Audio Engineering Society

Convention Paper 8074

Presented at the 128th Convention
2010 May 22–25 London, UK

Chameleon subwoofer arrays – Generalized theory of vectored sources in a closed acoustic space

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ABSTRACT

An equalization model is presented that seeks optimal solutions to wide area low-frequency sound reproduction in closed acoustic spaces. The methodology improves upon conventional wisdom by incorporating a generalized subwoofer array where individual frequency dependent loudspeaker polar responses are described by complex spherical harmonic frequency dependent functions. Multi-point system identification is performed using 3-dimensional finite-difference time-domain simulation with optimization applied to seek global equalization represented by a set of orthogonal transfer functions applied to each spherical harmonic of each subwoofer within the array. The system is evaluated within a 3-dimensional virtual acoustic space using both time and frequency domain metrics.

1. INTRODUCTION

Small rooms naturally experience great variation in low-frequency levels across the listening area. These variations arise from low-frequency half-wavelengths that fit perfectly within one or more dimensions of a room, resulting in standing waves, commonly referred to as room modes. These room modes often cause extreme pressure variations, diminishing the perceived quality of a loudspeaker system within a room.

Room modes are generally an issue in the frequency band below the diffuse (reverberant) range. A commonly accepted boundary for this is known as the Schroeder frequency [1]. Above this boundary, room modes are sufficiently dense to not be perceived by the human ear due to masking effects.

Methods have been proposed and implemented in previous research that attempt to address this issue of low-frequency correction. Some of these methods focus on strategic placement of single or multiple subwoofers while other methods attempt to provide correction by active means involving room measurements with the corresponding signal processing. While each of these methods can provide correction to certain aspects of low-frequency error within a space, none are capable of full error suppression in terms of time and frequency domain variations.

This paper presents an approach to low-frequency error correction that aims to provide error-free source signal reproduction in both the time and frequency domains across a large listening area within a generalized space. The goals for this new approach are to provide an easily implementable system that has low sensitivity to changes within a room while providing equal low-frequency energy across the subwoofer operating range for all points within a targeted listening area.

Existing low-frequency correction techniques will be analyzed to provide a benchmark for the new method to be judged against. The new correction method will first be tested on its core mathematical level using data from simulations and then applied to various three-dimensional simulations to judge its performance.

2. EXISTING LOW-FREQUENCY CORRECTION TECHNIQUES

Many different forms of low-frequency error correction exist as a result of years of research within the audio engineering community. Some of these methods are purely passive in nature, where correction is achieved by well-informed placement of both the system's subwoofer(s) and listening area. Other methods provide active correction, whether by means of active absorption or clever filter implementation based on room measurements.

Each of these techniques has positive and negative aspects that must be considered when choosing an appropriate approach. Key correction methods will be briefly discussed with the support of simulation data analysis.

All simulations within this paper were performed using a Finite-Difference Time-Domain (FDTD) acoustics simulation toolbox developed within the Audio Research Laboratory at the University of Essex [2][3]. FDTD was chosen as the simulation method due to its high accuracy below the diffuse frequency range as well as its great flexibility regarding simulation parameters.

2.1. Passive correction – source placement

Proper subwoofer placement is critical for a system with minimal signal processing capabilities. The definition of "proper," though, can vary depending on the user's requirements of the subwoofer system.

Often the goal is to achieve maximum low-frequency output without high levels of amplification. This has been demonstrated by many researchers to be achievable with subwoofer to room mode coupling in mind [4 – 7]. When a subwoofer is placed at an anti-node of a room mode, coupling will be maximized. When the subwoofer is placed at a node coupling will be minimized (theoretically zero) due to placement at the standing wave's zero crossing [11].

Generally, the optimal subwoofer position with pure low-frequency output in mind is any room corner. Room modes tend to have anti-nodes at room corners; therefore all the modes will be maximally excited with corner placement (Figure 2.1). Placing the subwoofer in the corner has the added benefit of the Waterhouse effect, where each nearby boundary contributes a 3 dB boost to the pressure level, giving a 9 dB boost when in a corner [8].

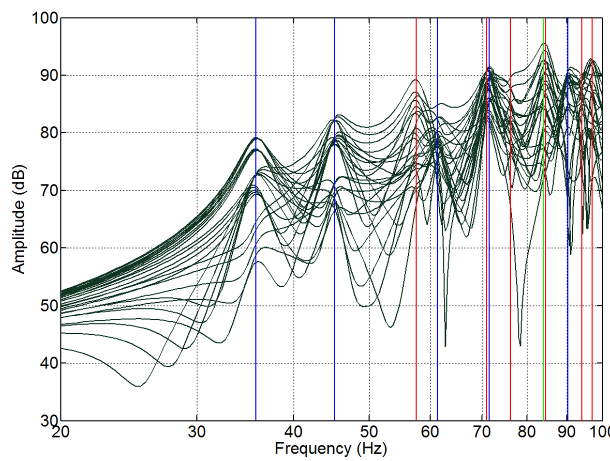

Figure 2.1: Simulated frequency response at 25 listening locations with a single subwoofer in a room corner (blue lines = axial modes, red lines = tangential modes, green lines = oblique modes)

While this simple technique sends ample low-frequency energy into the room, it does not provide an equal response at all listening points. This is due to the strong dependence on source to listening location coupling. A listener's position in regard to the nodes and antinodes of the room modes will give very different responses at varying locations [7].

The single corner subwoofer technique falls short in providing equal coverage over a large listening area. To provide equal coverage at low-frequencies, a single subwoofer should be placed as close to as many nodes as possible. The center of a rectangular room is generally a common node for many room modes. Although placement at the center of a room is not necessarily practical, it can effectively suppress many room modes providing a more even coverage to all listeners (Figure 2.2).

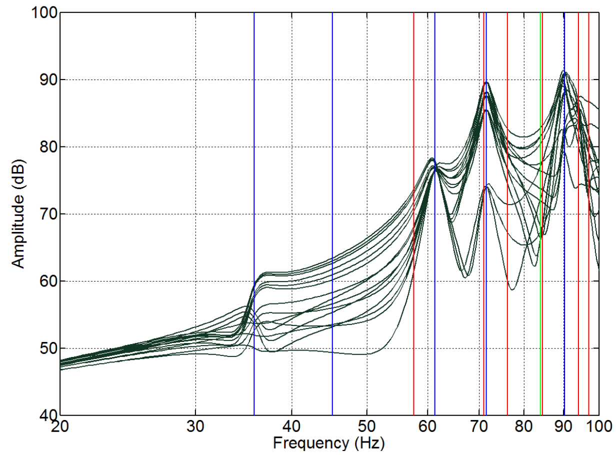

Figure 2.2: Simulated frequency response at 25 listening locations with a single subwoofer at the room floor center (blue lines = axial modes, red lines = tangential modes, green lines = oblique modes)

The disadvantage to nodal placement is that the subwoofer to room mode coupling in minimized, giving less acoustical output from the subwoofer than with corner placement.

Clearly, it is beneficial to have omnidirectional subwoofers near as many room boundaries as possible to take full advantage of the Waterhouse effect. With this in mind, it is often concluded that multiple subwoofer systems can provide superior results to single subwoofer systems [9 – 11].

Related research concerning this method has suggested that the optimal configuration places an omnidirectional subwoofer at each wall midpoint on the ground (Figure 2.3) [9 – 11]. This will offer similar benefits as a single source in the center of the room, but with additional output energy due to the proximity of the walls. While this variety of configuration results in minimal variation across a listening space, it can be highly inefficient, requiring very powerful subwoofers to produce acceptable sound pressure levels to match those of the main loudspeakers.

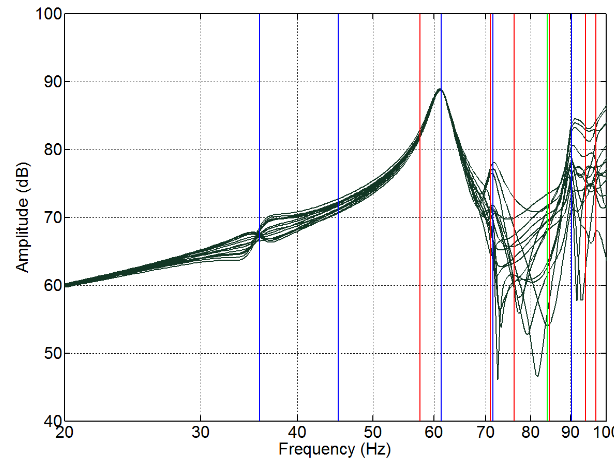

Figure 2.3: Simulated frequency response at 25 listening locations with a four subwoofers at the wall midpoints (blue lines = axial modes, red lines = tangential modes, green lines = oblique modes)

2.2. Single point equalization

A common form of room correction that is usually applied over the entire audio bandwidth is single point equalization. A measurement is taken at the target listening location and an inverse FIR filter is generated to give a flat response at the target listening location. Systems consisting of multiple subwoofers will have the same equalization filter applied to each subwoofer. While this technique can be effective above the Schroeder frequency, where the response across the listening area is not strongly affected by discrete room modes, it can provide poor results in the subwoofer operating range.

While the target listening location may exhibit a flat low-frequency response, other listening locations could have responses that are significantly worse than with the uncorrected system (Figure 2.4). This again is due to the subwoofer to listening location coupling, which differs greatly from point to point in a room. The single point correction targets only a single set of mode coupling factors which cannot be expected to have positive effects over an entire listening area (unless the room is anechoic).

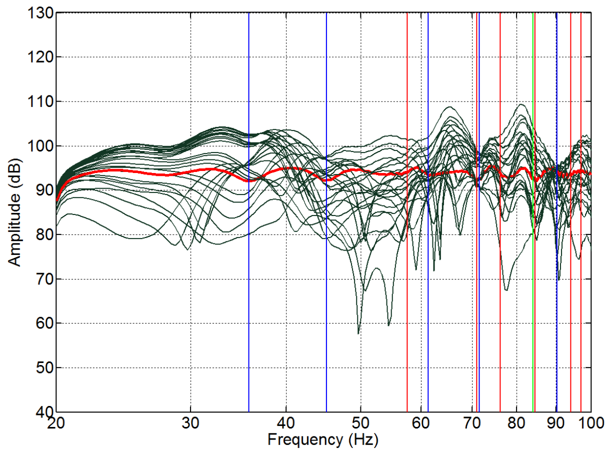

Figure 2.4: Single-point equalization results for configuration with unequalized response shown in Figure 2.1 (EQ point = bright red line)

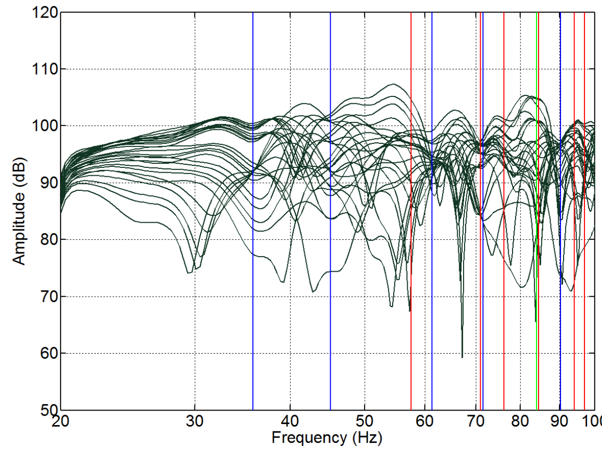

Figure 2.5: Multiple-point equalization (with a single equalizer) results for configuration with unequalized response shown in Figure 2.1

2.3. Multiple point equalization

Expanding upon the concept of single point equalization, multiple point equalization (assuming here a single equalizer applied to all subwoofers) takes measurements at several points within the listening area for room correction. The most basic method of this form of correction is to simply average the responses and generate an inverse filter based on this average (often with weighting applied). Again, the equalization is equally applied over all sources in a multiple subwoofer system. This windowing technique generally performs better than the single point method by giving a flatter average result, although spatial variations between listening locations inevitably remain unchanged (Figure 2.5).

Work has been carried out in this area with very positive results in some cases where clever weighting and grouping of measured responses (often using independent equalization for each source) gives a flat response over an entire set of target listening points with reduced spatial variation [12 – 16]. Some of these techniques do not entirely address the performance in the time domain (transient and waveform accuracy) as well as performance at non-targeted listening points.

2.4. Active absorption

An additional room correction method that has been the topic of investigations is active absorption [17]. Active absorption combines the principles of passive correction methods and single/multiple point equalization methods. This method generally operates with a set of one or two primary sources placed at one end of a room. In addition, an array of secondary sources is placed at the opposite end of the room, with each drive unit usually containing a measurement microphone to monitor the signals received from the primary sources. These secondary sources will then reproduce their measurements with reverse polarity in an attempt to suppress wall reflections, giving a traveling wave in the room as if it were anechoic.

These methods can require a large number of secondary drive units to effectively suppress reflections within the room, making them difficult and costly to implement. When properly calibrated, though, active absorption systems can create a virtual anechoic environment where all points (at a sufficient distance from the secondary units) will experience the same response both in the time and frequency domains.

3. GRADIENT LOUDSPEAKERS

All forms of room correction discussed to this point in the paper have been applied using omnidirectional subwoofers which are most common in commercially available subwoofer systems. It has been suggested that radiating low-frequency energy equally in all directions may be detrimental to the perceived performance of the subwoofer [18 – 21]. Focusing a subwoofer's polar pattern could avoid modal excitation over certain dimensions of a room without relying on acoustical cancellation to suppress the modes, resulting in a higher efficiency.

Low-frequency directivity is easily controlled using the principles presented by Olson concerning gradient loudspeakers [22]. Gradient loudspeakers utilize the principles of microphone polar pattern control, but in reverse. These methods (with the exception of the zero-order variety) call for two or more drive units within each source to achieve the desired directionality. A number of different configurations have been presented by Olson which provides a useful tool set for low-frequency polar pattern control.

3.1. Zero-order gradient sources

The zero-order gradient source is used as the building block for all higher order forms of gradient sources. It is a single drive unit which radiates energy equally in all directions (Figures 3.1 & 3.2).

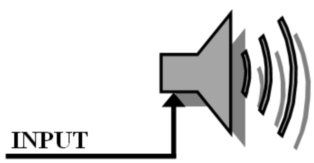

Figure 3.1: Zero-order gradient source configuration

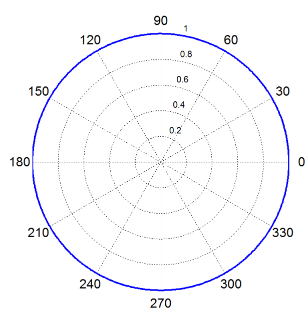

Figure 3.2: Zero-order gradient source polar pattern (horizontal plane)

3.2. First-order gradient sources (dipole)

The first higher-order variety of gradient sources is a combination of two zero-order sources with one source having reverse polarity (Figure 3.3). This configuration's polar pattern is highly dependent on the physical separation distance of the two sources (Equation 3.1). A separation distance of a quarter-wavelength of the target frequency results in a dipole pattern while a separation of a full wavelength gives a four-lobed polar pattern (Figure 3.4) [22]. Similar results can also be achieved using a single drive unit open baffle configuration.

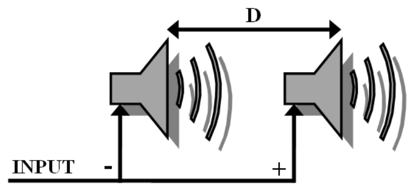

Figure 3.3: First-order gradient source configuration (dipole)

$$R_\theta = \sin\left(\frac{kD}{2}\cos\theta\right) \tag{3.1}$$

where R_θ is the system response at angle θ from perfectly on axis, k is the wave number and D is the source separation distance in meters.

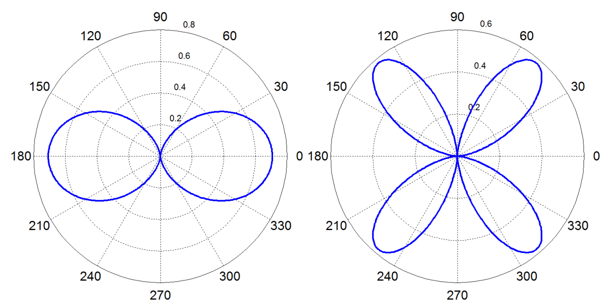

Figure 3.4: First-order gradient source (dipole) polar pattern (horizontal plane) with D = ¼ wavelength (left) and D = full wavelength (right)

3.3. First-order gradient sources (cardioid)

The first-order gradient source of the dipole variety can be adjusted to give cardioid-like polar patterns. This adjustment involves adding a delay to the second drive unit which directly corresponds to the driver separation distance (Figure 3.5).

Again, the polar pattern is highly sensitive to driver separation (Equation 3.2) as quarter wavelength separation will give a cardioid pattern while full wavelength separation gives a dipole pattern with 90° horizontal rotation (Figure 3.6) [22].

$$R_\theta = \sin\left(\frac{kD}{4} + \frac{kD}{4}\cos\theta\right) \tag{3.2}$$

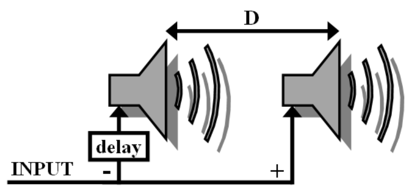

Figure 3.5: First-order gradient source configuration (cardioid)

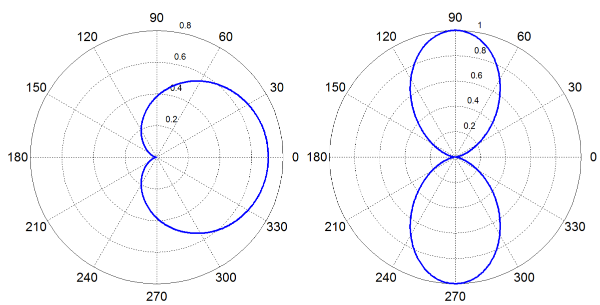

Figure 3.6: First-order gradient source (cardioid) polar pattern (horizontal plane) with D = ¼ wavelength (left) and D = full wavelength (right)

3.4. Second-order gradient sources

Second-order gradient sources can be formed by taking two first-order sources of the dipole variety and placing them together with a physical separation with the second first-order source being delayed by an amount directly corresponding to the separation distance (Figure 3.7).

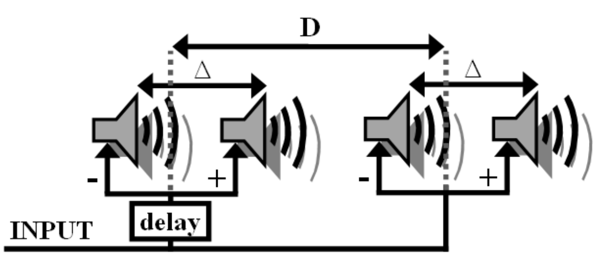

Figure 3.7: Second-order gradient source configuration

$$R_\theta = \sin\left(\frac{kD}{8}\cos\theta\right)\sin\left(\frac{kD}{4} + \frac{kD}{4}\cos\theta\right) \tag{3.3}$$

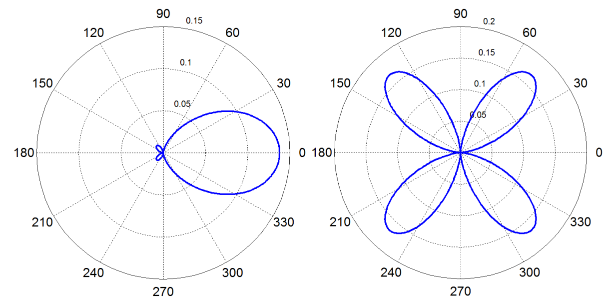

Figure 3.8: Second-order gradient source polar pattern (horizontal plane) with D = ¼ wavelength (left) and D = full wavelength (right)

Gradient sources of higher orders can be achieved by combining the presented configurations in a similar manner. It would be expected that as the order of the source increases, the polar pattern will become increasingly focused. It must be noted that as the gradient order increases, source efficiency decreases due to destructive interference between the drive units used to achieve the desired polar patterns [22]. However, the use of multiple drive units generalizes into the domain of phased-array systems where similar theory applies for both loudspeakers and microphones.

3.5. Low-frequency correction with gradient sources

Gradient sources can be very useful for low-frequency correction, if used correctly. Backman [20] has suggested in his work that a gradient source with a frequency-dependant polar pattern would be advantageous for room correction while also ensuring maximum system efficiency. Backman observes that below the lowest room mode, pressure variations with listener location become progressively smaller since no standing waves form within the room. In this frequency range, the subwoofer is simply pressurizing the room.

Backman therefore proposed that below the lowest mode the subwoofer should exhibit an omnidirectional pattern to ensure most energy is placed into the room for pressurization. However, above the lowest mode, the polar pattern can tighten so that less room modes are strongly excited [20].

This paper seeks to generalize this approach to frequency-dependant low-frequency polar pattern control and to adapt the concept to form a more holistic method of acoustic error correction. The proposed methodology is developed in the following sections.

4. CHAMELEON SUBWOOFER THEORY

A “chameleon” subwoofer array refers to a system consisting of multiple low-frequency components that can adjust itself to its surroundings to give equal low-frequency coverage within a defined listening area. The individual components of the system are arranged in clusters where each cluster contains a single omnidirectional component along with three orthogonal dipole components.

Each component within the system is driven by a dedicated signal which can be adjusted in amplitude and phase to achieve the desired results. This method is based on earlier work by Howe and Hawksford [23], where an approach similar to that used in Ambisonics was used to provide low-frequency equalization in a listening area. This system requires the listening area to be surrounded by sources (around eight) to achieve proper results.

The chameleon subwoofer array system does not require a specific source layout for proper performance. The sources will adapt themselves to their environment regardless of placement, hence the descriptor, “chameleon.”

4.1. Mathematical implementation

In the current incarnation, the core process involved in this class of room correction is based on straightforward matrix algebra. Impulse response measurements are taken at a number of points determined by the total number of source components in the subwoofer array (Figure 4.1). Next, the target response at each of the measurement points is defined in the frequency domain. The ability to define the complex frequency response at each target point allows for propagation delay to be taken into account which can help relax the requirements placed on the correction system.

The measured impulse responses and target responses are then used to calculate the ideal filter coefficients for each source component (Equations 4.1 & 4.2). This effectively creates a frequency-dependent polar pattern of the source clusters which will provide the specified responses at the listening locations.

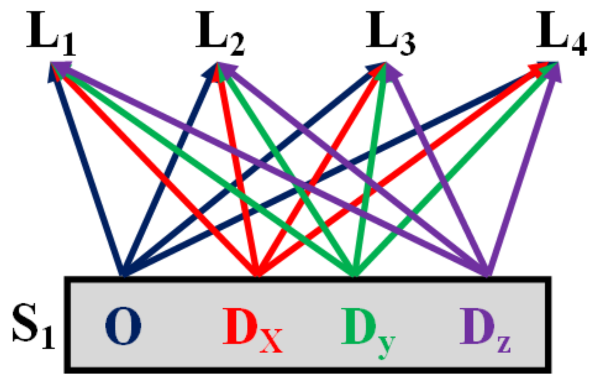

Figure 4.1: Impulse response measurement with one source cluster, S_1 (O = omni, D_X = X-dimension dipole, D_Y = Y-dimension dipole, D_Z = Z-dimension dipole, L_N = listening location N)

Equations 4.1 and 4.2 show the matrix composition for the system shown in Figure 4.1.

$$\underbrace{\begin{bmatrix} Y_1 \\ Y_2 \\ Y_3 \\ Y_4 \end{bmatrix}}_{Y} = \underbrace{\begin{bmatrix} X_{1,1} & X_{1,2} & X_{1,3} & X_{1,4} \\ X_{2,1} & X_{2,2} & X_{2,3} & X_{2,4} \\ X_{3,1} & X_{3,2} & X_{3,3} & X_{3,4} \\ X_{4,1} & X_{4,2} & X_{4,3} & X_{4,4} \end{bmatrix}}_{X} \underbrace{\begin{bmatrix} H_1 \\ H_2 \\ H_3 \\ H_4 \end{bmatrix}}_{H} \qquad (4.1)$$

$$H = X^{-1} Y \qquad (4.2)$$

where at listening location L and for source component S, Y_L is the target response, $X_{L,S}$ is the measured frequency response and H_S is the correction filter coefficients to achieve the target frequency response.

This mathematical description was validated by using simulated measurements from the FDTD toolbox while setting all target responses to flat (1 Pa, i.e. SPL ~93.98 dB with zero phase). The corresponding uncorrected and corrected frequency responses are displayed in Figures 4.2 and 4.3, respectively.

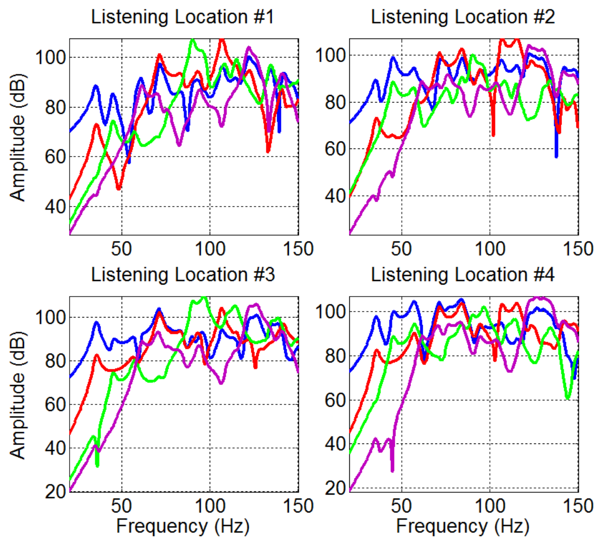

Figure 4.2: Uncorrected frequency response measurements for the system shown in Figure 4.1

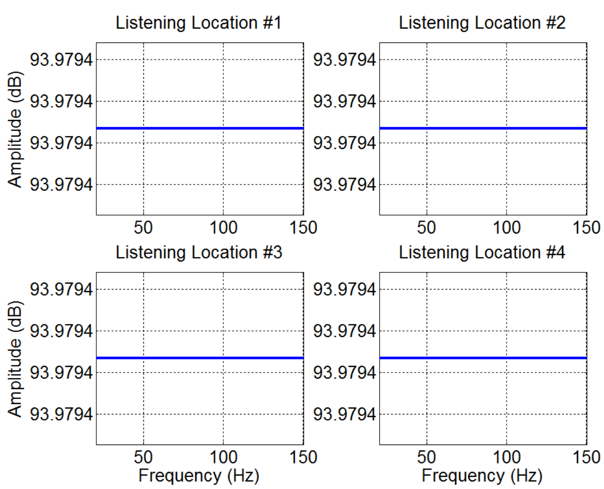

Figure 4.3: Corrected frequency response for the system shown in Figure 4.1 (93.9794 dB ≈ 1Pa)

The results from the direct matrix inversion of the simulated complex frequency responses give precisely flat responses at all four listening locations. However, while mathematically the system may perform perfectly there are a number of issues that need to be addressed in order to enable this system to perform in a real world environment. The principal issues to be addressed concern the amplitude of the correction coefficients as well as the overall stability of the room correction system.

4.2. Target frequency response considerations

It is essential to choose an appropriate set of target frequency responses when using a chameleon subwoofer array for room correction. The choice of idealized responses is unrealistic considering the performance capabilities of most conventional subwoofers (Figure 4.4). A subwoofer cannot be expected to reproduce a flat response with a frequency range extending to 20 Hz and below. Figure 4.4 highlights this constraint where the correction coefficients reveal a substantial increase in their magnitude below 50 Hz, well outside the safe operating range of a subwoofer.

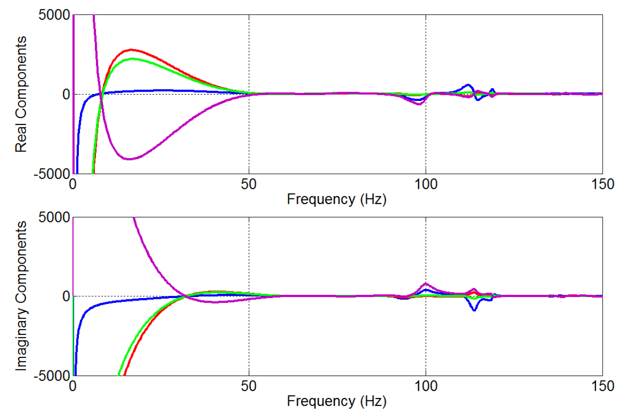

Figure 4.4: Correction coefficients for a flat target response (linear scale)

Also, it is probable that a listening location within the set of target points will be placed at or near a node of a room mode. This will result in unrealistic energy required to raise the response at the listening point to the desired amplitude. A target response must be chosen that will not place unrealistic requirements on the subwoofer system which could cause high distortion and possible system damage.

In the early stages of development the target response at each listening location was set to match the average response over all points. This should avoid unrealistically high correction filter coefficients, ensuring the system is kept within its safe operating range (Figure 4.5). Additional correction issues exist below the lowest room mode which will be addressed in the forthcoming sections.

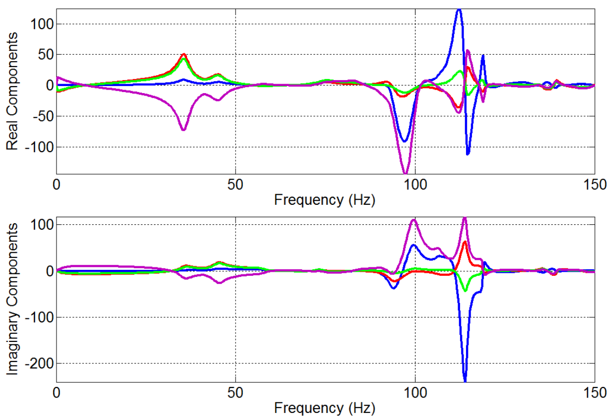

Figure 4.5: Correction coefficients for a target response corresponding to the average measured response of the target listening locations (linear scale)

4.3. Listening location considerations

Given that this system is highly dependent on the contributions of wall reflections to achieve the target responses, it is essential to ensure the target listening area is located at a sufficient distance from the source clusters to avoid the sources overpowering the response. The suggested minimum distance is approximately one meter.

In addition to minimum listening distance, the listening area which receives maximum correction benefits is enclosed by a line connecting the outermost listening points. If nine target points are used in a square 3 x 3 grid configuration with one meter separation, for example, the target listening area would be a 2 x 2 m square. Points outside this enclosed space are not guaranteed to benefit from the correction procedure.

4.4. Number and spacing of target points versus frequency

The number and spacing of target listening points used for room correction play an important role in determining the stability of the system. At very low frequencies (below the lowest room mode) most points within the room will have similar responses where room correction may not be necessary. Attempting to correct over a number of points at these frequencies could require unrealistically large correction coefficients.

Similarly, as frequency enters the discrete modal range (first room mode to the Schroeder frequency) spatial variation increases between points in the room. As the frequency rises it is necessary to have more measurement points to avoid any spatial aliasing, requiring the measurement points to be separated by less than half the shortest wavelength in the correction range. If this requirement is not met, listening points in between the target points will not benefit from room correction.

Above the Schroeder frequency the sound field is considered to be diffuse. This is generally outside the standard operating range of a subwoofer and would not be expected to benefit from correction as discrete room modes are not an issue in this range where differences in response between listening points are not strongly perceived by the human ear. Also, the range would require an unreasonably large number of measurement points to cover the entire correction range to avoid spatial aliasing.

Assuming the correction frequency range will be located within the discrete room mode range, target listening points can be arranged to give proper correction over the entire frequency band. At the lower boundary of this band where omnidirectional sources are more efficient than dipole sources, as seen in Figure 4.2, the dipole sources can effectively be turned off with correction being applied through the omnidirectional sources. This method will result in only a quarter of the target listening locations being used, reducing redundancy from locations with similar responses, thus enhancing system stability.

At higher frequencies within the correction range the dipole sources can be activated, along with the remaining target listening locations, so that this range can be corrected without the risk of spatial aliasing.

5. SIMULATION RESULTS

All important considerations when implementing a stable and realizable correction method, as mentioned above, can be highlighted by applying correction within the FDTD simulation toolbox. Correction calculations will be based on MLS measurements, again from within the FDTD toolbox. Simulation results should support the claims in the previous sections concerning correction stability and feasibility.

5.1. Filter stability considerations

In many cases the correction coefficient calculation procedure will produce a non-minimum phase response, resulting in an unstable filter. This is evident when examining the unprocessed impulse responses from the correction routine (Figure 5.1). There is clear ringing in the response that is due to a sharp spike in correction coefficient amplitude for a frequency in the upper range of the correction band (Figure 5.2).

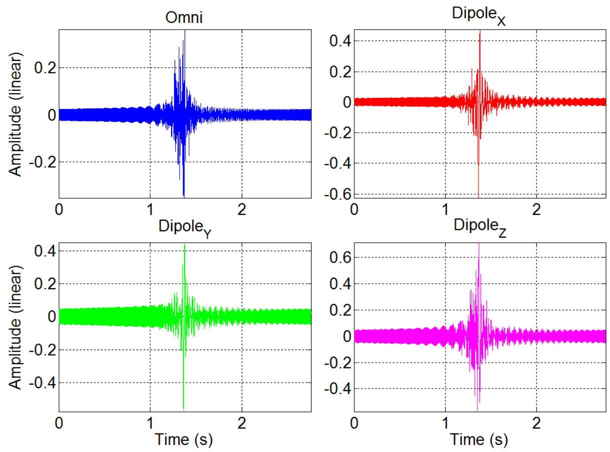

Figure 5.1: Correction impulse responses with clear ringing (correction up to 250 Hz)

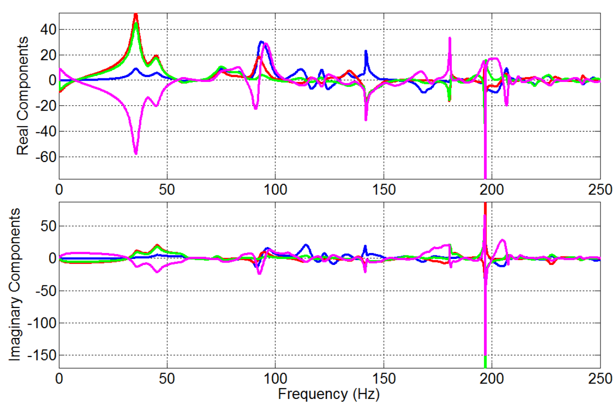

Figure 5.2: Correction coefficients first sharp spike around 170 Hz (correction up to 250 Hz)

The upper frequency limit for correction can be lowered to help suppress this ringing as it is likely caused by a frequency occupying the transition band between the discrete modal and diffuse regions. Reducing the high frequency limit reduces the instability in the response, but not entirely (Figure 5.3).

To force the correction impulse response to decay to zero a window function is applied to suppress residual ringing (Figure 5.4), although this does result in a minor loss of correction accuracy. With the elimination of persistent ringing in the correction impulse responses, room correction can be applied to a source signal without introducing significant pre or post-echoes.

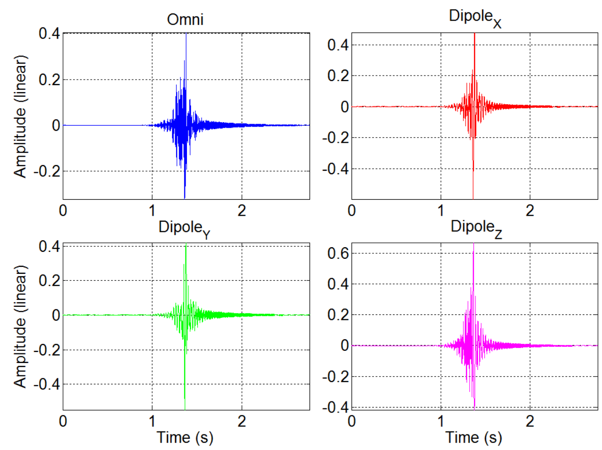

Figure 5.3: Correction impulse responses with reduced ringing (correction up to 150 Hz)

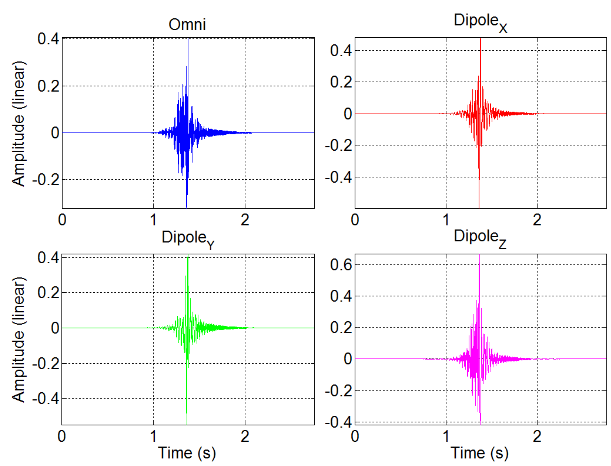

Figure 5.4: Correction impulse responses with windowing applied (correction up to 150 Hz)

5.2. Target listening location behavior

First, it is necessary to measure the corrected response at the target listening locations. Ideally, these points should show identical responses, both in the time and frequency domains. However, because of correction impulse response windowing minor errors in the measured responses are anticipated.

A 5 m x 4 m x 3 m room was set up in the FDTD simulation toolbox with 20 cm grid point separation and 10% absorption on all surfaces. A single source cluster (one omnidirectional and three dipole components) was placed at (1 m, 1 m, 1 m) with four target listening locations at a height of 1.8 m (Figure 5.5).

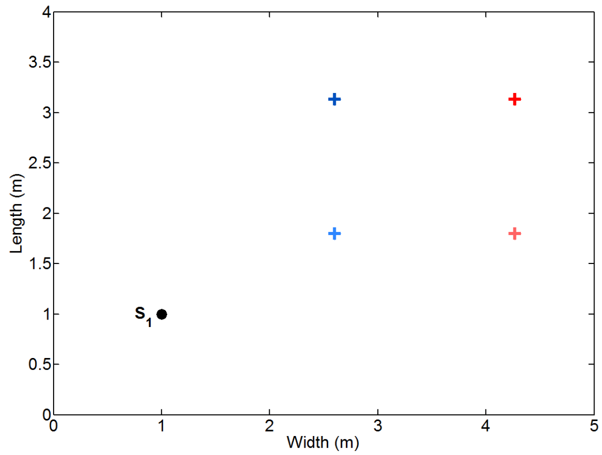

Figure 5.5: Room setup for correction method testing with four target points (S_1 = center of source cluster)

A 60 Hz sinusoid signal with five second duration was used for the first test case. The uncorrected system was simulated (Figure 5.6) and then compared to the corrected system using energy time domain representation computed using the Hilbert transform [24] (Figure 5.7).

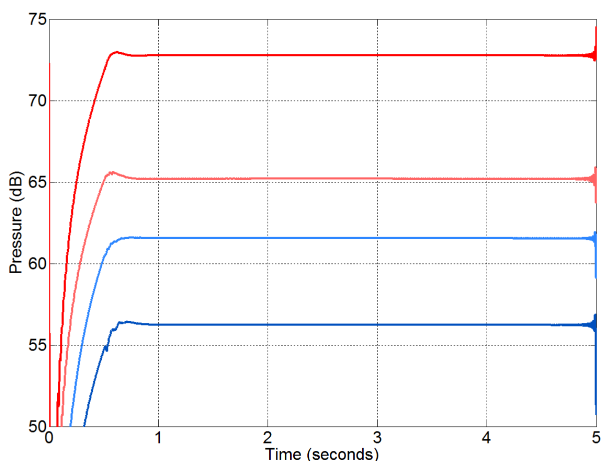

Figure 5.6: Uncorrected system time domain energy envelope for 60 Hz sinusoid

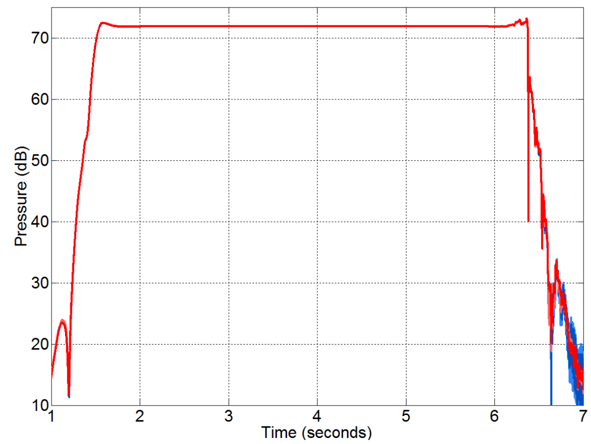

Figure 5.7: Corrected system time domain energy envelope for 60 Hz sinusoid

The anticipated correction results are shown in Figure 5.7. There is a lead-in and lead-out period due to convolution with the correction impulse responses (Figure 5.4), but the entire five second sinusoidal energy envelope is clearly visible with all target points falling within one tenth of a decibel. This is a large improvement from the uncorrected system where measured amplitudes deviated by upwards of 15 dB.

While the correction exhibits good steady state results, the system must also be examined when dealing with transient signals. Following the doctrine of Linkwitz [19], an 80 Hz tone burst test sequence was utilized consisting of five repeated segments of ten sinusoidal cycles. The uncorrected measurements (Figure 5.8) were again compared to the corrected measurements (Figure 5.9).

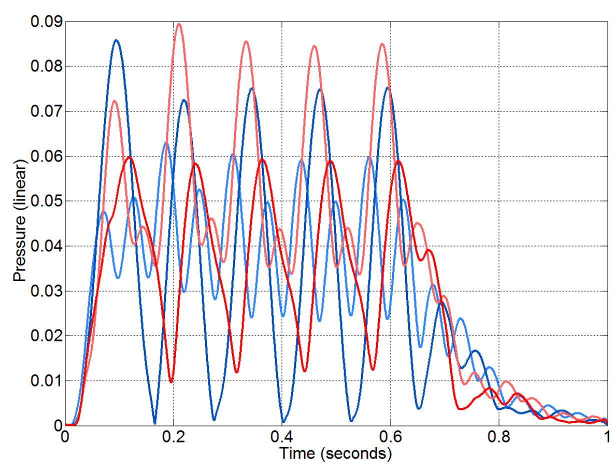

Figure 5.8: Uncorrected system time domain energy envelope for 80 Hz tone bursts (linear scale)

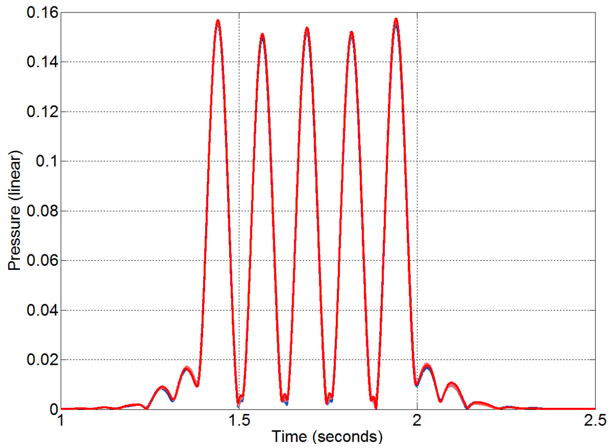

Figure 5.9: Corrected system time domain energy envelope for 80 Hz tone bursts (linear scale)

Again source-signal correction performed as expected revealing minimal error between the target points compared to the non-corrected results which revealed gross corruption resulting from time domain dispersion.

5.3. Non-target listening location behavior

While target points clearly benefit from room correction using a chameleon subwoofer array, it is desirable that points in between these target points benefit equally. This is where the issues concerning upper and lower frequency stability limits come into effect.

5.3.1. Four target points

A test was designed to simulate a walking path through a listening area where the path intersects a target point approximately every five steps. This was first set up using the four-point configuration used in the previous section (Figure 5.10). This configuration was tested with 40, 60 and 80 Hz sinusoidal and tone bursts signals.

The target points are separated by 1.4 and 1.8 m in the x- and y-dimensions, respectively. Taking the largest of these distances gives an expected upper limit of correction stability between target points of around 95 Hz (based on half wavelength spacing).

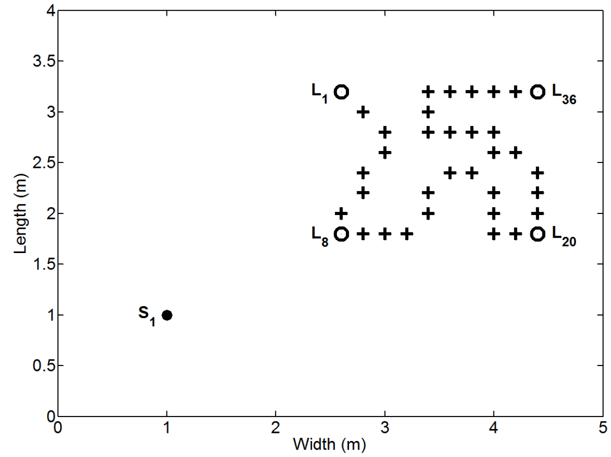

Figure 5.10: Walking test setup with 4 target points (circles) and 32 non-target points (crosses)

The correction results with the walking path test for 60 Hz (Figure 5.11) show that all points along the path have benefited equally from correction with only minimal deviations. However, when correction is applied at 80 Hz, large deviations are now visible between listening points (Figure 5.12). This suggests that the upper frequency limit for correction based on target point spacing needs to be a more conservative calculation such as one-third wavelength, as opposed to one-half wavelength spacing. The adjusted limit would equal 64 Hz for this four-point configuration.

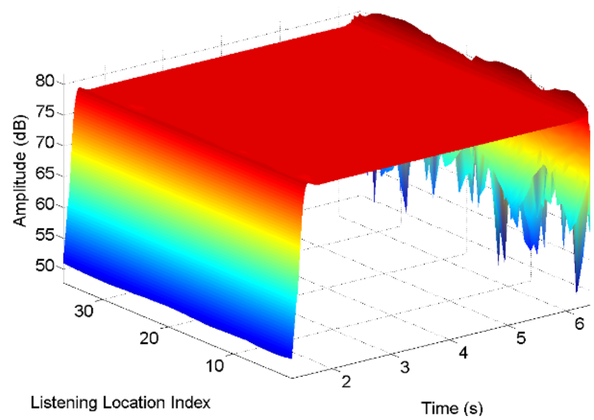

Figure 5.11: Walking test results for 60 Hz sinusoid (target points = 1, 8, 20 and 36)

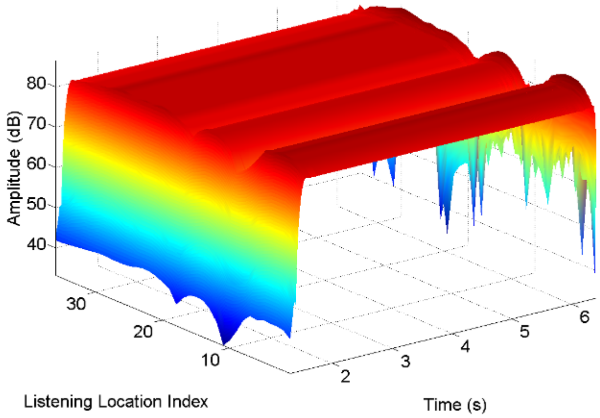

Figure 5.12: Walking test results for 80 Hz sinusoid (target points = 1, 8, 20 and 36)

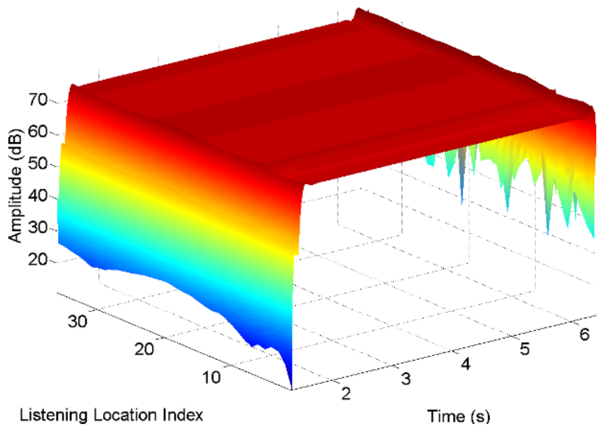

Figure 5.13: Walking test results for 40 Hz sinusoid (target points = 1, 8, 20 and 36)

At 40 Hz there is a slight boost in amplitude at points surrounding the nearest target point to the source cluster (listening location indices 6 – 10). Aside from this minor deviation in response, correction performs as expected at 40 Hz (Figure 5.13).

Tone burst testing presents similar results to the sinusoidal testing where there is a slight amplitude boost for points 6 – 10 at 40 Hz and unequal correction benefits along the walking path above 80 Hz (Figure 5.14). Again, 60 Hz gives good results at all points along the path (Figure 5.15).

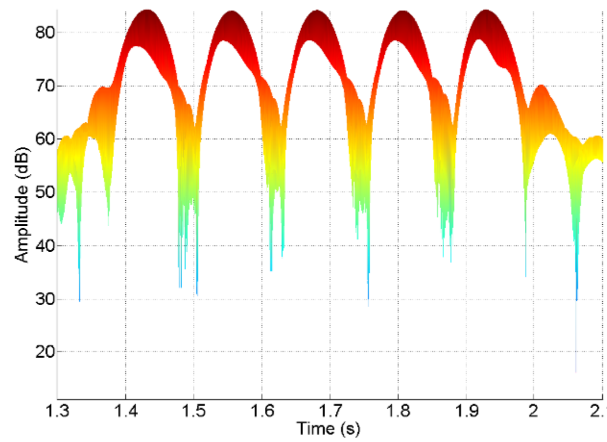

Figure 5.14: Walking test results for 80 Hz tone burst (XZ plane view) showing waveform degradation across the walking path

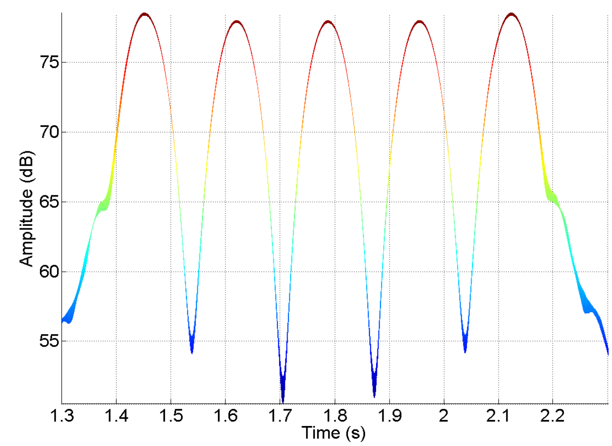

Figure 5.15: Walking test results for 60 Hz tone burst (XZ plane view) showing minimal waveform degradation across the walking path

5.3.2. Eight target points

It is necessary to have closer spaced target points to give more accurate correction results at higher frequencies. Given the findings that correction behavior falls off at frequencies above one-third wavelength spacing (~64 Hz in the four-point case) target point spacing should be reduced to around one meter to give accuracy up to 120 Hz which is approximately the upper boundary of a standard subwoofer's operating range.

More target points need to be added to ensure correction over a large listening area at these higher frequencies. A test was set up with eight target points separated by one meter in both the x- and y-directions. This spacing corresponds to one-third the wavelength of 114 Hz; very close to the required upper correction limit of 120 Hz. To achieve correction at eight points, two source clusters must be used, giving a total of eight source components within the chameleon array (Figure 5.16).

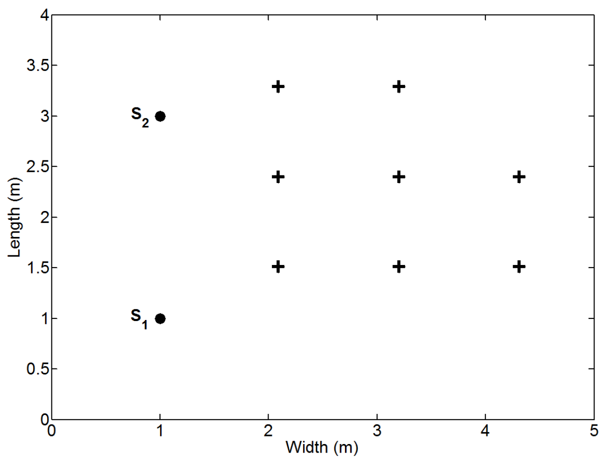

Figure 5.16: Room setup for correction method testing with eight target points

The tighter spacing of target points should raise the upper frequency limit for accurate correction, although performance below the lowest room mode (< 34 Hz for this example) will decrease in efficiency and accuracy due to minimal variation in response between target points, requiring the system to output large amounts of energy to adjust these minor differences. Also, the front row of target points is situated very close to the sources which should be reflected in the results, showing higher amplitudes near these target points.

The initial correction coefficient calculation for the eight-point configuration shows unreasonably high amplitudes near 60 Hz and below (Figure 5.17). The farthest spaced target points over this dimension are two meters from one another, which is approximately one-third the wavelength of 60 Hz. With this in mind, it can be concluded that variation between points below 60 Hz becomes minimal while the performance of the dipole sources will be highly inefficient, as discussed earlier.

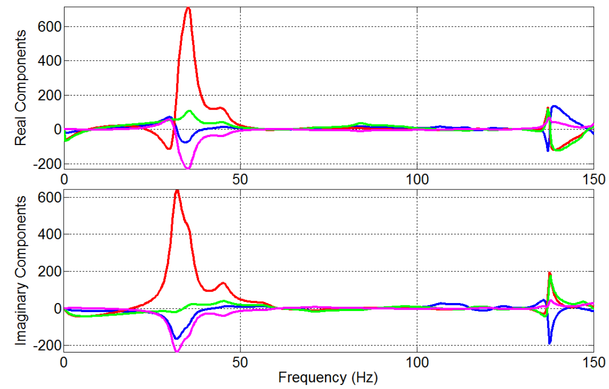

Figure 5.17: Correction coefficients (Source 1) for correction below 150 Hz (eight target points)

Given the correction problems below 60 Hz with these tightly spaced target points, the correction coefficients were recalculated with a restricted correction range from 60 Hz to 130 Hz (Figure 5.18). The new coefficients give realistic impulse responses that can provide accurate and efficient correction in the defined frequency range (Figure 5.19).

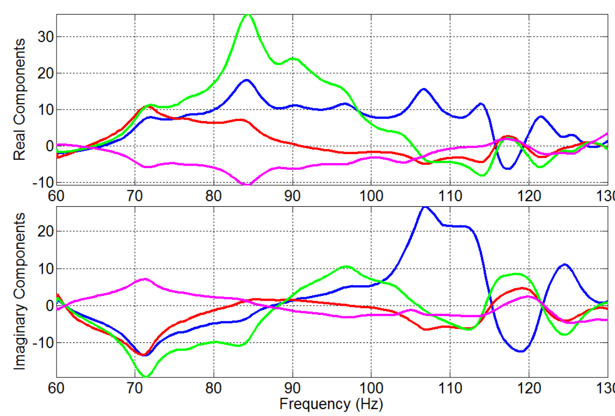

Figure 5.18: Correction coefficients (Source 1) for correction from 60 – 130 Hz (eight target points)

Having achieved realistic correction coefficients, testing was carried out using 80 and 120 Hz sinusoidal and tone burst signals. Again, a walking path between target points was simulated to analyze the response over the entire listening area (Figure 5.20).

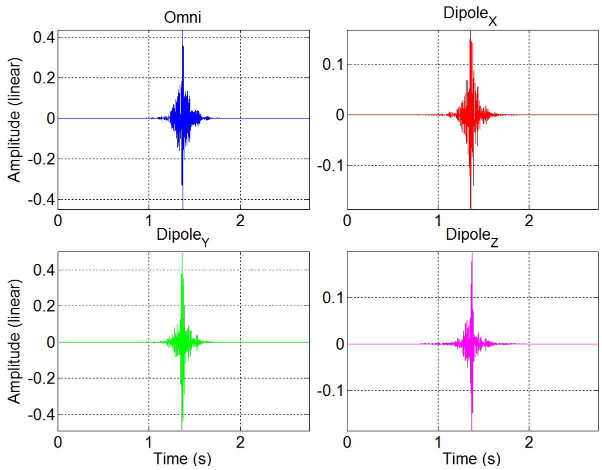

Figure 5.19: Correction impulse responses (Source 1) for correction from 60 – 130 Hz (eight target points)

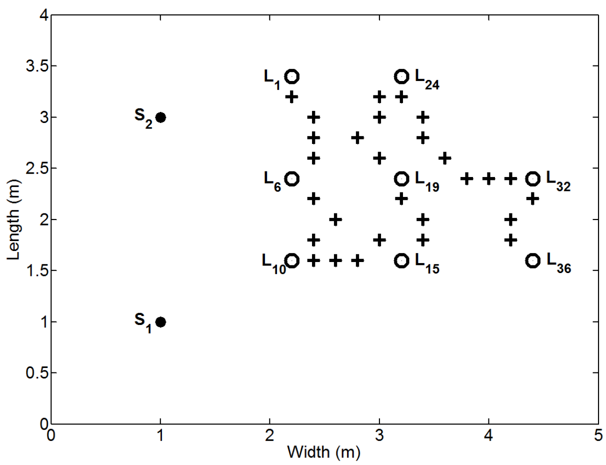

Figure 5.20: Walking test setup with 8 target points (circles) and 28 non-target points (crosses)

Results at 80 Hz show a significant improvement from the four-point configuration, where the only noticeable variation in response across the walking path occurs at the points closest to the sources (Figure 5.21). As with the 80 Hz signal in the four-point configuration (Figure 5.12), 120 Hz in the eight-point configuration shows a slightly decreased benefit in correction for non-target points (Figure 5.22).

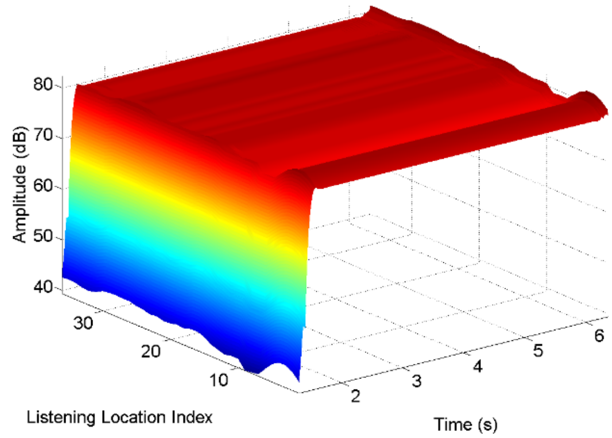

Figure 5.21: Walking test results for 80 Hz sinusoid (target points = 1, 6, 10, 15, 19, 24, 32 and 36)

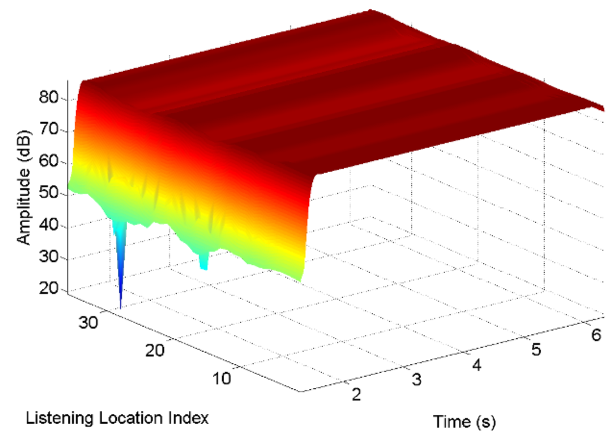

Figure 5.22: Walking test results for 120 Hz sinusoid (target points = 1, 6, 10, 15, 19, 24, 32 and 36)

Tone burst testing shows close agreement to the sinusoidal signal testing where at 80 Hz the only major variations occurred at the points closest to the sources, while all other points experience minimal waveform degradation (Figure 5.23). 120 Hz tone burst testing shows similar behavior to the 80 Hz testing with the four-point configuration (Figure 5.14) where there is noticeable waveform degradation across the walking path (Figure 5.24).

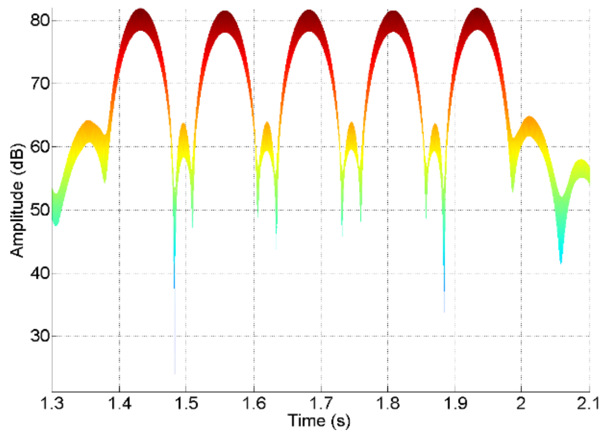

Figure 5.23: Walking test results for 80 Hz tone burst (XZ plane view) showing minimal waveform degradation across the walking path

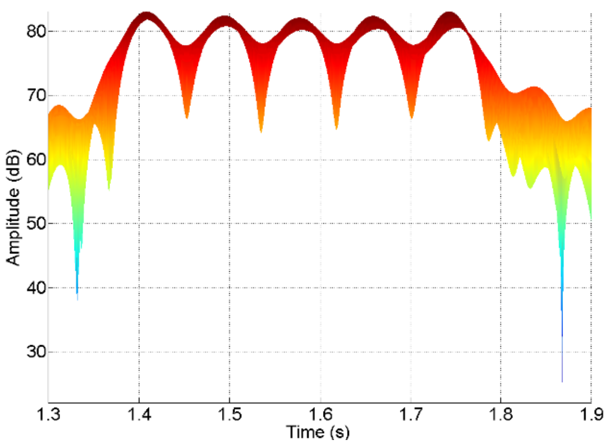

Figure 5.24: Walking test results for 120 Hz tone burst (XZ plane view) showing moderate waveform degradation across the walking path

Again, there appears to be an upper limit of correction accuracy which is related to the target point spacing. The eight-point configuration's one meter spacing corresponds to a one-third wavelength upper limit of 114 Hz. This would explain the error shown in the results with the 120 Hz signals.

A pattern which has emerged is that the bandwidth of correction is defined by the point-to-point spacing and the width of the listening area. The lower frequency limit can be approximated as the frequency with one-third wavelength matching the width of the listening area (Equation 5.1) while the upper limit is the frequency with one-third wavelength matching the target point separation (Equation 5.2).

$$f_{LOW} = {}^{c}\!/_{6D} \tag{5.1}$$

$$f_{HIGH} = {}^{c}\!/_{3D} \tag{5.2}$$

where c is the speed of sound in air and D is the separation distance between target points, in meters. A four-point configuration gives a frequency range of 32 – 64 Hz while an eight-point configuration gives a range of 57 – 114 Hz. While this chameleon subwoofer array can easily be expanded using even more clusters and target points, it is not necessary when focusing on a subwoofer operating range which does not exceed 120 Hz (unless dealing with a very large listening area).

6. COMPREHENSIVE CHAMELEON SUBWOOFER ARRAY SYSTEM

The evidence showing configurations of chameleon subwoofer arrays give proper listening area correction over a limited bandwidth based on the listening area layout suggests that a system can be constructed containing multiple operating bands, giving the desired performance over the entire subwoofer operating range (generally up to 120 Hz).

The FDTD simulation toolbox was set to model a three-dimensional rectangular room of dimensions 7 m x 5 m x 3 m with 20 cm grid spacing and wall absorption set to 10%. The target listening area was defined as a 3 m x 3 m area with 2 m height.

To give adequate correction at the lower limit of the subwoofer range, four target points with two meter spacing were placed in the listening area. To ensure proper correction for the upper frequency range, the gaps between the four existing target points were filled to give a 4 x 4 grid of target points with one meter spacing. The sixteen total target points require a system of four source clusters.

Based on extensive experimentation with various source layouts, it was found that when configuring a system with multiple source clusters it is beneficial to use the methods discussed earlier in the paper concerning room mode correction through careful source placement. While the correction method will work with any arbitrary source cluster placement, the system will operate more efficiently with symmetrical layouts which effectively suppress room modes simply due to their placement.

The four original target points will operate using the omnidirectional components of the clusters with the dipole components switched off. This is due to dipole source inefficiency at very low frequencies. The expected frequency range this system band will cover is 29 – 57 Hz (from Equations 5.1 & 5.2).

The upper frequency range will utilize all target points and source components. The one meter spacing of the target points gives a correction range of 57 – 114 Hz. The two operating bands match perfectly, giving an expected room correction range of 29 – 114 Hz, which is nearly the entire subwoofer range. The simulation layout for this test is shown in Figure 6.1. The correction coefficients for both bands with their corresponding impulse responses are shown in Figures 6.2 – 6.5 (Only one set of coefficients/impulse responses are shown for the high frequency band).

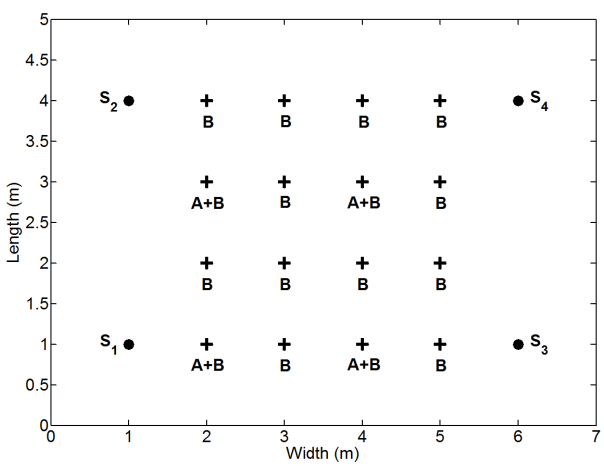

Figure 6.1: Layout for comprehensive system (A = low band target point, B = high band target point)

In order to handle the two bands of correction appropriately, a simple complementary linear phase FIR crossover network was created. The crossover point was set to 57 Hz with the low band output of the crossover corrected with the low band impulse responses and the high band output of the crossover corrected with the high band impulse responses. After correction was applied, the signals were summed into one signal for each source component in the chameleon subwoofer array.

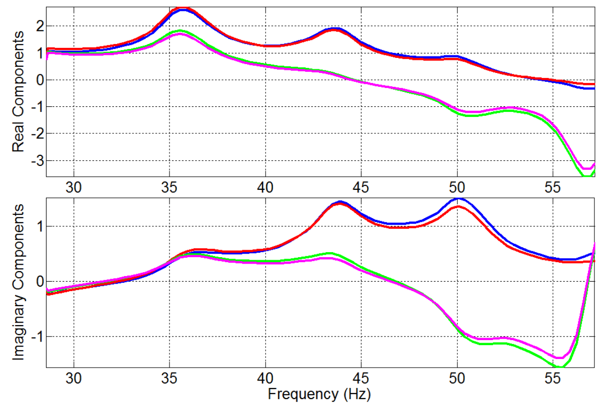

Figure 6.2: Low band correction coefficients for correction from 29 – 57 Hz (four target points)

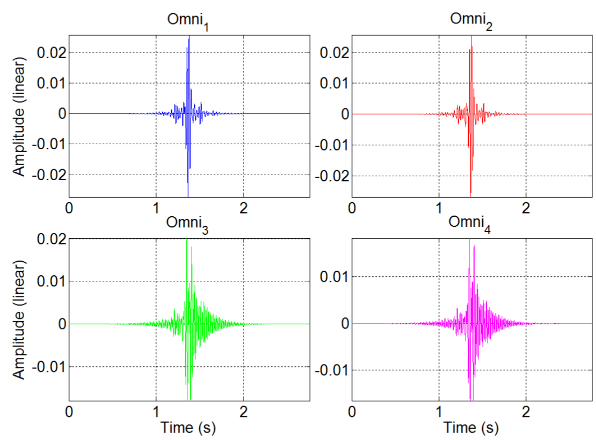

Figure 6.3: Low band correction impulse responses for correction from 29 – 57 Hz (four target points)

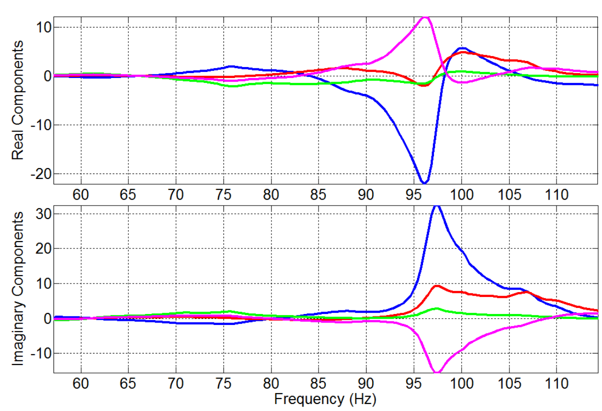

Figure 6.4: High band correction coefficients (Source 3) for correction from 57 – 114 Hz (sixteen target points)

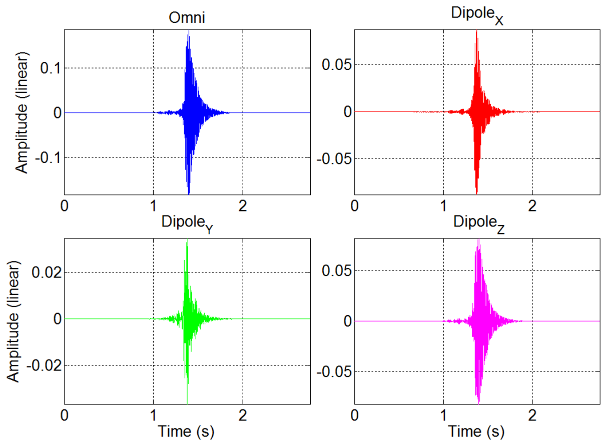

Figure 6.5: High band correction impulse responses (Source 3) for correction from 57 – 114 Hz (sixteen target points)

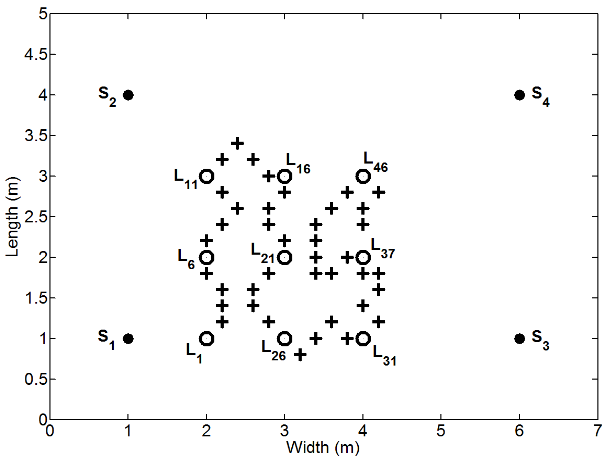

Figure 6.6: Walking test setup for comprehensive room correction system (target points = circles, non-target points = crosses)

As in previous tests, a walking path was simulated across the target listening area to best demonstrate the effectiveness of the room correction method at both target and non-target points (Figure 6.6).

The room correction was first judged by directly measuring the impulse response at each point along the walking path and then calculating the corresponding frequency responses. The responses with no correction (Figure 6.7) can be compared to those with the room correction applied (Figure 6.8).

While the uncorrected system shows a 4.53 dB average spatial variance across the walking path, the corrected system shows a much improved 0.95 dB average spatial

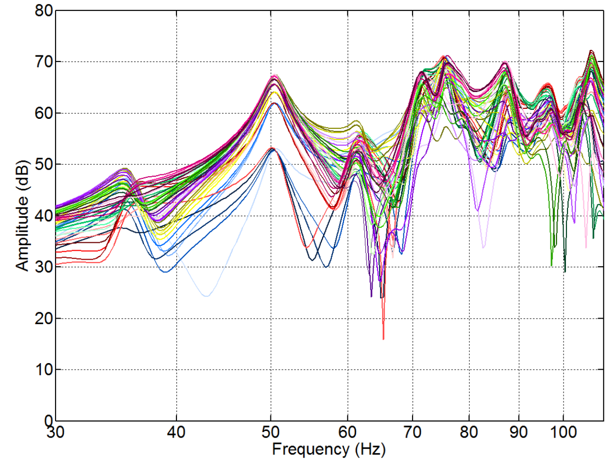

Figure 6.7: Uncorrected frequency response for all points along walking path

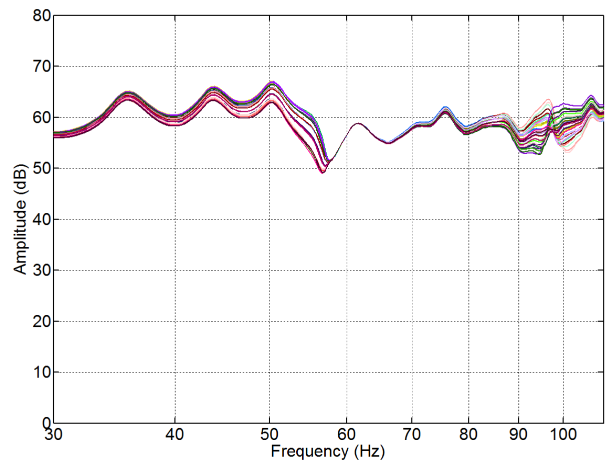

Figure 6.8: Corrected frequency response for all points along walking path

variance. This corresponds to a 79% decrease in variation between points within the listening area.

While there are clear benefits to correction in the frequency domain, time domain benefits must also be examined with attention given to waveform integrity across the listening area. The time domain benefits of the room correction can be analyzed with tone burst source signals. Five frequencies were chosen at 30, 50, 70, 90 and 100 Hz to be tested both with and without room correction and then compared (Figures 6.9 – 6.18).

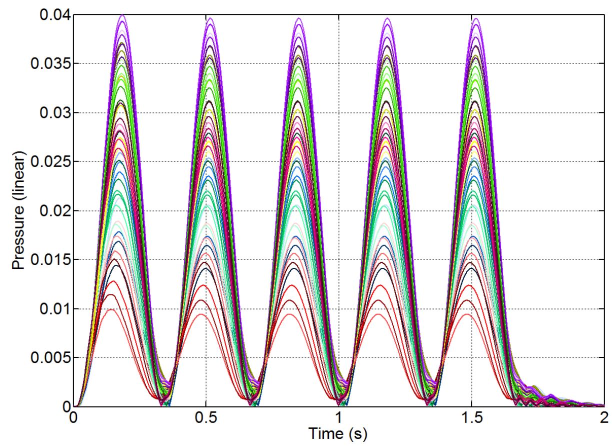

Figure 6.9: Uncorrected measured waveforms for 30 Hz tone burst testing (linear pressure scale)

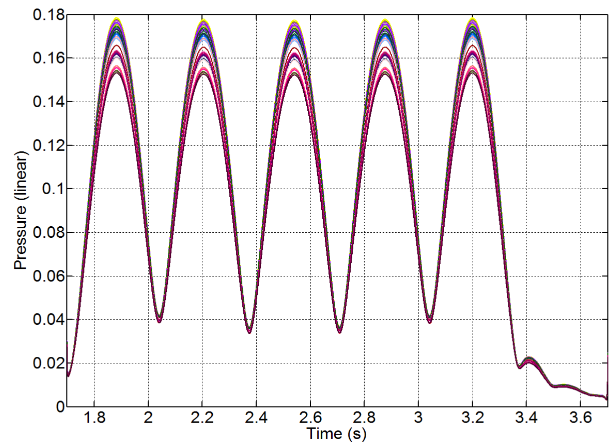

Figure 6.10: Corrected measured waveforms for 30 Hz tone burst testing (linear pressure scale)

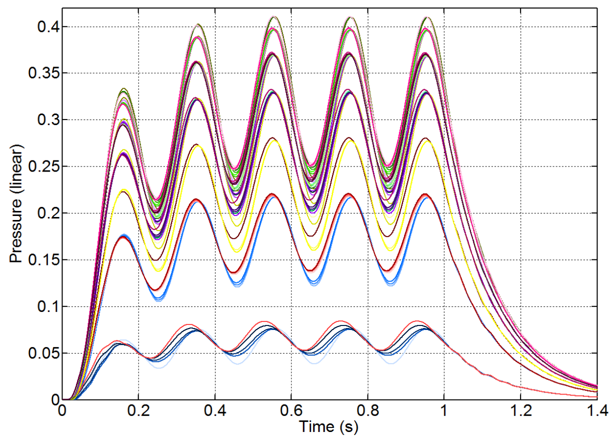

Figure 6.11: Uncorrected measured waveforms for 50 Hz tone burst testing (linear pressure scale)

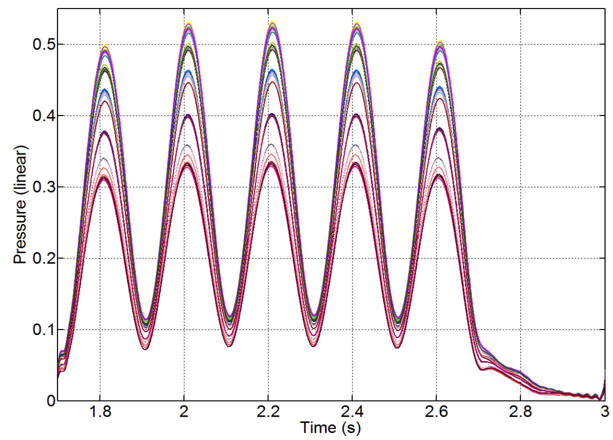

Figure 6.12: Corrected measured waveforms for 50 Hz tone burst testing (linear pressure scale)

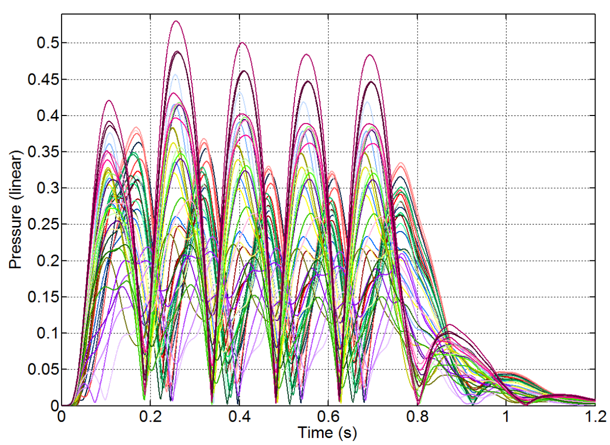

Figure 6.13: Uncorrected measured waveforms for 70 Hz tone burst testing (linear pressure scale)

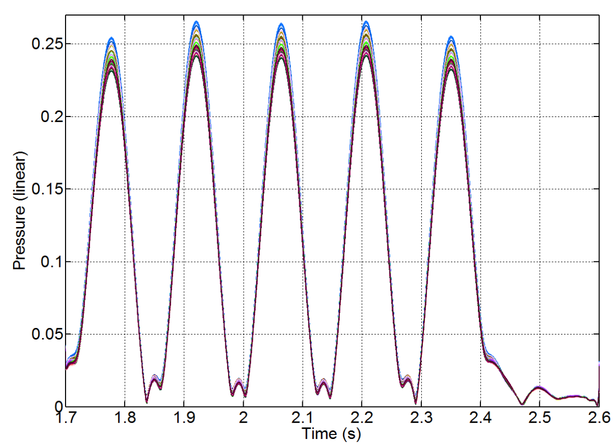

Figure 6.14: Corrected measured waveforms for 70 Hz tone burst testing (linear pressure scale)

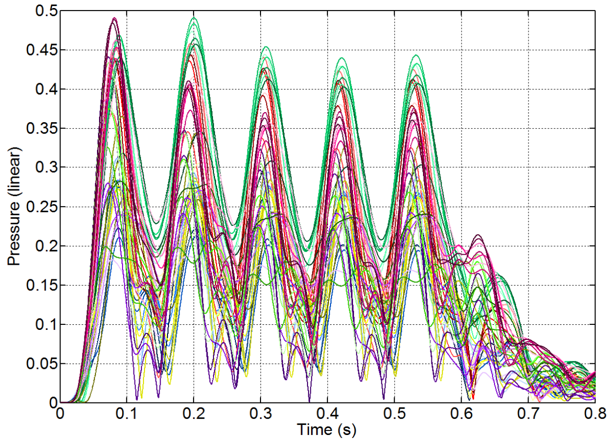

Figure 6.15: Uncorrected measured waveforms for 90 Hz tone burst testing (linear pressure scale)

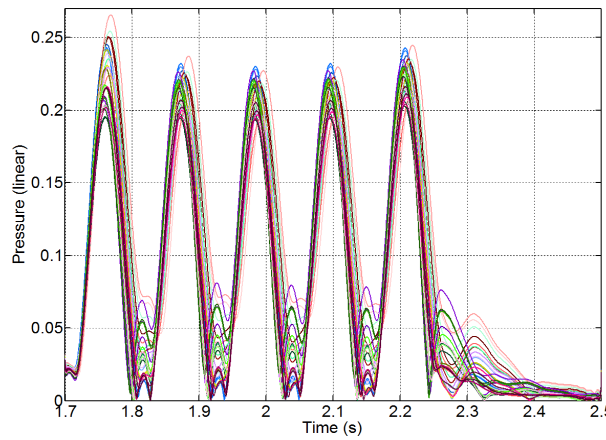

Figure 6.16: Corrected measured waveforms for 90 Hz tone burst testing (linear pressure scale)

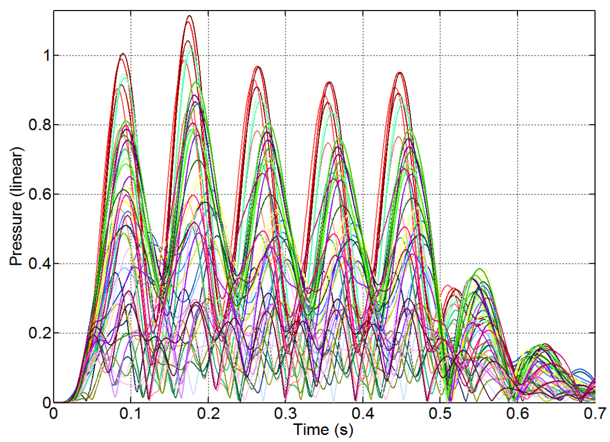

Figure 6.17: Uncorrected measured waveforms for 110 Hz tone burst testing (linear pressure scale)

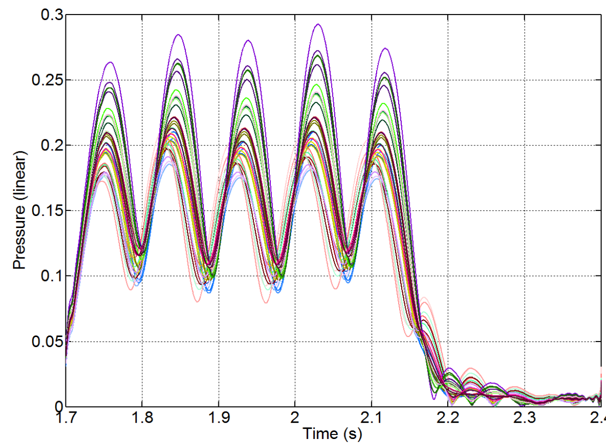

Figure 6.18: Corrected measured waveforms for 110 Hz tone burst testing (linear pressure scale)

The 30, 50, 70, 90 and 100 Hz signals show reduced waveform variation of 99.9%, 51.6%, 99.9%, 83.8% and 99.5%, respectively, which demonstrates the time domain accuracy of using chameleon subwoofer arrays for low frequency for room correction.

Uncorrected tone bursts above 70 Hz show very smeared waveforms indicating poor transient behavior of the system. After correction, though, time smearing has been minimized giving a significantly improved transient behavior. Reasoning for reduced performance at 50 Hz could be the proximity of the crossover point (57 Hz).

Use of the chameleon subwoofer array for low-frequency room mode correction is highly effective both in the time and frequency domains. The frequency domain shows an overall variance improvement of nearly 80% while the time domain shows variance improvements of nearly 100% in some cases, giving very accurate waveform accuracy for transient signals.

Imperfections in these results are similar to those in previous tests. Points within the walking path located near a source cluster generally will show reduced benefits of correction while correction around the crossover region of 57 Hz also may show certain imperfections due to interference between the two operating bands of the system.

7. COMPARISON TO EXISTING METHODS

In order to judge the effectiveness of this new room correction technique, performance must be compared against that of some of the methods mentioned earlier in the paper including single/multiple point equalization and strategic subwoofer placement.

A simulation was set up with a single omnidirectional subwoofer in the corner of the room with the uncorrected listening location frequency response shown in Figure 7.1. An MLS signal was used as the measurement signal for both equalization and source placement cases with the resulting frequency responses shown in Figures 7.2 – 7.4.

The uncorrected system gives a spatial variance of 4.4950 dB from 30 – 110 Hz (the range of chameleon subwoofer array correction). Both the single and multiple point equalization methods give exactly the same spatial variance value indicating that these methods do not give any improvement in terms of variance from point-to-point in the listening area.

A simple average (single equalizer) of the responses was used for the multiple point equalization, which is very basic compared to other multi-point techniques (with multiple equalizers), which have been shown in other work to give improvements in terms of spatial variance.

The passive room correction method of placing an omnidirectional subwoofer at each wall midpoint, on the other hand, gives an average spatial variance of 3.7576 dB, corresponding to a 16.4% improvement. This decrease in spatial variance can be attributed to the system's source-listening location coupling being changed with the updated source configuration. Both forms of equalization did not alter this factor, thus no improvement occurred.

While the passive correction method showed reasonable improvements to the room response, no method explored here could approach the 79% improvement in spatial variance demonstrated with the chameleon subwoofer array.

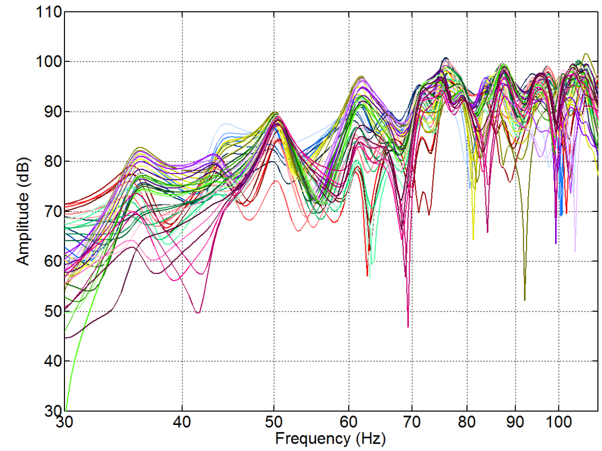

Figure 7.1: Uncorrected system frequency response for a 49-point walking path as shown in Figure 6.6

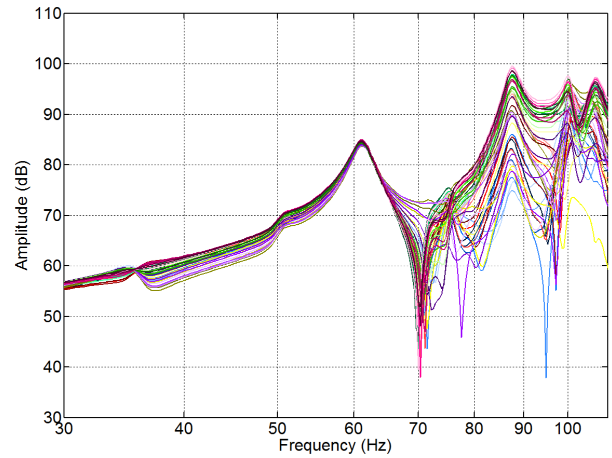

Figure 7.2: Corrected system frequency response using passive correction (four subwoofers at wall midpoints)

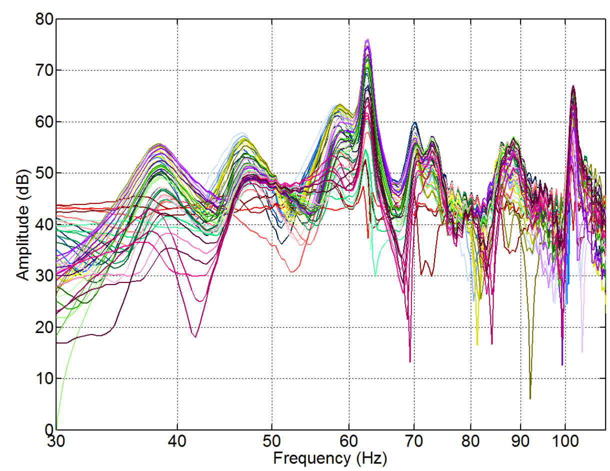

Figure 7.3: Corrected system frequency response using active correction (single point equalization)

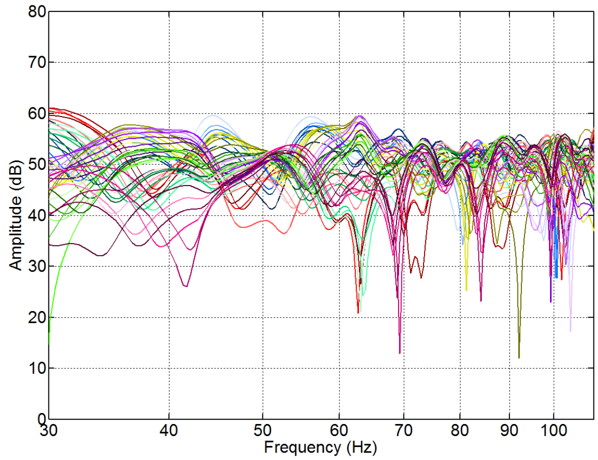

Figure 7.4: Corrected system frequency response using active correction (multiple point equalization)

8. CONCULSIONS AND FUTURE WORK

A new low-frequency room response correction system has been presented using "chameleon" subwoofer arrays. These arrays operate using a set of measurements from strategically placed locations within the targeted listening area. The simplest of these systems has two operating bands. The first band's range is defined by the overall dimensions of the listening area. In this low band only the omnidirectional components of the chameleon arrays are active due to the inefficiency of dipole sources at very low frequencies. This low band generally extends from just below the first room mode to around 50 Hz. The second band operates using all source components within the array and has a range that extends to around 120 Hz.

The example of this correction technique presented in Section 6 utilizes four source clusters within the chameleon array and sixteen measurement points within the listening area. Once the initial measurements are taken, the system is fully configured and no further measurements are required unless the listening area or source clusters are moved.

This form of room correction has shown to benefit all points within the listening area in both the frequency and time domains, where frequency behavior can improve by nearly 80% (in terms of spatial variance) and transient behavior can often improve by nearly 100% (in terms of waveform integrity) in some cases.

There are a number of issues that need to be addressed for a practical implementation of these chameleon subwoofer arrays. The current cluster configuration requires a sufficient height to give the z-direction dipole component enough clearance from the floor to operate efficiently. Also, tests must be conducted comparing an empty listening area (no people) to a full listening area to determine the system sensitivity to increased room absorption and room obstacles due to the presence of a group of people. Lastly, it has been noted that certain source cluster placements give more reasonable filter coefficients than others. Well performing configurations generally correspond to ideal omnidirectional subwoofer placement methods. This issue of preferred placement must be well understood before the implementation of a real-world system can be feasible.

There is a significant amount of research remaining concerning this new room correction method before it can be practically implemented. The results presented in this paper, though, give strong evidence that this method can be extremely effective and can provide greater low-frequency correction benefits over a large listening area than conventional room correction methods.

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