Audio Engineering Society
Convention Paper

Presented at the 132nd Convention
2012 April 26–29 Budapest, Hungary

This paper was peer-reviewed as a complete manuscript for presentation at this Convention. Additional papers may be obtained by sending request and remittance to Audio Engineering Society, 60 East 42nd Street, New York, New York 10165-2520, USA; also see www.aes.org. All rights reserved. Reproduction of this paper, or any portion thereof, is not permitted without direct permission from the Journal of the Audio Engineering Society.

Practical applications of chameleon subwoofer arrays

Adam J. Hill[1] and Malcolm O. J. Hawksford[2]

[1] ajhilla@essex.ac.uk [2] mjh@essex.ac.uk
Audio Research Laboratory, School of Computer Science and Electronic Engineering,
University of Essex, Colchester, Essex CO4 3SQ, UK

ABSTRACT

Spatiotemporal variations of the low-frequency response in a closed-space are predominantly caused by room-modes. Chameleon subwoofer arrays (CSA) were developed to minimize this variance over a listening area using multiple independently-controllable source components and calibrated with one-time measurements. Although CSAs are ideally implemented using hybrid (multiple source component) subwoofers, they can alternatively be realized using conventional subwoofers. This capability is exploited in this work where various CSA configurations are tested using commercially-available subwoofers in a small-sized listening room. Spectral and temporal evaluation is performed using tone-burst and maximum length sequence (MLS) measurements. The systems are implemented with practicality in mind, keeping the number of subwoofers and calibration measurements to a minimum while maintaining correction benefits.

1. INTRODUCTION

It is often the case that a low-frequency room-mode correction method performs extremely well in virtual/laboratory testing, but lacks in the area of real-world practicality. This work aims to highlight how the chameleon subwoofer array (CSA) correction methodology, which performs remarkably well in the virtual domain, can be adjusted to operate within an actual sound reproduction system purely in the form of digital signal processing (DSP).

Correcting for the detrimental effects of low-frequency room-modes is a longstanding problem within the domain of small-room acoustics. The modes are a result of standing waves between one or more set of parallel surfaces within a space, and occur in rectangular rooms at frequencies predicted by Eq. 1.1 [1]. Room-modes occur in non-rectangular spaces as well, but are more difficult to predict using closed-form solutions.

$$f_m = \frac{c}{2}\sqrt{\left(\frac{\eta_x}{l_x}\right)^2 + \left(\frac{\eta_y}{l_y}\right)^2 + \left(\frac{\eta_z}{l_z}\right)^2} \tag{1.1}$$

where the modal frequency, f_m, with integer modal indices, η_x, η_y and η_z, is predicted based on the rectangular room dimensions, l_x, l_y and l_z (m), and the speed of sound in air, c (m/s). Although room-modes exist across the audible frequency spectrum, at higher frequencies the modes are spatially and spectrally dense and therefore lack individual identifiable character.

A common estimate for the boundary between the discrete modal and diffuse sound fields is known as the Schroeder frequency, f_s, and is calculated with Eq. 1.2 [2] using the -60 dB reverb time, RT_{60} (s), and the room volume, V (m^3). Controlling the discrete room-modes, which fall below the Schroeder frequency, is the focus of this research.

$$f_s = 2000\sqrt{\frac{RT_{60}}{V}} \tag{1.2}$$

Numerous proposals for room-mode correction have been previously published. Some proposals are passive in nature, calling for physical alterations to the system. These methods include room dimension ratio optimization [3,4], single-source placement [1,5], multiple-source placement [1,5-7] and passive absorption [8,9]. There also exist many techniques involving signal processing in the form of single-point [10,11] and multiple-point [12-15] equalization, source polar pattern control [16,17], radiation resistance-based correction [18], ambisonics-style equalization [19] and active absorption [20,21]. Each technique exhibits its own unique advantages, disadvantages and overall correction capabilities. A detailed analysis of many of these methods is available in Chapter 2 of [22].

This paper concentrates on a practical CSA implementation in an attempt to illustrate how the technology can translate from a somewhat complicated proposal (in terms of required hardware that nonetheless performs very well in simulations) to an easily-realizable system that can be accommodated within most multi-channel home theater sound reproduction systems. Various configurations are examined using maximum length sequence (MLS) and tone-burst measurements to highlight the spectral and temporal benefits of CSA correction.

The CSA room-mode correction method is described in Section 2, highlighting an idealized implementation as well as the practical application of this technique within a standard home theater system. This description is followed by the experimental procedure and results presentation/analysis (Section 3) along with concluding remarks, including proposed future work (Section 4).

2. CHAMELEON SUBWOOFER ARRAYS

Chameleon subwoofer arrays (CSA) operate on the principle that increasing the available independently-controllable source components (degrees of freedom) enables greater manipulation of the low-frequency response over a wide-area. This correction methodology was first described in [23] having been inspired by the approach in [6], the quasi frequency-dependent subwoofer polar pattern control in [17] and the ambisonics-style room correction system in [19].

Ideally, CSAs are realized using *hybrid subwoofers*. These loudspeakers facilitate four source components: one omnidirectional and three dipolar (one in each primary spatial dimension) which correspond to the zero- and first-order components in an ambisonics configuration [19]. Hybrid subwoofers provide three more degrees of freedom than most conventional subwoofers, all within a similarly-sized enclosure. A practical upper limit of four subwoofers for home use has been proposed in [5] which correspond to sixteen degrees of freedom (as opposed to four using conventional units).

Because this CSA implementation requires new hardware, a reasonable alternative is to use CSA-specific DSP within existing sound reproduction systems. Although the degrees of freedom are likely to be limited, this approach requires little or no additional hardware since the DSP algorithm can be embedded within the existing processor.

System measurements (using MLS) are taken at target points covering a listening area with only one subwoofer source component activated at a time. The correction frequency range is limited by the mean target point spacing where the upper bound is determined by the corresponding quarter-wavelength equal to the point spacing. Target point spacing of 0.7 m limits correction to below approximately 120 Hz, which is generally the limit of the subwoofer operating band.

CSA filter synthesis is performed using a direct calculation procedure (Eq. 2.1) where the complex correction coefficients for each source component (H_{NSx1}, NS = number of components) are calculated by multiplying the inverse of the measured frequency response due to each individual source component at target points (X_{NLxNS}, NL = number of targets) with a set of target frequency responses for each listening location (Y_{NLx1}). The target responses are set, by default, to the measured room average response to preserve the natural room characteristics [20].

$$H_{NSx1} = X_{NLxNS}^{-1} Y_{NLx1} \tag{2.1}$$

Although Eq. 2.1 normally requires a square X matrix (equal number of source components and measurement points), this requirement can be relaxed using a pseudo-inverse matrix operation to enable increased spatial sampling of the target listening area.

In certain cases the direct calculation can result in excessively large correction coefficient amplitudes and addressed as follows: First, the measurement matrix's condition number is analyzed at each frequency bin to determine its sensitivity to system noise. Frequencies components with condition numbers above a set threshold are then attenuated and substituted with *virtual bass components* designed to subjectively reinforce the attenuated narrow bands (as detailed in [24]). Secondly, the overall amplitudes of the filter frequency responses are scaled so that the highest filter amplitude is at unity before amplification. Experimental tests have confirmed these techniques result in practical filters, maintaining efficient operation.

The experimental systems presented here utilize the alternative CSA implementation where DSP is applied to an existing sound reproduction system. A complete description and analysis of the CSA correction methodology is given in Chapter 5 of [22].

3. PRACTICAL IMPLEMENTATION

The DSP-only CSA low-frequency room-mode correction approach was applied to the primary sound reproduction system in the University of Essex Audio Research Laboratory listening room (dimensions: 8.16m x 6.20 m x 2.74 m). The system provided two independently-controllable subwoofer channels, hence two available degrees of freedom. The system crossover frequency was set to 80 Hz, giving a target correction range of 20 – 80 Hz. Two Bowers & Wilkins ASW 750 subwoofers were utilized in the following configurations:

A. Units located at wall midpoints directly to the left and right of the listening area

B. Units located at opposing room corners (front right and rear left)

C. One unit located at the front right room corner and the other at the right wall midpoint

The listening area measured 2.2 m x 2.2 m and was positioned approximately at the center of the room. Sixteen measurement points were equally spaced over the area (0.7 m spacing), providing accurate spatial sampling up to 120 Hz. The two degree of freedom CSA with sixteen target listening points took approximately six minutes to calibrate. The correction filter frequency responses for each configuration are shown in Fig 3.1.

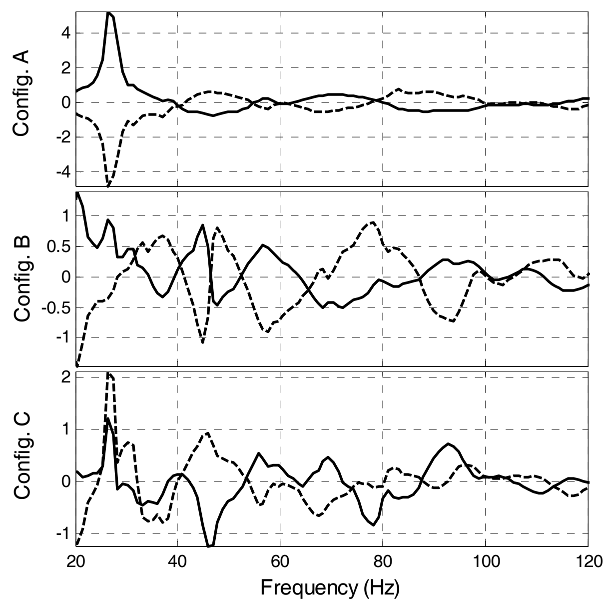

Fig 3.1 CSA correction filter frequency responses for configurations A, B and C (solid line = subwoofer 1, dashed line = subwoofer 2, linear amplitude scale)

All measurements were taken using a bespoke toolbox in Matlab [25] and a MOTU 2408 Mk3 multi-channel external sound card [26]. A 13^{th} order MLS served as the measurement excitation signal with a sample rate of 4 kHz. Averaging was used to reduce measurement noise/distortion from the MLS [27].

3.1. Correction performance: Steady-state

Each configuration was evaluated over a twenty-five point walking path covering the entire listening area as shown in Fig. 3.2 (the walking path points did not coincide with the sixteen measurement points). MLS measurements were taken for both the uncorrected and corrected systems and compared by calculating the change in spatial variance (Eq. 3.1) and mean output level (Eq. 3.2) [1,6].

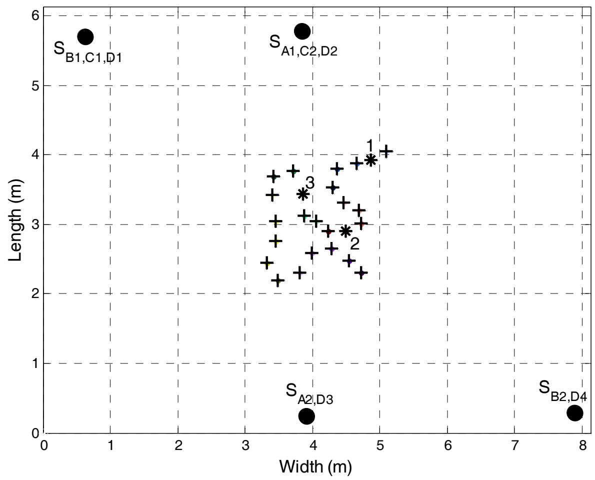

Fig. 3.2 Subwoofer configurations and 25-point walking path used for CSA performance evaluation (starred/numbered points indicate tone burst measurement points, as discussed in Section 3.2, configuration D discussed in Section 3.3)

$$SV = \frac{1}{N_f} \sum_{i=f_{lo}}^{f_{hi}} \sqrt{\frac{1}{N_p - 1} \sum_{p=1}^{N_p} \left(L_p(p,i) - \overline{L_p(i)} \right)^2} \quad (3.1)$$

$$MOL = \frac{1}{N_f N_p} \sum_{i=f_{lo}}^{f_{hi}} \sum_{p=1}^{N_p} L_p(p,i) \quad (3.2)$$

where the spatial variance (*SV*) and mean output level (*MOL*) are calculated over a frequency range, f_{lo} to f_{hi}, across N_p listening locations and N_f frequency bins. $L_p(p,i)$ represents the sound pressure level (dB) at position, p, and frequency bin, i. $\overline{L_p(i)}$ is the mean sound pressure level (dB) over all listening locations at frequency bin, i.

For additional illustration of the correction performance, the MLS measurements were used to calculate the individual frequency responses over the walking path before and after correction and plotted for direct examination. The resulting plots for configurations A, B and C are given in Figs. 3.3 – 3.5, respectively.

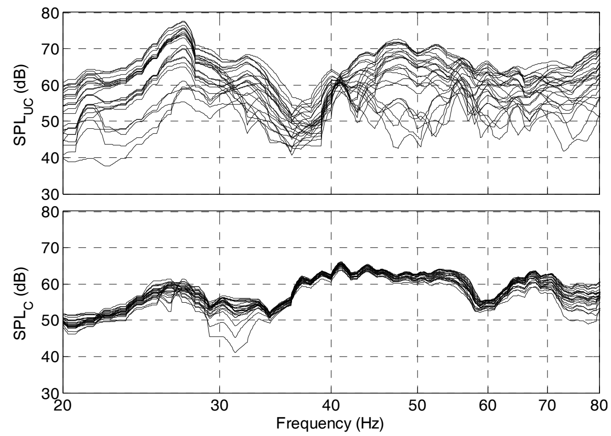

Fig. 3.3 Uncorrected (top) and corrected (bottom) frequency responses over a 25-point walking path using system configuration A

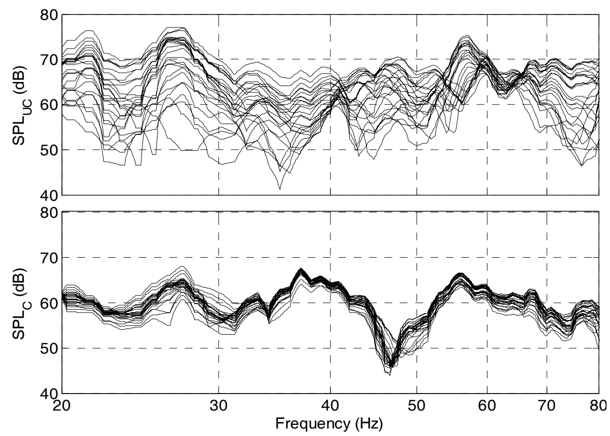

Fig. 3.4 Uncorrected (top) and corrected (bottom) frequency responses over a 25-point walking path using system configuration B

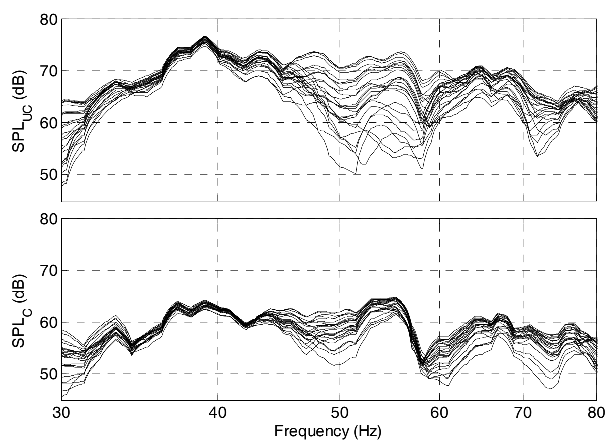

Fig. 3.5 Uncorrected (top) and corrected (bottom) frequency responses over a 25-point walking path using system configuration C

The calculated uncorrected vs. corrected system comparisons are as follows:

- Configuration A
  - Uncorrected SV = 5.43 dB
  - Corrected SV = 1.25 dB
  - SV reduction = 76.8%
  - Uncorrected MOL = 59.4 dB
  - Corrected MOL = 57.5 dB
  - MOL reduction = 3.1%
- Configuration B
  - Uncorrected SV = 4.83 dB
  - Corrected SV = 1.36 dB
  - SV reduction = 71.8%
  - Uncorrected MOL = 63.1 dB
  - Corrected MOL = 59.4 dB
  - MOL reduction = 5.7%
- Configuration C
  - Uncorrected SV = 2.46 dB
  - Corrected SV = 1.47 dB
  - SV reduction = 40.1%
  - Uncorrected MOL = 66.8 dB
  - Corrected MOL = 58.3 dB
  - MOL reduction = 12.7%

CSA correction delivers over 70% reduction in spatial variance for configurations A and B, without significant mean output level attenuation. Considering that the ideal CSA has sixteen degrees of freedom while these experimental systems contain only two attempting to correct over sixteen target points, this is considered an excellent result that strengthens the argument that practical CSAs can be implemented (often where ideal CSA conditions cannot be met) and still retain high correction benefits over a wide listening area.

Configuration C highlights the observation emphasized in Chapter 5 of [22] whereby CSA correction performs best with maximal spacing of the sources. In configuration C the two units are spaced within 3 m of one another (as opposed to over 6 m separation in configurations A and B) and as a result only provides efficient correction down to 30 Hz. Below 30 Hz, wavelengths are sufficiently long to where 3 m spacing corresponds to approximately a quarter-wavelength and closely resembles Olson's first-order gradient loudspeakers [16] as opposed to two distinct sources.

Close source spacing also results in reduced mean output level, which again can be explained with Olson's work [16]. These source spacing findings are in agreement with previous simulations which provide further indication that wide source separation is desirable.

Configurations A and B benefit from source spacing great enough to allow for distinct sound sources throughout the subwoofer band, hence the improved CSA correction performance, both in spatial variance reduction and mean output level.

3.2. Correction performance: Transients

Following the doctrine of Linkwitz [28,29], it is desirable to ensure transient behavior is maintained (and hopefully improved) after system correction. Configurations A and B were tested again, this time utilizing tone-bursts to specifically examine the transient response. Configuration C was excluded due to its non-ideal source layout.

Measurements were taken at three locations over the listening area (see Fig. 3.2) using tone-bursts (ten cycles per burst repeated three times) at 34.7, 50.2 and 74.4 Hz (each of which is a strong modal frequency). A raised cosine was used as the burst window, according to Linkwitz's investigations [28, 29]. The source and resulting waveforms are shown in Figs. 3.6 and 3.7 – 3.12, respectively.

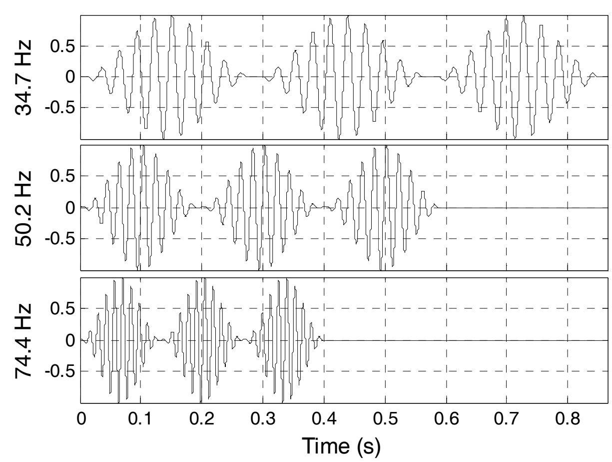

Fig. 3.6 Source tone burst waveforms (linear amplitude scale)

There are a number of interesting characteristics illuminated by the tone-burst testing. The uncorrected signals show differences in received amplitude and waveform shape between measurement points. In extreme cases (such as points 1 and 2 in Fig. 3.12) the received modulation frequency is actually double than expected, giving the impression of twice as many pulses in the test signal. In other cases (such as point 3 in Fig. 3.8) the response is smeared to the point where it is difficult to discern any pulses at all. Clearly, the transient behavior of a system must be considered within a correction algorithm; a point which is emphasized in [20,21,23].

The corrected tone-burst measurements highlight a number of key factors. First, CSA correction by default corrects to the average measured response. If the overall listening area naturally exhibits a poor transient response, then the corrected response is unlikely to exhibit much improvement, but will be more consistent across the listening area. An example of this is shown in Fig. 3.11 where the corrected measurements retain some level of waveform smearing. This issue can be addressed by targeting a flat frequency response, although prior simulations have indicated that this can potentially reduce system efficiency.

Aside from the highlighted cases, the comparison of tone-burst behavior before and after CSA correction indicates that transient behavior is handled properly within the algorithm as the corrected responses are very similar both in amplitude and waveform shape.

3.3. Higher-order CSA correction performance

Although this paper focuses on CSA correction applied to practical sound reproduction systems (maximum of two subwoofers), a higher-order CSA was configured to highlight the benefits of additional degrees of freedom (filter responses shown in Fig. 3.13). This system consisted of two Bowers & Wilkins ASW 750 subwoofers and two KEF PSW 1000.2 subwoofers. The units were arranged as a combination of configurations A and B (see Fig 3.2). The listening area was identical to the previous tests with the measured frequency responses shown in Fig. 3.14.

This configuration gives an 81.0% reduction in spatial variance with a mean output level reduction of 9.9%. The doubling of degrees of freedom does not result in a doubling of correction benefits. In this case spatial variance reduction is improved by around 10% (as compared to configurations A and B), but there is a lower mean output level than with the two-unit configurations. The decreased MOL is mostly due to the close proximity of the subwoofer pairs and can be reduced by increasing the mean source spacing.

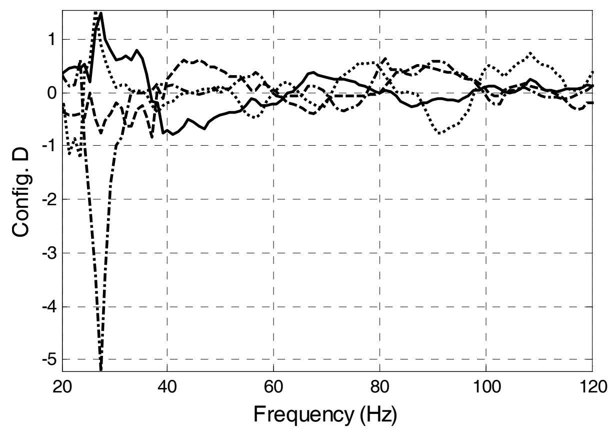

Fig. 3.13 CSA correction filter frequency responses for configuration D (4 subwoofers, linear amplitude scale)

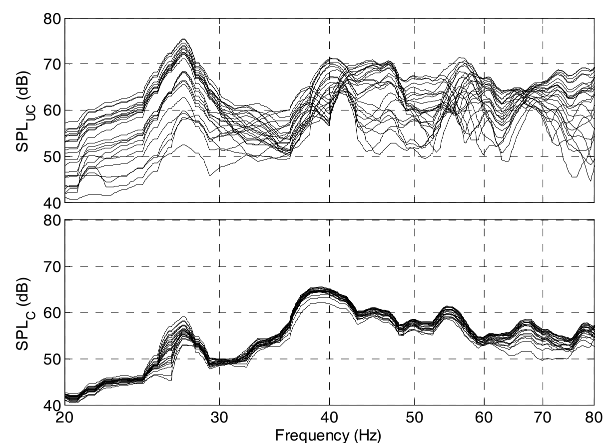

Fig. 3.14 Uncorrected (top) and corrected (bottom) frequency responses over a 25-point walking path using the four-subwoofer system configuration

The expanded, four-subwoofer system highlights the benefits of additional degrees of freedom. It is expected that as the available degrees of freedom approaches the number of target points, correction benefits will reach a maximum. This is beyond the scope of this current investigation, which focuses on practical applications, however a thorough analysis of CSA configuration and calibration techniques is presented in Chapter 5 of [22].

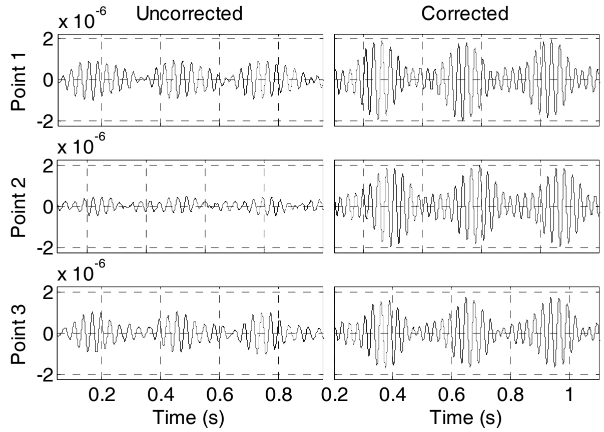

Fig. 3.7 Tone-burst measurements (34.7 Hz) at three locations using system configuration A

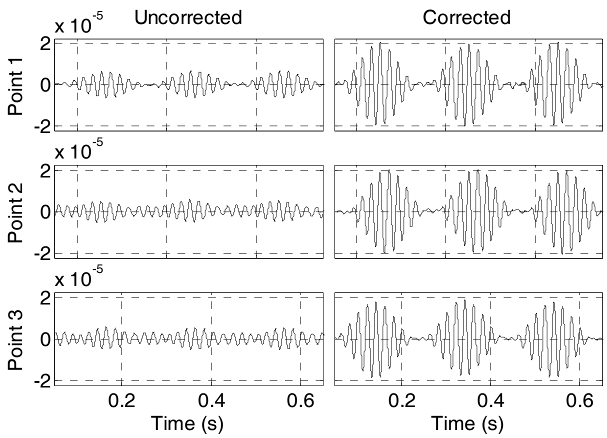

Fig. 3.8 Tone-burst measurements (50.2 Hz) at three locations using system configuration A

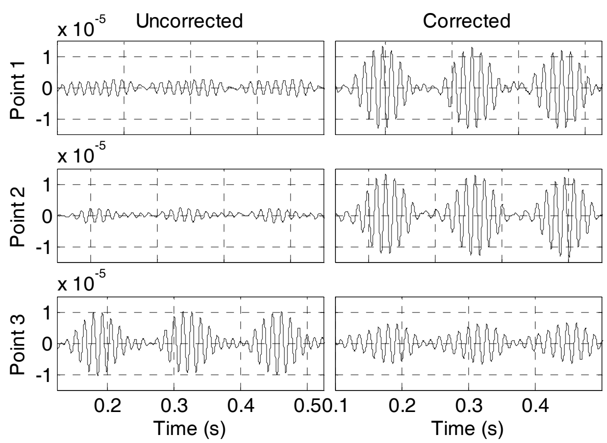

Fig. 3.9 Tone-burst measurements (74.4Hz) at three locations using system configuration A

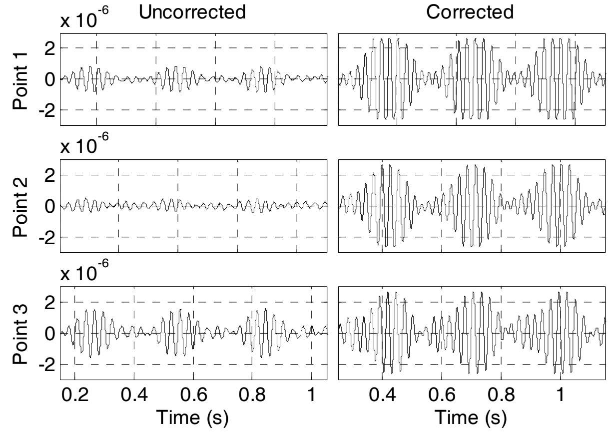

Fig. 3.10 Tone-burst measurements (34.7 Hz) at three locations using system configuration B

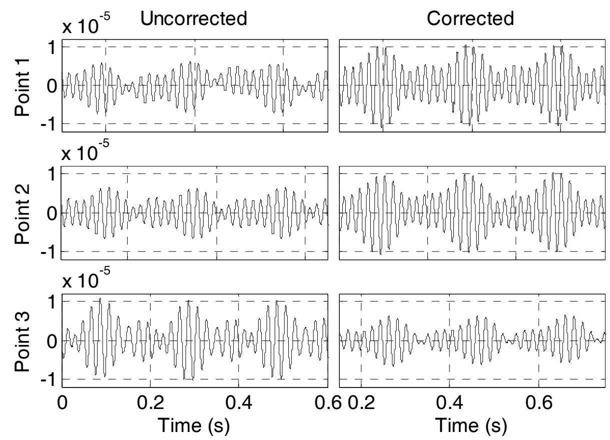

Fig. 3.11 Tone-burst measurements (50.2 Hz) at three locations using system configuration B

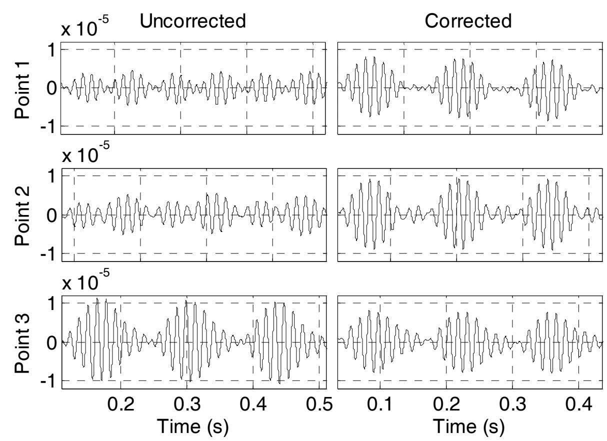

Fig. 3.12 Tone-burst measurements (74.4Hz) at three locations using system configuration B

4. CONCLUSIONS & FUTURE WORK

This investigation aimed to determine how well CSA low-frequency room-mode correction can perform when restricted to a typical, commercially-available sound reproduction system. Ideally, a CSA consists of up to four hybrid subwoofers, granting sixteen degrees of freedom. The tested system in this case allowed for two degrees of freedom (two independently-controllable subwoofers). Additionally, the number of required measurements was constrained to allow for system calibration in a reasonable time frame. Calibration was performed using sixteen measurement points, in this case, which should theoretically provide accurate spatial sampling beyond the generalized subwoofer band of 20 – 120 Hz. This process took on average six minutes.

The CSA-corrected two-subwoofer system resulted in over 70% spatial variance reduction in two of the three tested configurations as well as generally maintaining the mean output level and waveform accuracy. The third configuration resulted in 40% spatial variance reduction and a loss in output efficiency, which is attributed to the non-ideal source layout.

Expanding the experimental system to four-subwoofers gave an additional 10% spatial variance reduction, although with a reduced mean output level, again due to the source configuration. Further expansion of the system should give even greater low-frequency response control, but is beyond the scope of this work as the focus is on practical implementations.

Future work must focus on streamlining the calibration process which hopefully will lead to embedding the algorithm within an existing home theater processor. Additionally, future experiments will focus on implementing individualized frequency response control where each listener is granted real-time control of their localized response [30] as well as utilizing the low-frequency reproduction capabilities of all system loudspeakers, such as with a 5.1 surround system [22].

The experimental results presented in this work strongly indicate that DSP-based CSA correction can be implemented using most existing sound reproduction system provided the system lends more than one independently controllable low-frequency source. Even while limited to two degrees of freedom, spatial variance can be significantly reduced, thus providing all listeners an objectively-equal listening experience.

5. REFERENCES

[1] Toole, F.E. *Sound Reproduction: Loudspeakers and Rooms*. Focal Press, Boston. 2008.

[2] Schroeder, M.R. "The 'Schroeder frequency' revisited." Journal of the Acoustical Society of America, vol. 99, no. 5, pp 3240-3241. May, 1996.

[3] Walker, R. "Optimum dimension ratios for small rooms." 100th Convention of the Audio Engineering Society, paper 4191. May, 2002.

[4] Milner, J.R; R.J. Bernhard. "An investigation of the modal characteristics of nonrectangular reverberation rooms." Journal of the Acoustical Society of America, vol. 85, no. 2, pp. 772-779. February, 1989.

[5] Welti, T. "How many subwoofers are enough." 112th Convention of the Audio Engineering Society, paper 5602. May, 2002.

[6] Welti, T.; A. Devantier. "Low-frequency optimization using multiple subwoofers." Journal of the Audio Engineering Society, vol. 54, no. 5, pp. 347-364. May, 2006.

[7] Celestinos, A.; S. Birkedal Nielsen. "Optimizing placement and equalization of multiple low frequency loudspeakers in rooms." 119th Convention of the Audio Engineering Society, paper 6545. October, 2005.

[8] Fletcher, J.A. "The design of a modular sound absorber for very low frequencies." BBC Research Department, report number RD1992/10. 1992.

[9] Reiley, E.; A. Grimani. "Room mode bass absorption through combined diaphragmatic & Helmholtz resonance techniques: 'The Springzorber.'" 114th Convention of the Audio Engineering Society, paper 5760. March, 2003.

[10] Genereux, R.P. "Adaptive loudspeaker systems: correcting for the acoustics environment." 8th International Conference of the Audio Engineering Society. May, 1990.

[11] Kuriyama, J.; Y. Furukawa. "Adaptive loudspeaker system." Journal of the Audio Engineering Society, vol. 37, no. 11, pp. 919-926. November, 1989.

[12] Elliot, S.J.; P.A. Nelson. "Multiple-point equalization in a room using adaptive digital filters." Journal of the Audio Engineering Society, vol. 37, no. 11, pp. 899-907. November, 1989.

[13] Elliot, S.J.; L.P. Bhatia; D.W. Deghan; A.H. Fu; M.S. Stewart; D.W. Wilson. "Practical implementation of low-frequency equalization using adaptive digital filters." Journal of the Audio Engineering Society, vol. 42, no. 12, pp. 988-998. December, 1994.

[14] Santillan, A.O. "Spatially extended sound equalization in rectangular rooms." Journal of the Acoustical Society of America, vol. 110, no. 4, pp. 1989-1997. October, 2001.

[15] Bharitkar, S.; C. Kyriakakis. "Objective function and automatic multi-position equalization and bass management filter selection." 119th Convention of the Audio Engineering Society, paper 6608. October, 2005.

[16] Olson, H.F. "Gradient Loudspeakers." Journal of the Audio Engineering Society, vol. 21, no. 2, pp. 86-93. March, 1973.

[17] Backman, J. "Low-frequency polar pattern control for improved in-room response." 115th Convention of the Audio Engineering Society, paper 5867. October, 2003.

[18] Pedersen, J.A. "Adjusting a loudspeaker to its acoustic environment – The ABS system." 115th Convention of the Audio Engineering Society, paper 5880. October, 2003.

[19] Howe, R.M.; M.O.J. Hawksford. "Methods of local room equalization and their effect over the listening area." 91st Convention of the Audio Engineering Society, paper 3138. October, 1991.

[20] Vanderkooy, J. "Multi-source room equalization: Reducing room resonances." 123rd Convention of the Audio Engineering Society, paper 7262. October, 2007.

[21] Makivirta, A.; P. Antsalo; M. Karjalainen; V. Valimaki. "Modal equalization of loudspeaker-room responses at low frequencies." Journal of the Audio Engineering Society, vol. 51, no. 5, pp. 324-434. May, 2003.

[22] Hill, A.J. "Analysis, modeling and wide-area spatiotemporal control of low-frequency sound reproduction." Ph.D. Thesis, University of Essex, UK. January, 2012.

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

[24] Hill, A.J.; M.O.J. Hawksford. "Wide-area psychoacoustic correction for problematic room-modes using nonlinear synthesis." Journal of the Audio Engineering Society, vol. 59, no. 11, pp. 825-834. November, 2011.

[25] MATLAB version 7.6.0. Natick, Massachusetts: The MathWorks Inc., 2008.

[26] Mark of the Unicorn, Inc. "MOTU PCI-424 User's Guide for Windows." Cambridge, MA, USA. 2002. cdn-data.motu.com/manuals/pci-audio/PCI-424_User_Guide_Win.pdf

[27] Kemp, J.A. "Improvements to bore profile measurement in acoustic pulse reflectometry." Proc. Institute of Acoustics, vol. 30, pt. 2. 2008.

[28] Linkwitz, S. "Narrow band impulse testing of acoustical systems." 60th Convention of the Audio Engineering Society, paper 1342. May, 1978.

[29] Linkwitz, S. "Shaped tone-burst testing." Journal of the Audio Engineering Society, vol. 28, no. 4, pp. 250-258. April, 1980.

[30] Hill, A.J.; M.O.J. Hawksford. "Individualized low-frequency response manipulation for multiple listeners using chameleon subwoofers arrays." Proc. 3rd Computer Science and Electronics Engineering Conference (CEEC'11), IEEE Xplore, Colchester, UK. July, 2011.