSA method is capable of outperforming many conventional low-frequency correction methods. For completeness, additional configurations were simulated with their resulting spatial variances listed in Table 1.



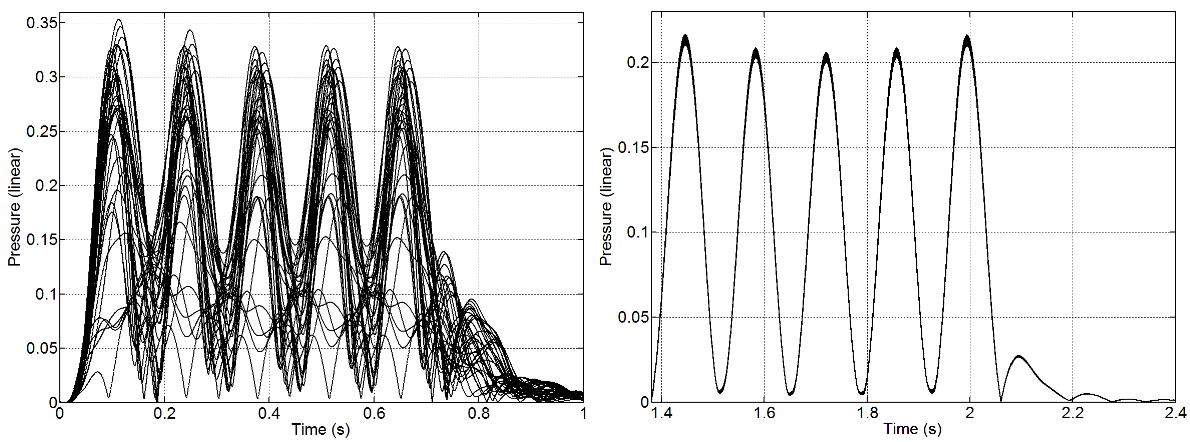
(a) (b)

Figure 2. Frequency response at points along a virtual walking path for (a) four omnidirectional subwoofers at wall midpoints and (b) a four-unit CSA at wall midpoints

Table 1. Spatial variance results for various correction methods

Correction Method (Configuration)	*Spatial Variance (dB)*	*Improvement*
CSA (4 units at room corners)	0.26462	94.9%
CSA (4 units at wall midpoints)	0.31295	94.0%
CSA (4 units across front of room)	0.54002	89.6%
Placement (4 omni subs at wall midpoints)	2.15880	58.3%
Placement (4 omni subs at room corners)	2.55420	50.7%
Placement (4 omni subs across front of room)	4.47040	13.7%
Single point EQ (1 sub at room corner)	5.17860	0.00%

It is also necessary to examine the time-domain transient response of the CSA system to ensure uniform performance across the listening area. The transient response was tested with the configuration in Figure 1(b) using tone bursts of a pure tone (110 Hz) [11] and monitoring the waveforms over the walking path (Figure 3). As with the overall frequency response, the CSA shows significant improvement in time-domain accuracy over the virtual walking path.



(a) (b)

Figure 3. Simulated waveforms along a virtual walking path from 110 Hz tone burst simulation for (a) uncorrected system and (b) CSA corrected system

6. CONCLUSIONS & FUTURE WORK

A new approach to low-frequency room response error correction has been presented, termed a Chameleon Subwoofer Array (CSA). The technique exploits an array of multi-source component subwoofers to increase the system degrees of freedom and to enhance correction accuracy over a large listening area. The CSA method has shown in simulations to remove over 90% of spatial variance in both the frequency and time domains, as compared to conventional methods which often can only achieve around 50% of improvement without much attention paid to time domain performance.

Work is underway towards building a CSA prototype for real-world testing. In addition, virtual bass signal processing is currently being investigated as a means to subjectively reinforce difficult room modes encountered within the CSA domain. Advanced applications of the system have been hypothesized to include discrete real-time target response control for each individual in the listening area as well as useful live sound implementations.

7. REFERENCES

[1] Schroeder, M.R.; K.H. Kuttruff. "On frequency response curves in rooms. Comparison of experimental, theoretical, and Monte Carlo results for the average frequency spacing between maxima." Journal Acoustical Soc. of America, Vol. 34, Issue 1, 1962, pp. 76-80.

[2] Toole, Floyd E. "Loudspeakers and rooms for sound reproduction – A scientific review." Journal of the Audio Engineering Soc., Vol. 54, No. 6, 2006, pp. 451-476.

[3] Toole, Floyd E. *Sound Reproduction: Loudspeakers and Rooms.* Focal Press, New York, 2008.

[4] Welti, Todd; Allan Devantier. "Low-frequency optimization using multiple subwoofers." Journal of the Audio Engineering Soc., Vol. 54, No. 5, 2006, pp. 347-364.

[5] Elliott, S.J.; P.A. Nelson. "Multiple-point equalization in a room using adaptive digital filters." Journal of the Audio Engineering Soc., Vol. 27, No. 11, 1989, pp. 899-907.

[6] Pedersen, Jan Abildgaard. "Adjusting a loudspeaker to its acoustic environment – The ABC system." Audio Engineering Soc. Convention #115, paper 5880. October, 2003.

[7] Bharitkar, Sunil; Chris Kyriakakis. "Nonlinear projection algorithm for evaluating multiple listener equalization performance." Audio Engineering Soc. Convention #117, paper 6206. Oct, 2004.

[8] Hill, A.J.; M.O.J Hawksford. "Chameleon subwoofer arrays – Generalized theory of vectored sources in a close acoustic space." Audio Engineering Soc. Convention #128, paper 8074. May, 2010.

[9] Howe, R.M.; M.O.J. Hawksford. "Methods of local room equalization and their effect over the listening area." Audio Engineering Soc. Convention #91, paper 3138. Oct, 1991.

[10] Olson, Harry F. "Gradient Loudspeakers." Journal of the Audio Engineering Soc., Vol. 21, No. 2, 1973, pp. 86-93.

[11] Linkwitz, Siegfried. "Investigation of sound quality differences between monopolar and dipolar woofers." Audio Engineering Soc. Convention #105, paper 4786. Sept, 1998.

[12] Hill, A.J.; M.O.J. Hawksford. "Visualization and analysis tools for low frequency propagation in a generalized 3D acoustic space." Audio Engineering Soc. Convention #127, paper 7901. Oct, 2009.

[13] Hill, A.J.; M.O.J. Hawksford. "Seeing through the darkness: Visualizing low frequency behavior in small rooms." Proc. Institute of Acoustics, Reproduced Sound 25. Nov, 2009.