IMPLEMENTING WAVE FIELD SYNTHESIS IN AN ITU SPEC LISTENING ROOM PART 2: BASS WITHOUT MODES

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1   INTRODUCTION

Accurate reproduction of sub-bass in small rooms can be problematic because the low modal density and damping over this frequency range can cause different frequency components to be greatly emphasised or de-emphasised. In addition, at frequencies where one mode dominates sound transmission significant spatial variation will be seen in the frequency response, sometimes ranging over as much as 40dB between locations of pressure maximum and minima, meaning that corrective equalisation (EQ) cannot be globally effective. Finally, the low damping of modes also manifests as a long decay time, adding tonal artefacts which are particular audible with transient sounds such as a kick drum. Traditionally these problems have been treated by carefully choosing room dimensions and installing resonant acoustic absorption. However there has recently been significant interest in designing electroacoustic systems which avoid exciting problem room modes. This paper describes design of such a system and evaluates its performance.

2   CASE STUDY DESCRIPTION

The specification was to implement a low frequency reproduction system for critical listening 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[1], BS 6840-13 and IEC 268-13[2] and it was important that the new installation did not affect the effectiveness of the acoustic treatment or the room's accreditation. The dimensions of the room are shown in the diagram below. The acoustic design follows the reflection free zone principle, so there is a great deal of acoustic treatment (indicated grey) installed around head height, hence subwoofers could only be installed on the floor or next to the ceiling.
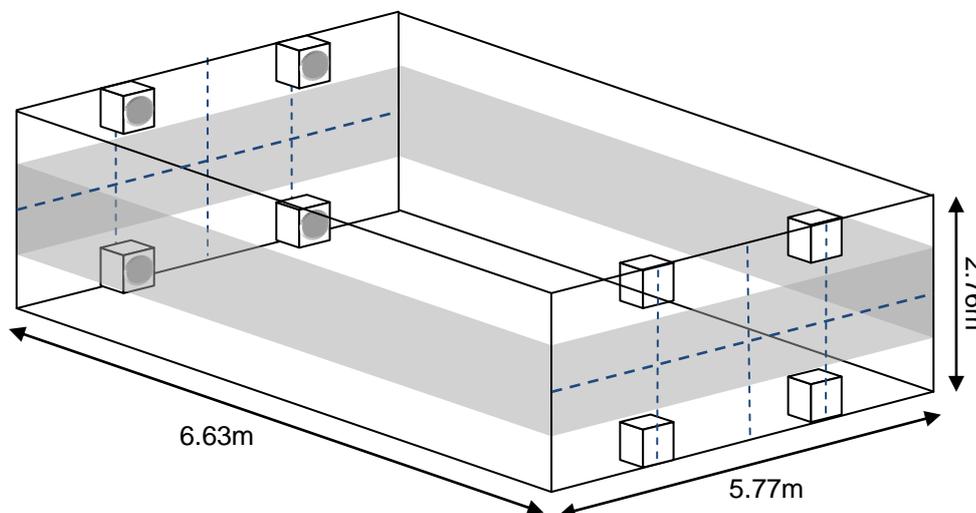

Figure 1: Dimensions of the listening room and layout of the subwoofers

The aim was to achieve a nominally flat frequency response between 25Hz and 120Hz. A Controlled Acoustic Bass System (CABS)[3] subwoofer configuration was chosen since this had already been implemented in the room on a temporary basis and given extremely good modal control. When locating subwoofers in a CABS system the objective is to reduce excitation of cross-modes (up to a certain order) by driving them either at their nulls or in anti-phase so that the 0th order plane wave term is dominant. Symmetrical placement nullifies all odd-order modes, since the mode shape has opposite phase at each subwoofer, so this principle was applied to both horizontal and vertical placement. In addition the horizontal positions were chosen to be ¼ and ¾ across the wall so that all modes with a 2nd order horizontal component would be driven on their nodal lines. The lowest frequency mode not nullified by this configuration is the 4th horizontal axial mode which has a frequency of 188Hz, roughly corresponding to the upper design limit of the system. Modes along the room are not considered here since they will be addressed later using the source / sink principle. It is worth noting that the same frequency limit could have been achieved by locating four subwoofers in the same plan but halfway up the wall as was proposed in the original CABS paper; that solution wasn't possible here due to the practical consideration of leaving the existing acoustic treatment intact and this restriction has effectively doubled the cost of the system by doubling the number of subwoofers required. Such compromises are not described by the CABS notation proposed by Celestinos and Nielsen, which simply counts the front and rear loudspeakers. A notation which gave the highest order of horizontal and vertical modes cancelled by the system might be more informative; e.g. this system would be CABS3.1

Eight identical Genelec 7050B active subwoofers were chosen for the installation. These have a frequency response from 25Hz to 85Hz or 120Hz (selectable) ±3dB. Sensitivity on each unit was set to +12dB dBU for 100dB SPL at 1m, such that four operating together on each wall would output 100dB SPL when input with 0dBU (taken to be the target sensitivity of the system). Since CABS feeds all the loudspeakers on each wall with an identical signal this could arguably have been more economically achieved using a two channel amplifier and eight passive loudspeakers, but this room is part of a research facility and it was seen as beneficial to support the option to control each subwoofer individually. Accordingly all the subwoofers are feed by separate signal cables which terminate in a small patch bay, where the two signals from the DSP unit (a dbx Driverack 220i) are split and fed to four subwoofers each.

3   BENCHMARK MEASUREMENTS

To assess the effectiveness of the CABS system two other configurations were measured. All measurements were performed using a B&K Pulse type 3560B Front End with four type 4165 (or equivalent) ½" microphone capsules and type 2269 (or equivalent) preamps. These were laid out in an asymmetrical configuration (as symmetrical measurements of modal behaviour are redundant) with height ranging from 1.0m to 1.6m in 0.2m intervals. Harmonic distortion was not deemed to be a concern since the loudspeakers were measured at low output level, so pink noise excitation and the cross-power spectral density method were used to find the frequency response functions.

Welti[4] identified that spatial uniformity is critical to achieving an accurate bass sound, since frequency response correction EQ cannot be globally effective if different locations experience significantly different SPLs for the same frequency. He proposed a metric to measure this property and plotting this along with the individual microphone measurements aids insight and allows system configurations with spatially uniform, and so correctable, frequency responses to be easily identified. However whereas Welti performed his statistics on the SPL in dB, metrics in this paper will be calculated on pressure magnitude squared. This is well justified for the mean response, since this in the power average and can be considered an estimate of the sound power present in the room irrespective of listener position[5]. It is defined by:

$$\mu(f) = \sqrt{\underset{m \in mics}{\mathrm{mean}} |p_m(f)|^2}$$

The standard deviation was also calculated based on the pressure magnitude squared:

$$\sigma(f) = \sqrt{\underset{m \in mics}{\text{mean}} ||p(f)|^2 - |\mu(f)|^2|^2}$$

This has the attractive property that the confidence limits always lie within the range of measured responses; this is not the case for dB since notches in the frequency response are emphasised in a log scale and skew the results. 67% confidence limits are found by adding or subtracting the standard deviation $\sigma(f)$ from the mean $\mu(f)$ of the frequency responses. These statistics are then converted to dB and plotted alongside the raw data.

3.1   One subwoofer in corner

First the worst-case scenario of one subwoofer located in a corner was measured. The frequency response is given in Figure 2 and can be seen to be very uneven, even the power average varies by more than 20dB over the operating frequency range, with a great number of resonant peaks which are audible as tonal artefacts. Problem frequencies can be associated with the following modes (where the bracketed numbers indicate the mode order as [N_L N_W N_H]): 26Hz [1 0 0], 28Hz [0 1 0], 37.5Hz [1 1 0], 52Hz [2 0 0], 59-61Hz [0 2 0], [2 1 0], [0 0 1] and 75-80Hz [3 0 0], [2 2 0] and [2 0 1]. The range of the confidence limits is quite large, indicating that reproduced sound will have frequency components which are significantly emphasised or deemphasised differently depending on where you are in the room, hence global correction EQ cannot be applied. Figure 3 shows the same measurement as an impulse response. The slow decay rate of -40dB/s will have a detrimental effect on the definition of the bass sound in the room and, given that it is associated is associated with high Q peaks in the frequency response, the tail is also lightly to be tonal in nature and highly audible. These are not qualities that are desirable in a critical listening environment.
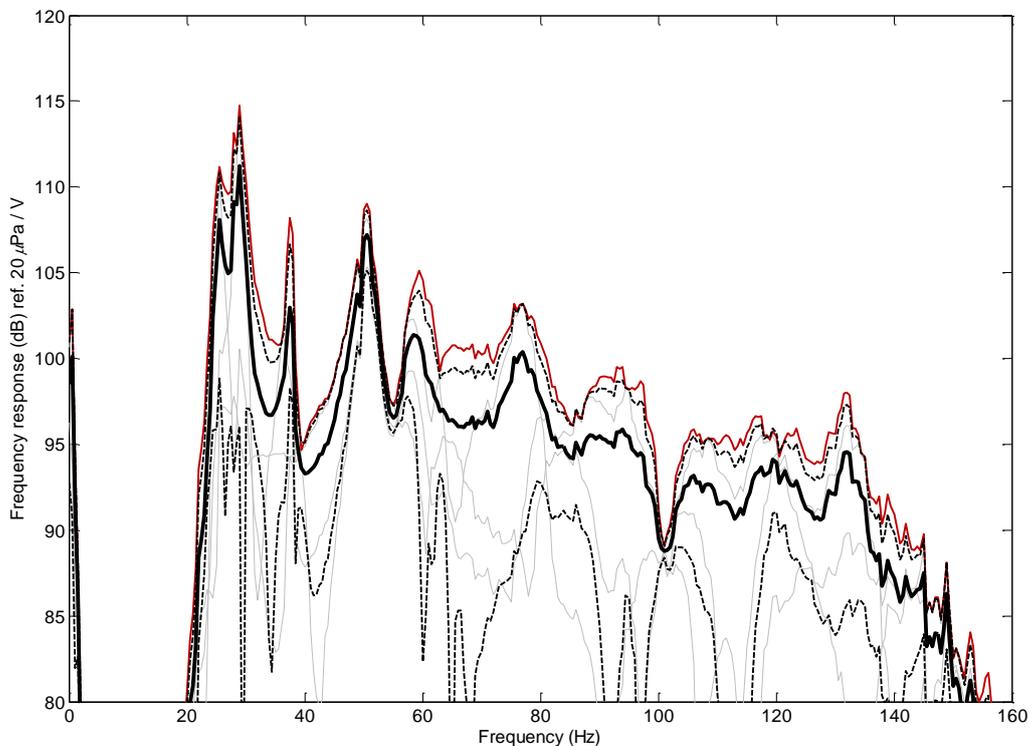

Figure 2: Frequency response of a single subwoofer located in a corner. The grey lines are individual measurements at 4 microphone positions. The maximum measured pressure is plotted in red and the power average in thick black. The dashed black lines are the 67% confidence limits.
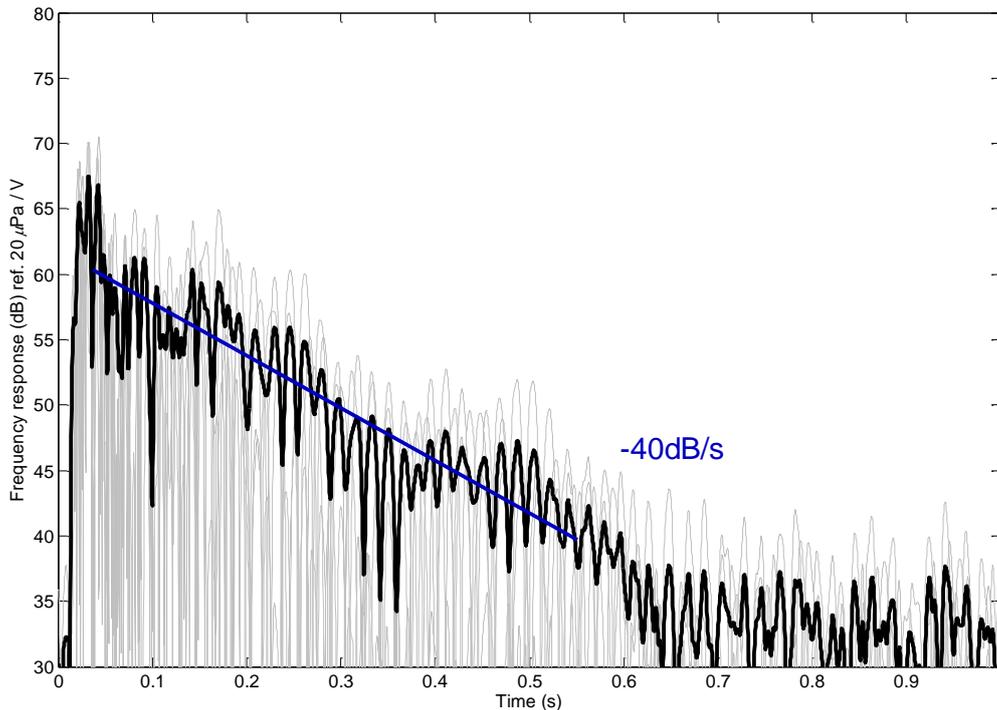
Figure 3: Impulse response of a single subwoofer located in a corner. The grey lines are individual measurements at 4 microphone positions, the thick black line is their power average and the blue line is the approximate decay trend with its gradient written in blue.

3.2 All installed subwoofers in phase

As a second benchmark scenario the system was measured with all the installed subwoofers being fed the same signal in phase. This was in principle quite similar to some of the systems that Welti favoured except that less attention has been paid to the control of lengthways modes. The frequency response is shown in Figure 4. The response is much smoother than for the single subwoofer, but a large resonance is present at 51Hz where the subwoofers all drive the second lengthways axial mode in phase. Interestingly the standard deviation of pressure squared across the measurement positions is seen to be very low all through this region up to approximately 80Hz, meaning that corrective EQ could in principle be applied. However the Q of this peak is quite high, greater than 16 so above the audibility threshold[7], and care must be taken as slightly mismatched corrective EQ could cause audible artefacts. Above 80Hz the response is less uniform due to the increasing modal density and the standard deviation increases, however the power average is still quite flat. Figure 5 shows the same measurement as an impulse response. The decay rate has been improved to -80dB/s, however the response tail is dominated by the 52Hz resonance (demonstrated by the uniform decay and regular spacing of approximately 0.01seconds between amplitude peaks) so is lightly to be discernable on transient sounds such as kick drums. Corrective system EQ would reduce the excitation of this decay term but it would still decay more slowly than other frequencies, resulting in a dual-slope decay trend the bottom part of which would be tonal and could be audible at high listening SPLs. Increasing the modal damping to reduce the decay time and then applying corrective EQ to fine tune the system is a more robust approach likely to give better overall results and this will be implemented in the next section.
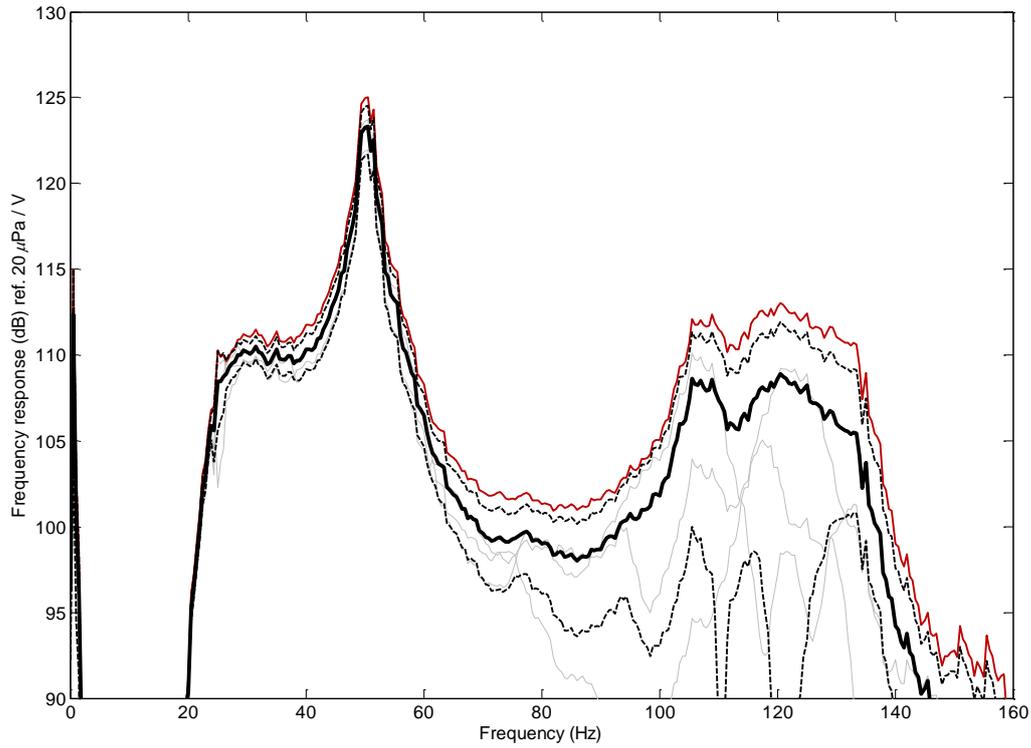
Figure 4: Frequency response of all installed subwoofers operating in phase. The grey lines are individual measurements at 4 microphone positions. The maximum measured pressure is plotted in red and the power average in thick black. The dashed black lines are the 67% confidence limits.
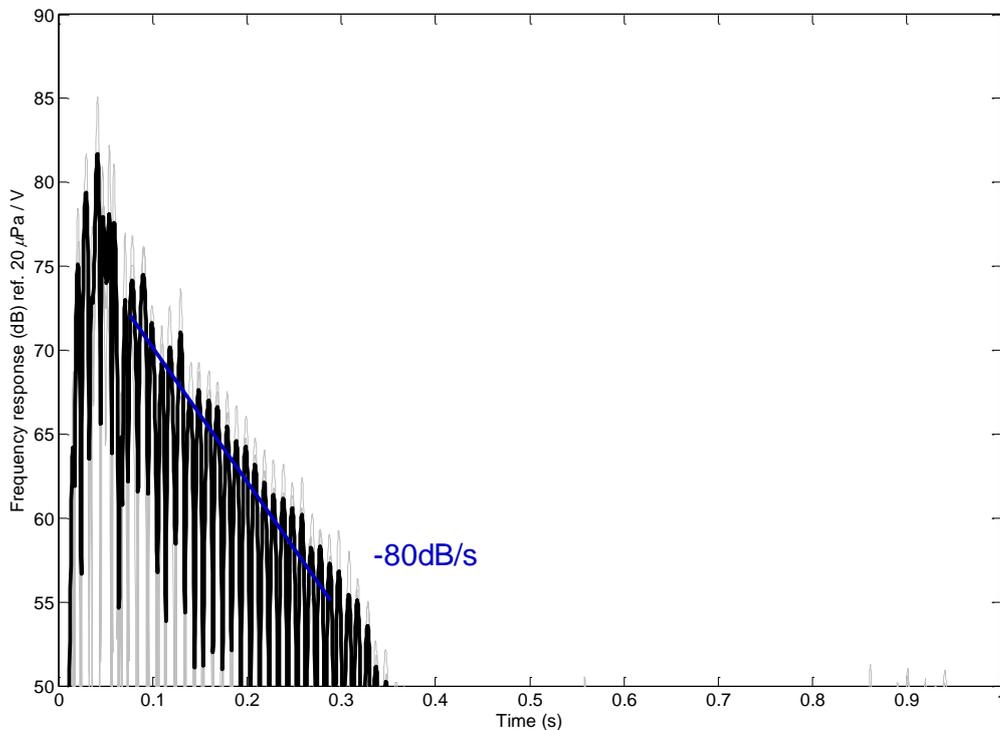

Figure 5: Impulse response of all installed subwoofers operating in phase. The grey lines are individual measurements at 4 microphone positions, the thick black line is their power average and the blue line is the approximate decay trend with its gradient written in blue.

4 CONTROLLED ACOUSTIC BASS SYSTEM (CABS)

Performance of a CABS implementation using the hardware described above will now be evaluated. The system uses the set of subwoofers on the front wall as sources and the subwoofers on the rear wall as sinks. Phase reversal, propagation delay and attenuation (to compensate for absorption in the room) are applied to the rear channel to minimise the reflection off the rear wall so that a quasi-anechoic termination arises and lengthways modes cease to exist. The crux of the scheme is therefore the concept of launching a travelling plane wave down the room and absorbing it at the back wall, and in this sense the system performs Wave Field Synthesis (WFS). The usual limit that transducers must be spaced less than half a wavelength apart applies horizontally and vertically, however the system is able to use far fewer transducers than a typical WFS array by exploiting the array of image sources that reflections from the room boundaries provide.

4.1 Choosing the CABS parameters

The first step is to choose the CABS parameters, being the propagation delay T and the rear attenuation a. Celestinos and Nielsen provide a simple analytical formula for the former and suggest that the latter is chosen empirically. For the listening room under study the propagation delay T can be evaluated by dividing the length of the room 6.63m by the speed of sound in air:

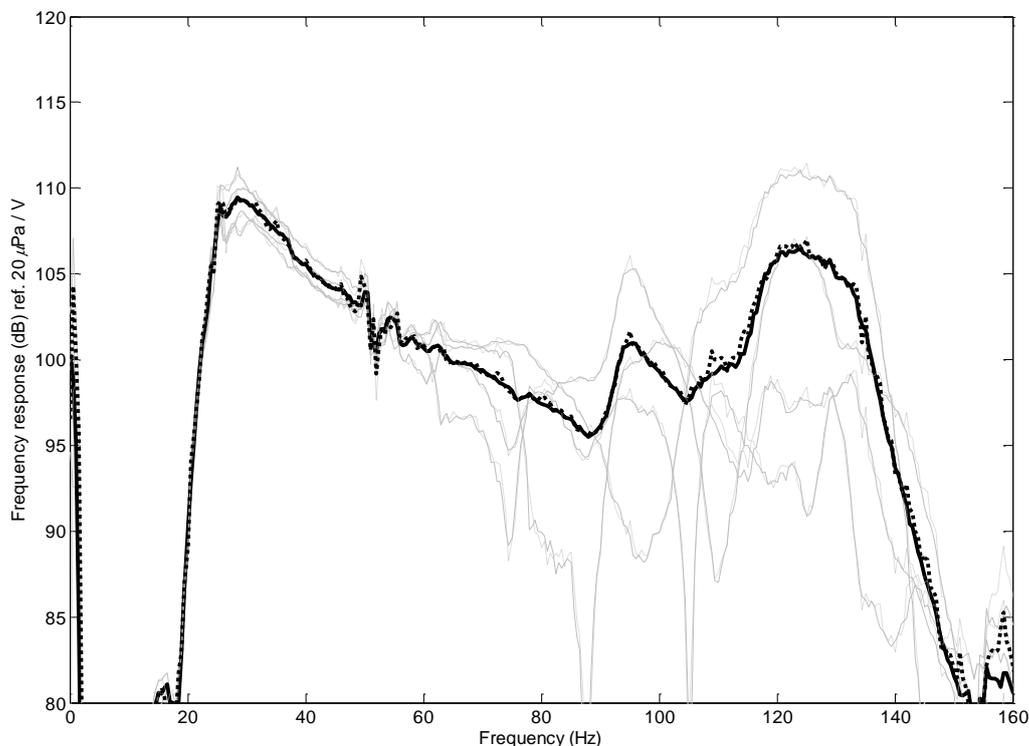$$T = L/c = 6.63\text{m}/343\text{ms}^{-1} = 19.3\text{ms}$$

Figure 6: Frequency response of the CABS system with a delay of 19.5ms and attenuation of -2.0dB. Measured (solid lines) versus simulation (dotted lines) using numerical combination of separate measurements of front and rear loudspeakers. The thin grey lines are individual measurements at 4 microphone positions and the thick black line is the power average.

It would be desirable to have an automated means of finding the optimum value for a. To this end a numerical algorithm was designed which combines individual measurements of the front and rear subwoofers (H_f and H_r) with the delay and attenuation parameters to predict the total response H:

$$H(f) = H_f(f) - aH_r(f)e^{i2\pi fT}$$

Some may raise concern that this does not adequately take into account non-linear effects, such as radiation loading on the subwoofers, but Figure 6 compares a measured response of the entire system with one predicted using the above numerical method and no significant error occurs in the operating frequency range.

This simulation approach allows the delay and attenuation parameters to be numerically optimised. Crucial to this is selection of an appropriate cost function which penalises configurations which give a rapidly varying frequency response or long decay time. Various cost functions were tried and their performance evaluated. The standard deviation measure described in section 3 was an obvious choice, but as seen in Figure 4 it does not penalise even modes which give fairly uniform coverage over the listening area. Testing the 2nd derivative of the spectrums with respect to frequency was expected to be able to identify narrow peaks, but turned out to be highly susceptible to background noise. Ultimately a simple cost function which measured the mean deviation in dB from a flat response for all microphones proved to be most robust:

$$p_{\text{target}} = \underset{25<f<120}{\text{mean}} 20\log_{10}\mu(f) \qquad \text{Cost} = \underset{25<f<120}{\text{mean}} \underset{m\in\text{mics}}{\text{mean}} \left|20\log_{10}p_m(f) - p_{\text{target}}\right|$$
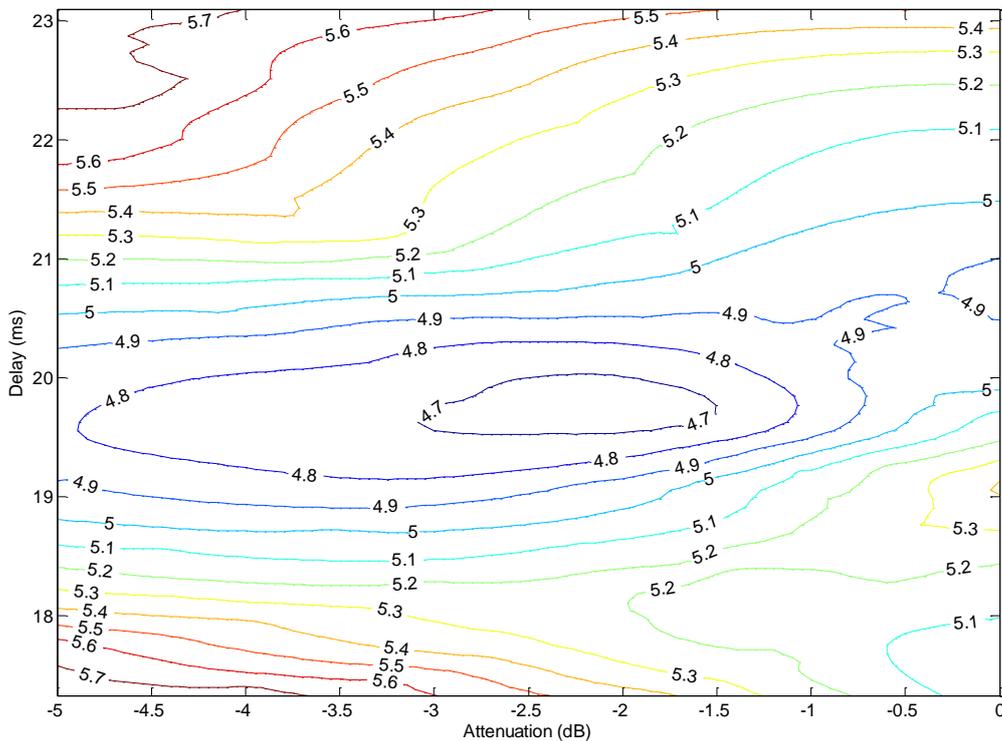

Figure 7: "Cost" penalty for frequency responses arising from a range of combinations of rear channel delay and attenuation.

A minimum is present between -1.5dB and 3dB attenuation and 19.4ms to 19.9ms delay, suggesting the parameters be chosen in these ranges. The delay range is slightly longer than was predicted by the analytical formula, but this could be an idiosyncrasy of the rather simplistic cost function used. The shape of the minima is interesting though, since it indicates that the

performance of the system is more dependent on delay time than attenuation, supporting Celestinos and Nielsen's suggestion that delay should be calculated analytically and attenuation tuned empirically. In this installation fine tuning of both parameters was performed manually while repeatedly re-measuring the frequency response and a delay of 19.5ms and an attenuation of -2.0dB were chosen, giving the frequency response already shown in Figure 6. This was the best compromise as other values resulted in small peaks around the first and second lengthways modal frequencies (26Hz and 52Hz). It is worth noting that the system response inherits the low spatial variation seen in Figure 4 when all the subwoofers were driven in phase, but without the large 52Hz resonance. The -6dB/oct spectral slope typical in small rooms at LF is clearly evident.

4.2   The installation

Having achieved the relatively smooth and uniform frequency response in Figure 6 corrective EQ can now be applied. The key concern here is to avoid applying any high-Q boosts which may themselves introduce tonal artefacts or beating. Notches in the frequency response are also deemed more acceptable than peaks since they do not introduce tonal artefacts or long decays and are less perceptible to the human ear under broadband stimuli. The ⅓ octave graphic EQ module of the DSP unit was used to perform broad spectral shaping and the parametric EQ used in cut only to smooth those significant peaks that remained. The settings used were as follows:

Graphic EQ:

Freq.	31.5Hz	40.0Hz	50.0Hz	63.0Hz	80.0Hz	100Hz	125Hz
Gain	-6.0dB	-3.5dB	-1.0dB	+0.0dB	+3.0dB	+5.0dB	-4.0dB

Parametric EQ:

Freq.	26.5Hz	96.4Hz	121Hz
Q	5.02	12.4	9.55
Gain	-5.0dB	-4.0dB	-3.0dB

The measured frequency response of the final commissioned system is shown in Figure 8. Extra microphone measurements were made giving a total of 16 positions around the room; two sets of four distributed as previously described, plus one smaller cluster close to the centre of the room and one quite close to the walls. Spectral and spatial uniformity is excellent from 30Hz to 70Hz, with all measurements well within ±3dB of 100dB SPL per dBU. It is worth noting that the small peak just below 25Hz only occurs for the microphones near the edge of the room and if these are removed from the statistics (just leaving the microphones in the typical listening area) then the response stays within the green limits right down to the -3dB crossing at 22Hz. 76Hz is the first significant problem frequency where a notch occurs, though again this predominantly affects microphones located towards the edge of the room. Above 85Hz irregularities start to occur which strongly affect the response in the typical listening area and this is reflected by a larger confidence interval, though the power average, upper confidence limit and maximum stay within ±3dB and ±6dB of 100dB SPL per dBU respectively. A dip at 88Hz was tolerated since corrective EQ would require a high Q boost. The power average drops back below the -3dB limit at 136Hz. It may therefore be concluded that within the central listening area the system produces 100dB SPL per dBU ±3dB within 22Hz to 85Hz, and nominally 100dB SPL per dBU ±6dB from 85Hz to 136Hz. It is interesting to note that the CABS configuration appears to have effectively extended the frequency response of the subwoofers beyond the manufacturers stated -3dB limits in free field without electronic gain being required (in fact the DSP unit is applying attenuation at these frequencies so gives additional headroom). Figure 9 shows the same measurements as an impulse response. The decay is impressively quick, equivalent to 60dB drop in 0.3s, which matches the room's reverb time at higher frequencies. Subjectively the system sounded tight and deep but quite bass light. Recent studies at Salford[7] have suggested that there is an expectation for some degree of resonance at low frequencies and complete elimination of this is not favoured by listeners. However from a critical listening perspective it is surely preferable to start with a neutral (flat) system available and then choose to add bass energy electronically if desired.
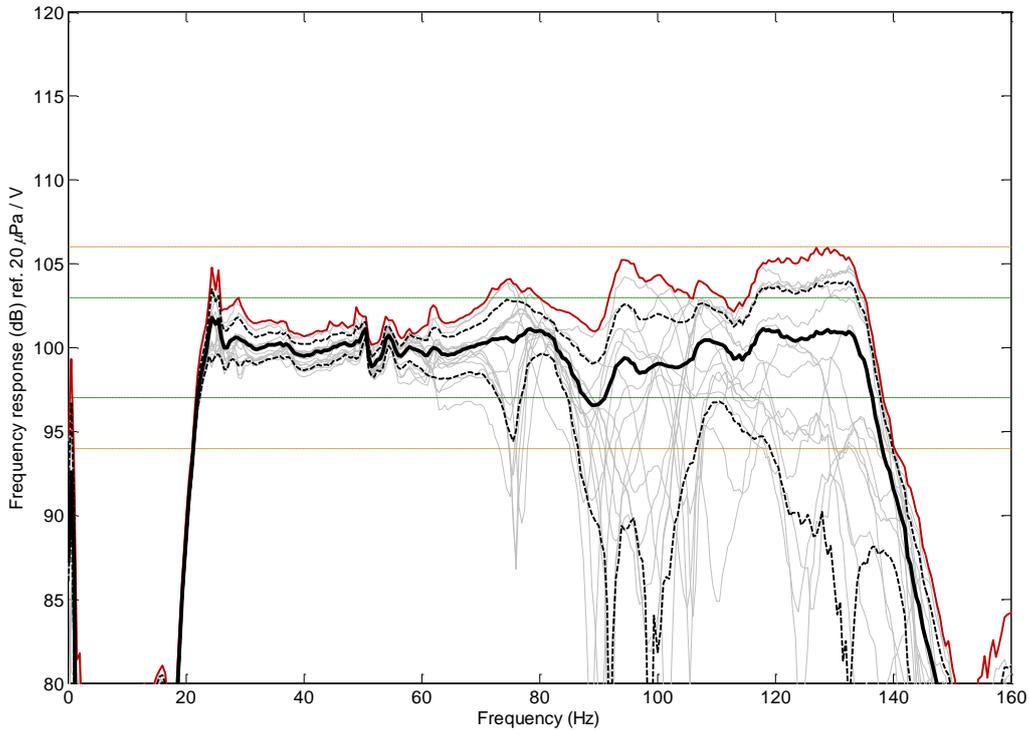
Figure 8: Frequency response of the installed CABS. The grey lines are individual measurements at 16 microphone positions. The maximum measured pressure is plotted in red and the power average in thick black. The dashed black lines are the 67% confidence limits and the dashed green and orange lines are 100dB SPL per dBU ±3dB and ±6dB respectively.
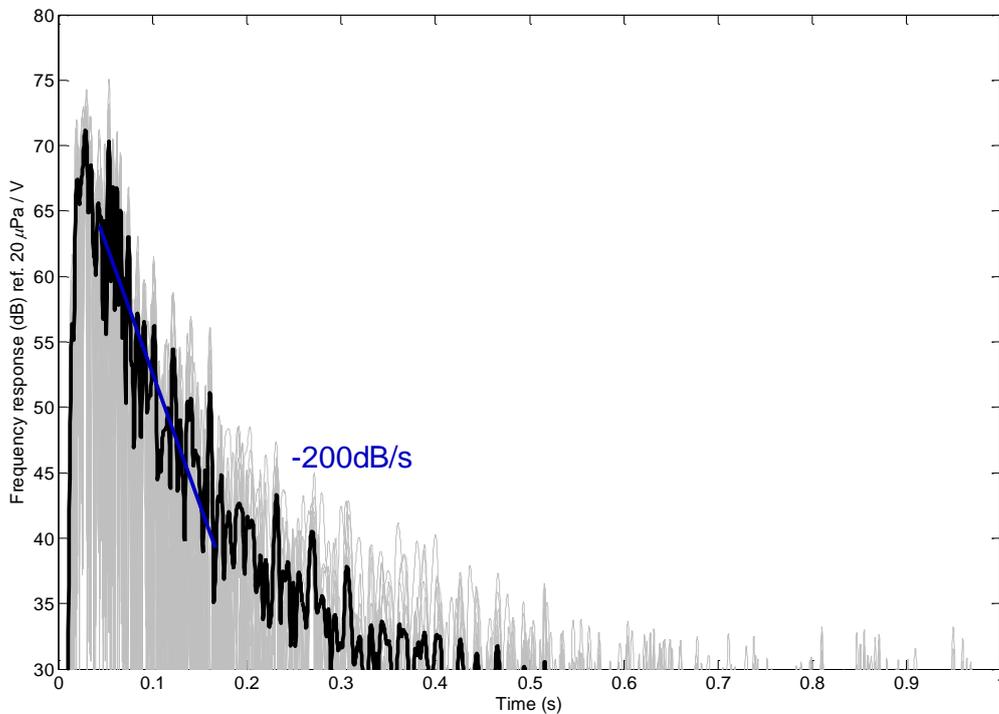

Figure 9: Impulse response of the installed CABS. The grey lines are individual measurements at 16 microphone positions, the thick black line is their power average and the blue line is the approximate decay trend with its gradient written in blue.

5 CONCLUSIONS

A low-frequency sound reproduction system was installed in a critical listening room following CABS design principles. Eight subwoofers were located on the floor and ceiling to avoid modification of existing acoustic treatment around the room which could affect the room satisfying the ITU specification. The optimal choice of delay and attenuation parameters was found by minimising a cost function designed to penalise deviation from a flat frequency response. The equalised system produced 100dB SPL per dBU ±3dB between 22Hz and 85Hz in the central listening area and ±6dB close to the walls and up to 136Hz. The transient response was excellent with a decay equivalent to 60dB in 0.3 seconds. Further research could involve designing better optimisation cost functions, investigating if frequency dependent rear channel attenuation or individual subwoofer control could give even better results, and investigating whether the CABS source / sink principle could be useful for tackling room reflections when rendering waves with WFS at higher frequencies

6 REFERENCES

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