

# Subjective Preference of Modal Control Methods in Listening Rooms

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Room modes are well known to cause unwanted effects in the correct reproduction of low frequencies in critical listening rooms. Methods to control these problems range from simple loudspeaker/listener positioning to quite complex digital signal processing. Nonetheless, the subjective importance and impact of these methods has rarely been quantified subjectively. A number of simple control methods have been implemented in an IEC standard listening environment. Eight different systems were setup in the room simultaneously and could therefore be tested in direct comparison to each other. A panel of 20 listeners was asked to state their preferred configuration using the method of paired comparison. Results show clear winners and losers, indicating an informed but simple strategy for efficient control.

## 0 INTRODUCTION

The problems of room modes and their effects on the correct reproduction of low frequency audio have warranted a large body of research. Initial studies into architectural design for listening rooms concentrated on objective factors such as modal densities and distributions and their room counterparts of volume and room aspect ratios [1, 2]. Albeit impressively accurate in terms of predicting objective parameters of the room response, these initial studies led to a number of theories for room design that were lacking support from a sound knowledge of listener's perception in the room. In many cases a number of design techniques, mainly based on room aspect ratios, were developed, supposedly to warrant an optimally sounding low frequency reproduction [3–5].

Later studies into the subjective perception of modal factors [6, 7] provided some further guidance into the design of rooms that could be used in optimization techniques such as those described by Cox et al. [8]. Further studies into perceptual aspects of room modes provided the scientific evidence required to attribute a relative importance to the various modal parameters [9–12].

Parallel advances in the area of loudspeaker and digital signal processing (DSP) design have led to a number of modal control techniques that attempt to improve the loudspeaker-room interaction through a number of methods (see [8, 13, 14] for examples). More recently, a psychoacoustic method where problematic modes are

equalized and supplemented with “virtual bass” has been presented [15]. These techniques have various degrees of success in the improvement of objective modal response parameters and necessarily involve various levels of cost/efficiency.

It could be argued that the efficacy of a given method may be measured by how well it improves the perceived quality rather than some objective metric obtained from the room response. Of course, to find the correlation between objective metrics and subjective quality is the ultimate goal and this work is ongoing. For highest level of performance a system should be scrutinized under critical listening conditions (this concept is defined in Section 2.1 of this paper).

Research into the subjective perception of modal control techniques is scarce. Perhaps the earlier and best example is provided by Antsalo et al. where two modal equalization methods are investigated subjectively [16]. The performance of commercially available room correction methods have also been investigated by Olive et al. [17]. In contrast, the work presented here investigates a number of configurations based on simple principles such as source and listener positioning for single and multiple source configurations, simple magnitude equalization, and one type of “active” modal control recently published. To afford a practicality and applicability aspect to this work, it is the intention of the authors to investigate configurations that are simple to implement and within reach of most professional music studios. With this aim, all of the configurations implemented use only one or two subwoofers, with the exception

of one configuration that uses four. Two of the configurations require a simple DSP unit providing delay and gain adjustment for a single channel. Another two require a pink noise generator and a frequency analyzer.

The paper describes the implementation of each configuration in detail in Section 1 to allow interested readers to replicate the conditions tested. For the listening tests, all configurations had to be implemented simultaneously, requiring a total of eight matched subwoofer loudspeakers. Consequently, there is substantial overlap in terms of loudspeaker location between some configurations and, understandably, not all possible configurations have been tested. The aim is to demonstrate that a few simple, well informed steps may afford substantial perceptual improvement in the loudspeaker-room interaction at low frequencies. A single listening position was tested.

The listening test methodology, described in Section 2, has been designed to allow allocating each configuration to a subjective preference scale. The eight configurations have been simultaneously installed in an IEC standard listening room and a panel of 20 healthy hearing listeners have been asked to select their preferred configuration through a series of paired comparisons consisting of all possible combinations. The paired comparisons have then been converted to a ratio scale using the *law of comparative judgment* ([18], quoted in [19]).

## 1 SYSTEM IMPLEMENTATION

Eight modal control configurations of varying degrees of technical complexity have been set-up in the Listening Room at the Acoustics Research Centre, University of Salford, UK. The room meets the standards set out in ITU-R BS 1116-1 [20], BS 6840-13, and IEC 268-13 [21] and has the dimensions  $L_x = 5.8$ ,  $L_y = 6.6$ , and  $L_z = 2.8$  m.

To implement all configurations simultaneously, eight active Genelec 7050B subwoofers were used. These were set to reproduce low frequency signals below 120 Hz. A Genelec active monitor (1029A) was used to reproduce the high frequency content. Signals sent to this speaker were high passed at 120 Hz and the speaker was placed in front of the listener at a distance of 2 m. For each configuration, the same mono bass signal is used to drive each speaker. The distance from listener to each subwoofer corresponds to their positions according to each configuration listed in Table 3 (Section 7) and no compensation for group delay between subwoofer and full frequency range speaker has been attempted. The listening position was chosen as the absolute center of the room along the width and length dimensions (2.9,3.3,1.2). The listeners were seated, with ear height at 1.2 m. This particular listening position was chosen due to the relationship between modes (i.e., the modal pressure distribution for a single mode) and listener location. The listener position is on the intersection between nodal lines for odd order modes along the length and width dimension of the room. In theory, this seating position should not be strongly affected by room modes

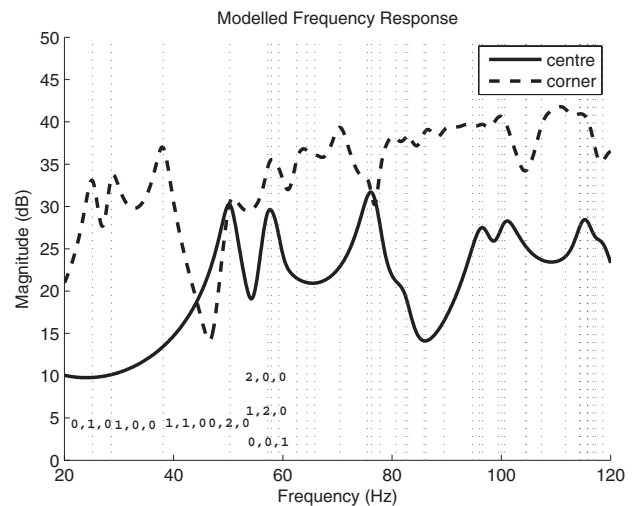


Fig. 1. Modeled responses for subwoofer in corner and two receiver positions. Eigenfrequencies are indicated by straight lines. Modal orders for first six modes are indicated for width, length, and height respectively.

associated with these orders as is evident in Fig. 1, which shows the modeled response with subwoofer in one corner and receiver in the center. For reference, a corner-to-corner prediction is also shown in Fig. 1 (dashed curve). The vertical lines indicate the eigenfrequencies supported by this combination of room ratio and volume. The modal orders for the first six modes are indicated next to their modal eigenfrequencies.

Although, only one listening position was studied, measurements were taken around this and show the expected differences in the magnitude response; anecdotal evidence collected among the listeners has shown that the general character of the response for each configuration did not differ noticeably at positions around the listening position, suggesting these results may be extrapolated to a small listening area around it. An acoustically transparent curtain was used to hide all loudspeakers.

Coupling of subwoofers to modes is a crucial element in modal control systems and as such the definition of modal order, and its relation to modes, is important. Modal order,  $n$ , describes the number of half-wavelengths  $n \times \frac{\lambda}{2}$  in the modeshape that “fits” within a certain room dimension to produce a room mode. Table 3 lists details for number of subwoofers, their positions, control method applied, and resulting modal coupling strength.

Measurements were taken for each system at the listening position using an appropriately defined sine sweep method. Third-octave low frequency decays have been calculated from the measured impulse responses. Plots with measured magnitude frequency responses and modal decays are presented for each system. The thresholds for detection of modal decay defined by Avis et al. [9] are also presented in these plots for direct comparison. Table 4 in Section 6 lists the measured decays numerically.

The following sections provide a detailed description of each system tested.

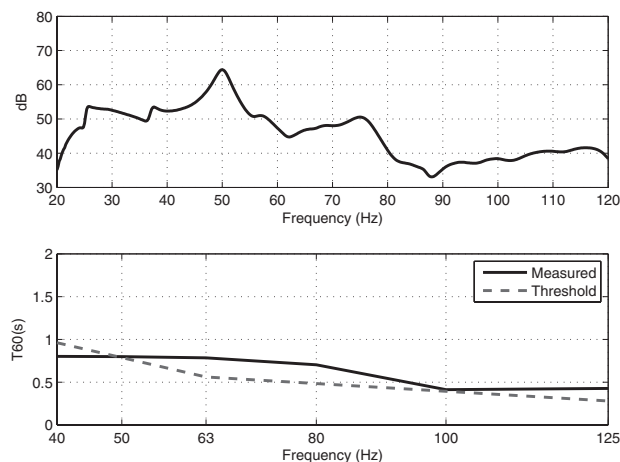


Fig. 2. Measured magnitude frequency responses and third-octave decay times ( $T_{60}$ ) for single subwoofer placed in the corner. Measured at listening position. The dashed line in decay plots represents the threshold of detection for low frequency decays defined by Avis et al. [9].

### 1.1 Corner Configuration with No Equalization

This is the most basic configuration tested. It is simply based on a subwoofer placed in the front left corner of the room, on the floor. This could be deemed as the “worst case scenario,” since a subwoofer placed in the corner of the room, where modal pressure is at its highest, will couple strongly to all room modes.

The modal features in the measured response are evident in the form of resonant peaks in the frequency response and long low frequency decays extending to nearly one second (Fig. 2). A comparison can be made between the measured response shown and its modeled prediction shown in Fig. 1. It is seen that these are generally in agreement, especially at strong resonant frequencies of 50 and 75 Hz, although some differences can be observed. These differences between modeled and measured data are to be expected as even small discrepancies in room dimensions, physical positions of microphone, and subwoofer have a bearing on the response. Furthermore, the accuracy of the model in predicting the exact damping conditions in the room also influences the modeled response.

### 1.2 Corner Configuration with Magnitude Equalization

This configuration is identical to that described in Section 1.1 where the subwoofer is placed in the corner. However, in this case, magnitude equalization has been applied to reduce the magnitude frequency response variation measured at the listening position. This is a common approach, now also available in various commercial software applications (e.g., [22]), and may be typically achieved either with graphic or parametric equalizer units placed between the signal source and the loudspeaker. In this work parametric equalizers have been used and applied only to the subwoofer signal, below 120 Hz. The equalization procedure applied is as follows.

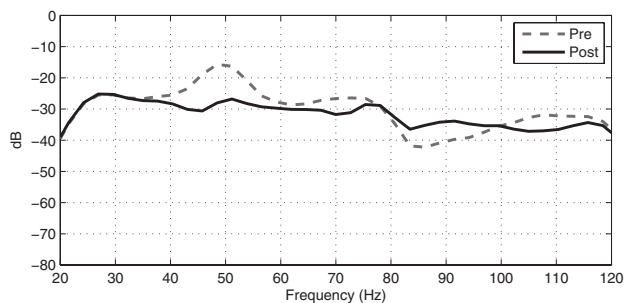


Fig. 3. Measurement of frequency response at the listening point (center of the room) with one subwoofer placed in the corner—before and after equalization has been applied.

A pink noise source signal was replayed through the subwoofer reproduction system. Reading the signal at a reference microphone placed at the listening position, the parameters of the filters were adjusted until the microphone response matched, as close as possible, the original frequency response curve of the pink noise—this process can be easily achieved with an audio workstation wave editor. Once this process was completed, the filter settings were applied to the audio signals being sent to the subwoofer.

Important notes about this type of procedure:

1. The application of drastic filter parameters using high Q-factor and gain settings is not advisable. High gains may drive the loudspeaker into non linear behavior; and very high Q-factor filters have a long decay artifact in the time domain that may also be noticeable, oddly enough, as a resonance! As this type of equalization is applied to the signal *before* it is reproduced by the loudspeaker, the first wavefront, as it passes through the listening position and before it gets modified by the room response, will contain these artifacts, which may be audible and degrade the perceived quality.
2. In most cases where this equalization procedure is applied, one makes use of a frequency analyzer that smooths the response in third-octave or even octave bands. Fig. 3 shows the response of the system before and after equalization in third-octave bands. The improvement is clear. However, it should be noted that non-smoothed data (Fig. 4) shows a different outcome. An optimal application of this equalization procedure requires higher resolution for the display of frequency domain data in order to correctly identify center frequencies and bandwidths of room modes. The parametric equalization settings can then be adjusted accordingly.

It is clear from a comparison of Figs. 2 and 4 that the application of filtering has merely shifted some problems, such as the strong mode at 50 Hz, while introducing other irregularities in the response. Another interesting result is the modifications obtained in decay times. There is a clear reduction at 60 Hz and above but an unexpected increase at 40 and 50 Hz, to beyond 1 second! Whether this procedure

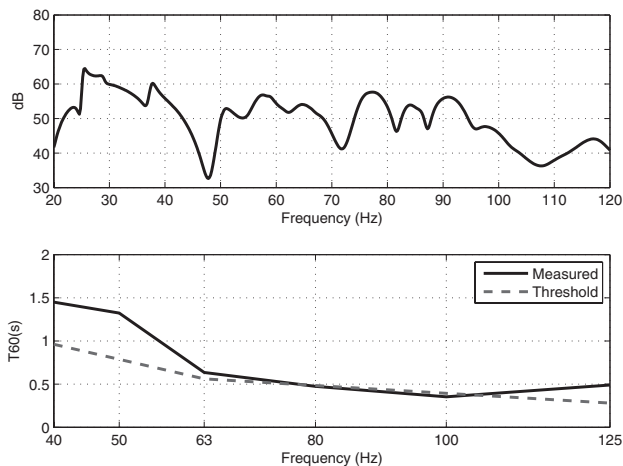


Fig. 4. Measured magnitude frequency responses and third-octave decay times ( $T_{60}$ ) for single subwoofer placed in the corner with applied equalization. Measured at listening position. The dashed line in decay plots represents the threshold of detection for low frequency decays defined by Avis et al. [9].

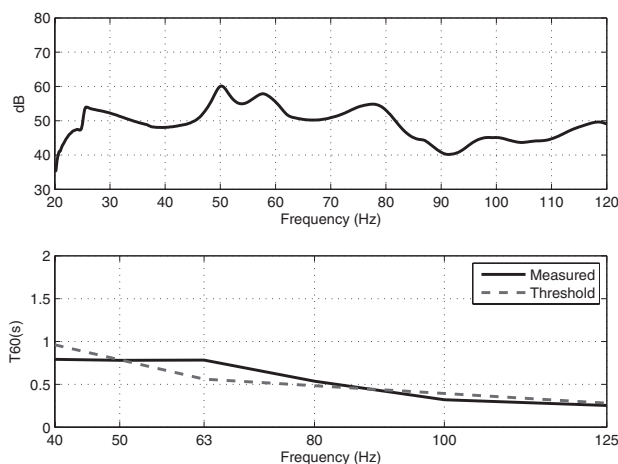


Fig. 5. Measured magnitude frequency responses and third-octave decay times ( $T_{60}$ ) for single subwoofer placed in the center. Measured at listening position. The dashed line in decay plots represents the threshold of detection for low frequency decays defined by Avis et al. [9].

has resulted in a perceptual improvement will be discussed in Section 2.2.

### 1.3 Center Configuration with No Equalization

This configuration is simply based on a single subwoofer placed on the floor near the front wall, directly in front of the listener. The speaker is effectively on the width-wise symmetry line of the room and thus weakly coupled to any modes that have a null in their pressure response along this line (i.e., odd order width modes). This speaker position is not uncommon in professional and home studios, where the speaker is simply placed under or behind the mixing desk, often for convenience. It may be argued that most studio owners/users will not place the subwoofer exactly on the symmetry line as is tested here, but slightly displaced to one side. While this could be considered good practice in trying to “miss out” the node associated with odd order

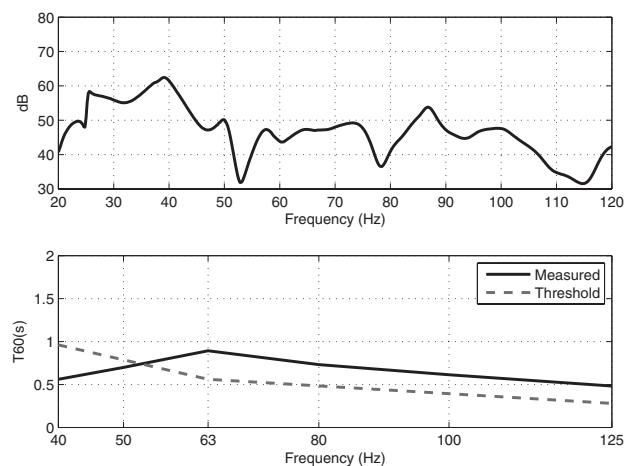


Fig. 6. Measured magnitude frequency responses and third-octave decay times ( $T_{60}$ ) for single subwoofer placed in the center with applied equalization. Measured at listening position. The dashed line in decay plots represents the threshold of detection for low frequency decays defined by Avis et al. [9].

modes along the width dimension, it has been shown that noticeable differences are only evident when substantial displacements in the order of more than 0.5 m are applied [23].

A further reason for using this configuration is that the subwoofer position is identical to those used by more complex (multiple subwoofer) configurations, such as the “Front-Back” (see Section 1.6) thus allowing one more system configuration to be included in the test.

### 1.4 Center Configuration with Magnitude Equalization

Identical in placement to the configuration described in Section 1.3, this configuration includes equalization of the magnitude frequency response. The same equalization method as described in Section 1.2 has been employed. Third-octave band results clearly showed an improvement over the “pre EQ” response but closer inspection of the non-smoothed data (Fig. 6) shows a somewhat less desirable magnitude frequency response. The modifications to decay times are also interesting: there appears to be a reduction of decays at 50 Hz and below; and an increase in decay times at 80 Hz and above.

### 1.5 Controlled Acoustic Bass System (C.A.B.S.)

This configuration has been recently proposed by Celestinos and Nielsen [24]. Interestingly, a patent relating to the same principle had been submitted in 2000 by [25] (Genelec) and around the same time Goertz et al. also published a paper describing a very similar system [26]. The acronym suggested by Celestinos and Nielsen (C.A.B.S.) will be used throughout this paper when referring to this system.

C.A.B.S. is perhaps the most complex and costly implementation of all tested here. It is based on a “source to sink” principle where the sound wave is generated by two subwoofers placed at the front of the room and “absorbed” by



two rear subwoofers as it reaches the rear wall. The configuration therefore employs four subwoofers in total, where the “source” speakers are placed at:

$$y_1 = y_2 = 0 \quad (1)$$

$$x_1 = 1/4L_x; x_2 = 3/4L_x \quad (2)$$

$$z_1 = z_2 = 1/2L_z \quad (3)$$

and the “sink” speakers are placed in exact mirror positions against the rear wall of the room.

Critical to this system, and as a result of the subwoofer placement, a plane wave is created along the  $y$  (length) dimension of the room. With the “source” speakers placed equidistantly from the boundaries in the  $x$  (width) and  $z$  (height) planes, the reflections from these boundaries act as a two-dimensional source array creating the plane wave. It is important to note that this principle has an upper cut-off frequency of:

$$f_{max} = \frac{c}{d} - \Delta\epsilon \quad (4)$$

where  $d$  is the distance between the speakers and  $\Delta\epsilon$  is a factor depending on the room absorption. Assuming negligible absorption at very low frequencies (not untypical for a well isolated, brick wall, room) and our distance of 2.9 m between speakers, we get an upper frequency of approximately 120 Hz, which is close to the cross-over between subwoofer and “satellite” speaker. A similar frequency limit applies to the  $z$  dimension since the subwoofers are placed at a distance of 1.4 m from both floor and ceiling.

The “active” aspect of this control configuration comes into place as the plane wave reaches the rear wall. At this location, the “sink” speakers reproduce the same signal in anti-phase, which has the effect of cancelling the plane wave reflection from the rear wall. The delay between “source” and “sink” speakers has to be aligned to the time taken for the wave to propagate along the length of the room. This can be determined using:

$$\Delta t = \frac{L_y}{c} \quad (5)$$

where  $L_y$  corresponds to the length of the room. Distances are in meters ( $m$ ) and  $c$  is the speed of sound (343  $m/s$  for the calculations presented). The delay used for our tests was 0.0201 seconds.

The gain reduction applied to the “sink” speakers must match the attenuation undergone by the traveling wave. Since it is considered as a plane wave, simple spherical propagation rules do not apply. Indeed, for a theoretical plane wave no amplitude reduction would be expected. Celestinos and Nielsen [24] mention that “delay and gain were fine tuned empirically” in their system. A simple process where the variation in the frequency response at the listening position is minimized by adjusting the gain of the rear speakers may be used.

For the tests presented here, we defined the required gain reduction using an optimization procedure based on measurements taken in the room. The cost function minimized was the standard deviation of the transfer function (magni-

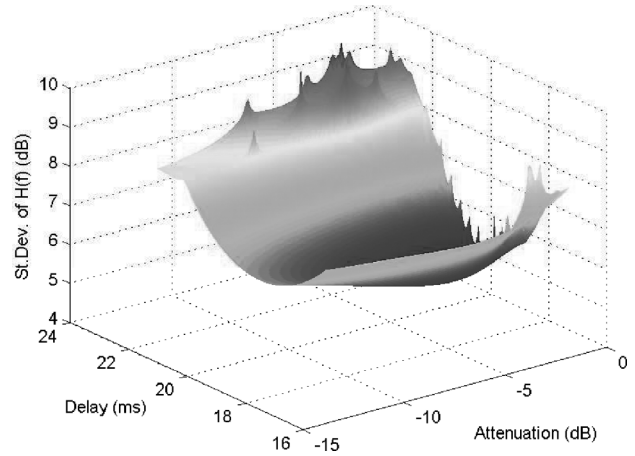


Fig. 7. Search surface used to optimize C.A.B.S. delay and gain parameters.

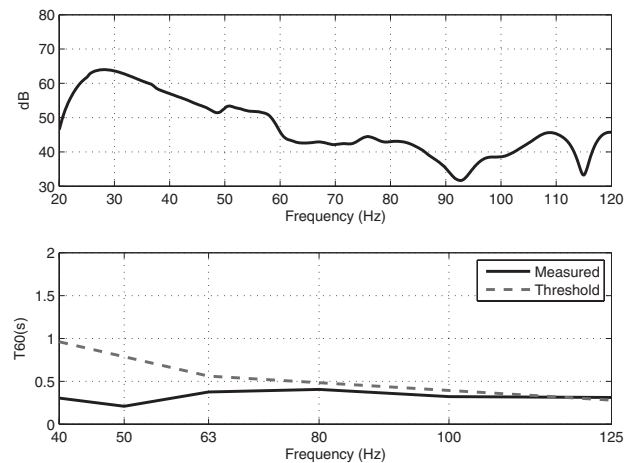


Fig. 8. Measured magnitude frequency responses and third-octave decay times ( $T_{60}$ ) for C.A.B.S. Measured at listening position. The dashed line in decay plots represents the threshold of detection for low frequency decays defined by Avis et al. [9].

tude frequency) obtained from combining each of the four subwoofers at the listening position. The optimization parameters were gain reduction and delay of the rear speakers. The optimized delay closely matches that obtained from Eq. 1; the optimized gain reduction for the configuration tested was found to be 1.8 dB. Fig. 7 shows the search space used to find the optimal parameters for C.A.B.S.

Comparing the response of the C.A.B.S. system (Fig. 8) with that of the corner system (Fig. 2) it is clear that both the magnitude and  $Q$ 's of modal peaks have been significantly reduced. The magnitude response is one of the smoothest of all the systems measured. Also noteworthy is the drastic reduction of energy in the room. The decay times at the very low frequencies of 40 and 50 Hz have been reduced to about 0.3 seconds. At such low frequencies, and considering that no added damping has been introduced in the room, the objective performance of this configuration is impressive.

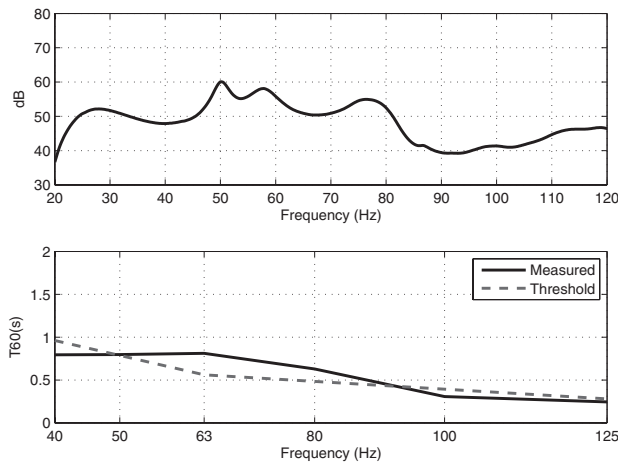


Fig. 9. Measured magnitude frequency responses and third-octave decay times ( $T_{60}$ ) for “Front-Back.” Measured at listening position. The dashed line in decay plots represents the threshold of detection for low frequency decays defined by Avis et al. [9].

### 1.6 Front-Back (FB) Configuration

Welti and Devantier have performed an extensive study on methods to optimize low frequency reproduction using multiple subwoofer configurations [13]. The aim of the optimization procedures applied in the study was to achieve an even spatial distribution over a defined listening area. The systems described employ multiple subwoofers and signal processing control over phase, attenuation, and cut-off frequency. Interestingly, one of the simplest configurations presented is shown to reduce modal problems and provide low seat-to-seat variation even though no digital control is applied to the subwoofers. This configuration uses two subwoofers, one placed at the front and another at the rear wall, directly in front and behind the listening position (see Table 3 for more detail). Speakers are placed on the floor and driven in phase. This configuration will be here described as “Front-Back” (FB).

Given its particular positioning and phase relationship, weak coupling is expected for odd order modes along the x and y dimensions resulting in an improved response compared to excitation of a single subwoofer.

Observation of Fig. 9 reveals some modal artifacts may be seen in the frequency response and the system is associated with shorter decays compared to single subwoofer in the higher frequency range. Interestingly, the decays at the lower frequencies (40, 50, and 63 Hz) are similar to single subwoofer configurations whereas the higher range under study (80 and 125 Hz) shows a significant reduction. The specific positioning of speakers and listener and the in-phase excitation of the two subwoofers results in an interesting coupling pattern with modeshapes (refer to Fig. 1 for detail of modal orders):

- Coupling to odd order width modes is weak for both subwoofer and listener—this means the effects of modes at 28 Hz and 38 Hz are not revealed in the measured response as expected.

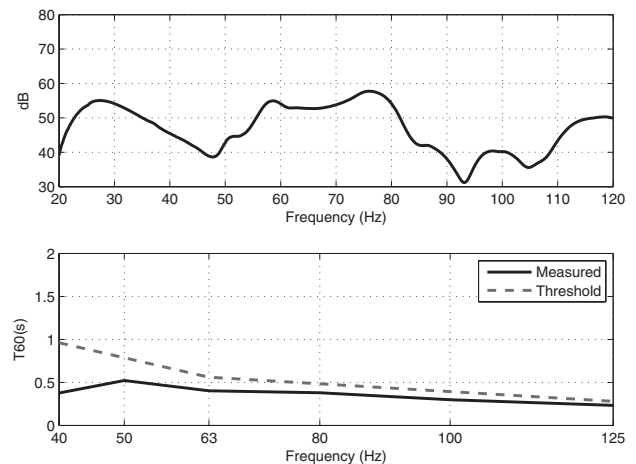


Fig. 10. Measured magnitude frequency responses and third-octave decay times ( $T_{60}$ ) for “Source-to-Sink.” Measured at listening position. The dashed line in decay plots represents the threshold of detection for low frequency decays defined by Avis et al. [9].

- Coupling to even order width modes is strong and thus their effects are clear, for example at about 58 Hz (see mode 2,0,0 in Fig. 1).
- Coupling to odd order length modes is weak at source, since both subs try to drive out of phase portions of the mode-shape; and receiver which sits in a nodal line. Room modes at 25 Hz (0,1,0) and 38 Hz (1,1,0) are not noticeable in the response.
- Coupling to even order length modes is strong for both subwoofers and receiver. As a result, the modes 0,2,0 at 50 Hz and 1,2,0 at around 58 Hz are very evident in the measured response.

### 1.7 Single Source-to-Sink (SSS)

This is an optimization on the previous configuration, where the rear subwoofer is now used as a “sink” radiator in similar fashion to the C.A.B.S. system described in Section 1.5. The settings for gain and delay are identical to the C.A.B.S. since the distance between speakers is the same. The use of a single generating subwoofer and its specific modal coupling no longer ensure y dimension plane wave propagation under the same conditions as those in C.A.B.S. Since the subwoofer is now equidistant to both walls by  $L_y/2 = 3.3m$ , the cut-off frequency for plane wave propagation has been reduced to around 50 Hz. A similar limit frequency is obtained for the height.

Surprisingly, the SSS configuration still achieves a fairly flat frequency response and a strikingly short time domain response. In contrast to the “Front-Back” configuration, this improvement of the response is clearly afforded by the “active” nature that the source-to-sink approach brings.

### 1.8 Opposite Phase-Opposite Corner (OPOC)

This implementation is based on the theory that specific subwoofer/listener position and phase inversion achieve weak coupling to all modes. Both speakers are placed on

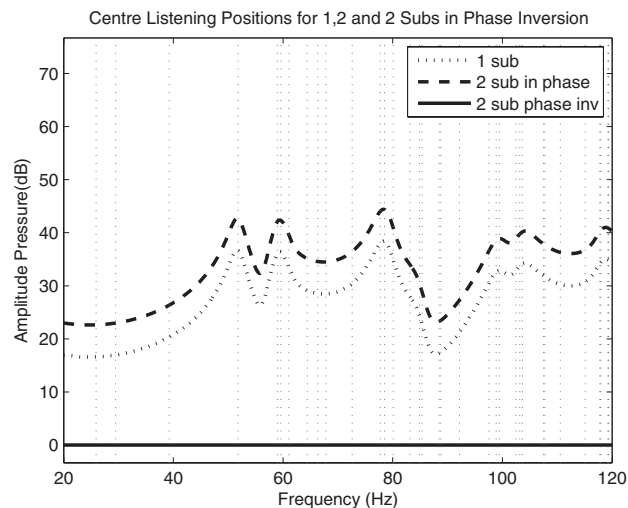


Fig. 11. Modeled response for OPOC system at listener position. Single subwoofer, two subs in phase, and the combined anti-phase configuration.

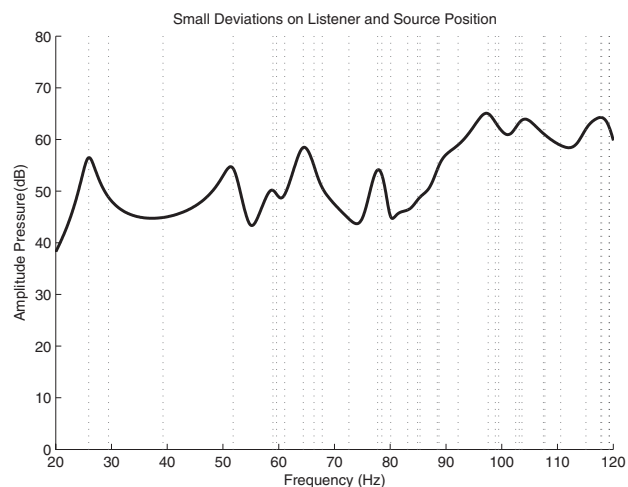


Fig. 12. Modeled response for OPOC with deliberate "errors" for source and receiver positions. Combined anti-phase configuration.

the floor in opposite corners of the room (side to side and front to back) and set in opposite phase. According to their placement and phase relationship, the subwoofers should not couple to even order modes in both width ( $x$ ) and length ( $y$ ) dimensions. They do couple strongly to odd order modes in these dimensions, but the chosen listener position, at the intersection of nodal lines for odd order modes, ensures that these are not picked up by the listener.

Theoretically, the response of the OPOC system where the listening position is the absolute center of the room, should result in a totally flat response with no excitation or reception of any room modes except those in the  $z$  dimension. Fig. 11 models the response of a single subwoofer in the corner, a second subwoofer in the opposite corner acting in phase with the first, and finally the theoretical scenario of both subwoofers out of phase—a flat response at 0 dB. In a real room scenario however, the positions of both speakers and listeners are never exact, and so this ideal theoretical response is highly unlikely. Fig. 12 models the response

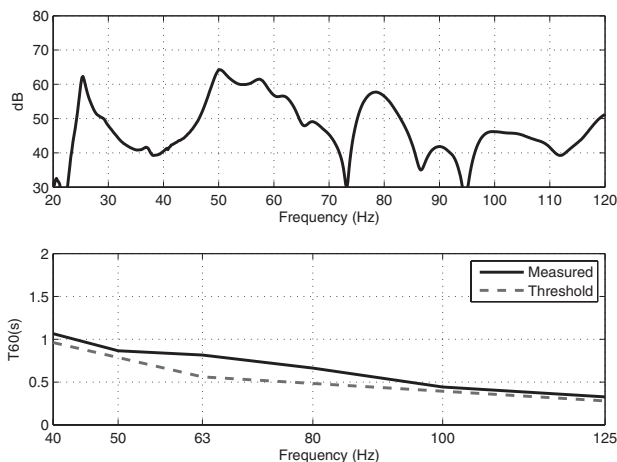


Fig. 13. Measured magnitude frequency responses and third-octave decay times ( $T_{60}$ ) for single "Opposite-Phase-Opposite-Corner." Measured at listening position. The dashed line in decay plots represents the threshold of detection for low frequency decays defined by Avis et al. [9].

obtained if the source and receiver positions are allowed to deviate just 20 cm from their ideal symmetrical positions. The result is now similar to that actually measured in the room for this configuration (Fig. 13).

Indeed, in the measured response there is clear evidence of modal peaks and, perhaps more worryingly, the decays in the lowest range have been extended beyond the values obtained for a single subwoofer. Anecdotal evidence from the listening test also revealed that an audible flutter could be heard, giving a particularly unnatural feeling to the playback. A periodic pattern could also be observed in the time domain impulse response (not included in this paper).

## 2 LISTENING TESTS

### 2.1 Methodology and Setup

A panel of 20 listeners was tested. Twelve of the listeners tested were part of an expert panel participating in a larger study on low frequency reproduction quality in rooms [27]. The remaining eight listeners may be considered naïve listeners, with no prior experience in listening tests for the assessment of low frequency reproduction conditions.

Two commercially available music samples have been used. These were chosen according to their low frequency content and how adequately they allowed effects such as resonances and frequency imbalances caused by modal behavior in the room, to be heard. A short description of temporal and tonal content for each sample follows:

*Sample A (Dynamite)—Fast paced bass guitar notes in a funk genre. Bass notes closely spaced in time.*

*Sample B (Lenine)—Slow, individual bass notes with short attack and decay; defined and isolated bass drum hits.*

All listeners were allowed a training period where they could get familiar with the samples and configurations under test, as well as the test interface. A touch screen and a MATLAB written user interface was used to assist in the selection of samples to be played and to collect listeners'

Table 1. Coefficient of consistency for subjects tested using music Sample A.

Subject	1	2	3	4	5	6	7	8	9	10
$\xi$	0.55	0.20	1.00	0.85	0.65	0.95	0.80	0.80	0.80	0.70
Subject	11	12	13	14	15	16	17	18	19	20
$\xi$	0.80	0.70	0.80	0.50	0.55	0.05	0.60	0.80	0.25	0.80

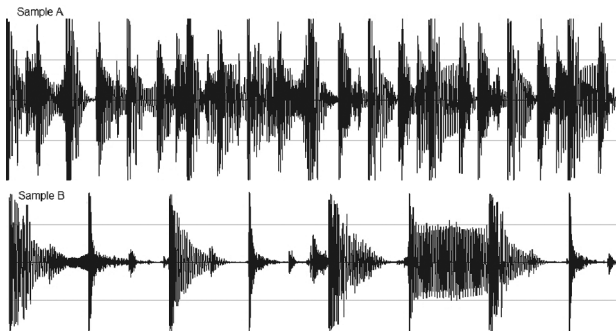


Fig. 14. Waveforms of the two music samples.

responses. The touch screen was placed within reach of the listener but such that it wouldn't cause any unwanted reflection artifacts.

The test methodology is based on the *Law of Comparative Judgment* (commonly known as “paired comparison”), which poses a very simple task to the subjects: For each presentation pair, subjects were asked to state their preference based on the configuration they believed to provide the best low frequency reproduction for critical listening. The underlying methodology for this test is described in more detail in [27], which is also reported in this issue of the Journal. The definition of critical listening is provided to the subjects as:

*The process where you listen to the audio program in a way that allows you to evaluate and interpret its characteristics in depth and make decisions regarding any problematic features such as resonances, frequency or level imbalances, lack of definition, etc. An example of a critical listening environment would be in a recording or mastering studio.*

Subjects were also asked to concentrate on the low frequency reproduction aspects of each configuration. All possible system pairs have been tested, which corresponds to 28 auditions for each music sample.

All systems were reproduced in mono and calibrated to reproduce an  $L_{eq}$  level of 85 dB (corresponding to  $LA_{eq}$  of 76 dB) across the frequency range, for the duration of the sample (approx. 6 seconds).

## 2.2 Results

Before a ratio scale could be built from the paired comparison results, the data was analyzed for individual subject performance. This is known as the “coefficient of consistency”

and is a normalized ratio of the number of circular triads in the data set for a given subject compared to the total number possible. A circular triad is evidence of intransitivity. For example, where the symbol “>” denotes preference:

$$A > B > C > A$$

is regarded as a circular triad. A more consistent result would be:

$$A > B > C < A$$

Tables 1 and 2 show this coefficient of consistency for each of the 20 subjects grouped by music sample. In Table 1 six subjects show a consistency of 0.6 or lower—results where the number of circular triads is likely to result from random answers. In Table 2, five subjects have scored below 0.6. The data for these subjects was thus removed before the paired comparison results were calculated.

Analysis was carried out for data collected grouped by music. The results from the paired comparison tests were analyzed using Thurstone's *Law of Comparative Judgment* [18]. This methodology allows ordering the systems under test on a ratio preference scale, based on normalized scores, or z scores. The z scores relate to how consistently each configuration has, on average, been rated better than all the others. We have analyzed statistical significance in three tiers:

1.  $z < 1.96$ ,  $p > 0.05$ , not significant
2.  $1.96 < z < 2.58$ ,  $p < 0.05$ , significant at 5% level
3.  $z > 2.58$ ,  $p < 0.01$ , significant at 1% level.

The z score for each system can be used to determine whether a significant improvement in reproduction quality has been achieved.

Results are shown in Fig. 15, grouped by sample. Results for each sample have been normalized to the lowest scoring system (subwoofer in the corner for both samples). Direct comparison across music sample data is meaningless.

Analysis of data reveals a striking difference in trends between results collected using Sample A and those using Sample B. Results for Sample A show that it is not particularly helpful in revealing perceptual differences between the systems under test—all systems lie within one z-score of the worst system and never cross the 5% significance level. We believe the characteristic temporal and musical

Table 2. Coefficient of consistency for subjects tested using music Sample B.

Subject	1	2	3	4	5	6	7	8	9	10
$\xi$	0.50	0.70	0.80	0.90	0.90	1.00	0.70	0.15	0.35	0.60
Subject	11	12	13	14	15	16	17	18	19	20
$\xi$	0.95	0.85	0.80	0.90	1.00	0.80	0.65	0.95	0.40	0.40



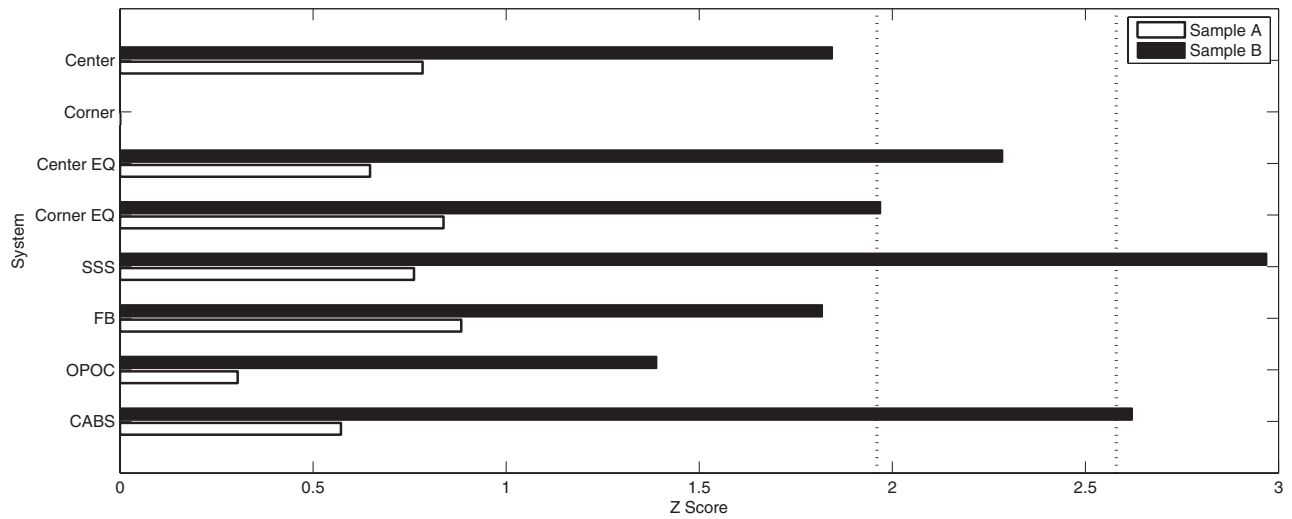


Fig. 15. Performance for each system in terms of z score grouped by music sample.

differences between the samples are responsible for this. These characteristics in Sample B allow modal artifacts to be heard more clearly and thus lead to better informed judgment regarding the reproduction quality of each system. Results for Sample A lack the statistical evidence to warrant further discussion. However, this result is in itself of extreme importance since it establishes that the selection of tests sample(s), particularly in this type of “realistic” testing, is vital to extract meaningful and reliable results.

The analysis will now focus on results obtained with Sample B. Under these test conditions, the system based on one single subwoofer placed in the corner of the room is deemed as worst. Moving the subwoofer to the center appears to improve the perceived quality as do the FB and OPOC systems although the z score does not reach the minimum level for a statistically significant result. This level is, however, reached when the single sub systems are equalized. In this case the perceived difference leads to a z

Table 4. Low frequency decay times, in seconds, calculated in third-octave bands for all systems tested.

System Name	40	50	63	80	100	125
corner	0.80	0.80	0.78	0.72	0.41	0.43
center	0.78	0.78	0.78	0.55	0.32	0.25
cornerEQ	1.42	1.33	0.64	0.49	0.35	0.48
centerEQ	0.52	0.69	0.89	0.74	0.61	0.47
FB	0.79	0.80	0.80	0.65	0.31	0.24
OPOC	1.07	0.87	0.82	0.68	0.44	0.32
SSS	0.38	0.52	0.40	0.39	0.30	0.22
CABS	0.32	0.21	0.37	0.40	0.32	0.31

score above 1.96 suggesting the perceived improvement in quality is indeed significant. The SSS and C.A.B.S. systems achieve a z score above 2.58, a highly significant score, suggesting that, according to our panel, these are considered the best quality systems. Another interesting outcome is that

Table 3. System Details. Excitation coupling is shown for room dimensions x and y. All systems have subwoofers placed on the floor where modal coupling to z modes is strong; except for C.A.B.S. where subwoofers are placed at mid height thus coupling weakly to all odd order z modes. Listener position is in the absolute center of the room (Lx/2,Ly/2) at a height of 1.2m, coupling weakly to all odd order modes.

System Name	Control Method	Subwoofer Coordinates	Coupling in x dimension	Coupling in y dimension
corner	single subwoofer	s1: 0.2, 0.2, 0.2	all n=strong	all n=strong
center	single subwoofer and positioning	s1: 2.9, 0.2, 0.2	odd n=weak even n=strong	all n=strong
cornerEQ	single subwoofer and magnitude EQ	s1: 0.2, 0.2, 0.2	all n=strong	all n=strong
centerEQ	single subwoofer and positioning with magnitude EQ	s1: 2.9, 0.2, 0.2	odd n=weak even n=strong	all n=strong
FB	dual subwoofers in phase	s1: 2.9, 0.2, 0.2 s2: 2.9, 6.4, 0.2	odd n=weak even n=strong	odd n=weak even n=strong
OPOC	dual subwoofers in opposite corners and inversed polarity	s1: 0.2, 0.2, 0.2 s2: 5.6, 6.4, 0.2	odd n=strong even n=weak	odd n=strong even n=weak
SSS	dual subwoofers, source-to-sink	s1: 2.9, 0.2, 0.2 s2: 2.9, 6.4, 0.2	odd n=weak even n=strong	see section 2.7
CABS	quad subwoofers, source-to-sink, positioning and DSP	s1: 1.45, 0.2, 1.4 s2: 4.35, 0.2, 1.4 s3: 1.45, 6.4, 1.4 s4: 4.35, 6.4, 1.4	see section 2.5	

the process of applying simple magnitude equalization to the single subwoofer systems improves perceived quality significantly, particularly when applied to “worst” systems (i.e., the corner subwoofer).

### 3 DISCUSSION

In general, there are three tiers of perceived performance established by these results:

1. The configurations associated with the highest quality are based on active removal of energy for the room (i.e., C.A.B.S. and SSS). These are also the most expensive to implement since they require, at least, two subwoofers and a digital signal processing unit. If multiple subwoofers are available, the results obtained here suggest that their most cost effective use is through the application of a simple “source-to-sink” method.
2. The use of simple magnitude equalization improves a poor response significantly and should be attempted when repositioning of loudspeakers is not possible. Reasonable reproduction quality, and a noticeable improvement over a poor response, may also be obtained by simple repositioning of subwoofers. This should preferably be underpinned by basic knowledge of modal theory in small rooms where particular coupling or otherwise to certain modeshapes is attempted. Nevertheless, reliance on techniques that prescribe precise positioning of subwoofers and listeners to avoid modal excitation, such as the OPOC example presented here, are likely to fail due to the unavoidable deviations that occur between theoretical predictions and their real implementation in the physical space.
3. The worst that can be done is to disregard any thought to loudspeaker placement (and listening position). Most likely this will result in a poor low frequency reproduction quality, particularly if the loudspeaker is placed at or near the room corners.

### 4 CONCLUSIONS

A rigorous scientific experiment has been conducted to evaluate the perceived quality of eight low frequency reproduction systems where some of the systems incorporated simple methods to control the unwanted effects of room modes in a standard listening room.

One factor under test was the significance of the nature of music samples used to test for attributes relevant to low frequency reproduction quality. It has been shown that the musical character of the music sample is significant in enabling accurate judgment of modal artifacts. Under the test conditions presented here, a music sample with sufficient low frequency content, enough temporal gaps between notes and a degree of transient and sustained sound supports significant perceptual results. It is granted that rooms and systems are designed to listen to any type of music. However, it is clear that for valid and revealing subjective testing,

particularly if music samples rather than test signals are to be used, carefully selected test musical stimuli are required.

A strong correlation has been demonstrated between the perceived improvement in reproduction quality and the decay times of low frequency energy in the room. This is in line with previous research into aspects of modal perception [7, 9, 12] and corroborates the previous suggestions that modal control methods that are based on direct reduction of modal decays are more likely to achieve a perceptually efficient outcome.

The two systems that achieve decay times below the thresholds identified by Avis et al. [9] are the only ones scoring a highly significant result. It is interesting to note that significant differences between these two systems are not evident, suggesting that once the decays have been controlled to levels below the perceptual threshold, no further treatment might be deemed as effective. The practical implication of this result is that cost savings can be made by targeting thresholds that are based on room response decay times.

One very clear result is that one single subwoofer positioned in the corner of the room, with no equalization, is not advisable. Simple control steps such as moving the subwoofer toward nodal lines of offending modeshapes or applying magnitude equalization will improve reproduction quality noticeably.

Interestingly, perceptual improvements afforded by position control, multiple subwoofers, or magnitude equalization are in general associated with a reduction of decay times for parts of the frequency but may involve a consequent increase in others. This is an interesting result that raises the question of which regions, within the frequency range under study, are more likely to be associated with the largest perceptual improvements when acted upon. This is a topic of current study for the authors.

In contrast to modal decay reductions, a significant perceptual improvement resulting from the direct reduction of frequency response variation is not always evident.

In conclusion, the results obtained show the benefits afforded by simple modal control methods from a subjective standpoint. It appears that, for high quality critical listening conditions, those systems ensuring a faster decay of low frequency energy are preferred over those attempting a direct “flattening” of the magnitude frequency response. These results are generally not surprising, they rather provide the evidence, based on perceptual data, to support the existing good practice in industry.

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Jonathan Hirst worked as a live and sometimes studio sound engineer for 15 years, before joining the University of Salford in 1996 as an undergraduate, enrolling on the B.Sc. (Hons) in audio technology. Since graduating in 1999, he studied toward a Ph.D. in the research area of spatial audio. More specifically, he investigated objective methods of assessing the spatial capabilities of surround sound systems as well as developing spatializing techniques for multichannel musical synthesis. He was awarded a Ph.D. degree in 2006.