

Automated In-situ Frequency Response Optimisation of Active Loudspeakers

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ABSTRACT

This paper presents a novel method for robust automatic selection of optimal in-situ acoustical frequency response within a discrete-valued set of responses offered by room response controls on an active loudspeaker. A frequency response measurement is used as the input data for the algorithm. The rationale of the room response control system is described. The response controls are described for each supported loudspeaker type. The optimisation algorithm is described. Examples of the optimisation process are given. The efficiency and performance of the algorithm are discussed. The algorithm dramatically improves the speed of optimisation compared to an exhaustive search. It improves the acoustical similarity between loudspeakers in one space and performs robustly and systematically in widely varying acoustical environments. The algorithm is currently in active use by specialists who set up and tune studios and listening rooms.

1. INTRODUCTION

This paper presents a system to optimally set the room response controls currently found on full-range active loudspeakers to achieve a desired in-room frequency response.

The active loudspeakers [1] to be optimised are designed and calibrated in anechoic conditions to have a flat frequency response magnitude within the design limits of ± 2.5 dB. When a loudspeaker is placed into

the listening environment, response changes due to the loudspeaker-room interaction. To help alleviate this, these active loudspeakers incorporate a pragmatic set of room response controls accounting for some common acoustic issues found in professional listening rooms.

Although many users have the facility to measure loudspeaker in-situ frequency responses, they often do not have the experience of calibrating active loudspeakers. Even with experienced system calibrators a

significant amount of variance between calibrations can be seen. With a number of different people calibrating loudspeaker systems there will be an additional variance in results. For these reasons a method to ensure consistency of calibrations is required.

Presented first in this paper is the discrete-valued room response equalizer employed in the active loudspeakers. Then, the algorithm for automated value selection is presented. This includes software structure, algorithm, features and operation. The performance of the optimisation algorithm is then investigated with case studies. Finally, limitations of the acoustic measurement system, room response controls and the algorithm are discussed together with the case study results.

2. IN-SITU EQUALISATION AND ROOM RESPONSE CONTROLS

2.1. Equalisation Techniques

The purpose of room equalisation is to improve the perceived quality of sound reproduction in a listening environment. The goal of equalisation is usually not to convert the listening room to anechoic. In fact, listeners prefer to hear some room response in the form of liveliness that can create a spatial impression and some envelopment [2]. Electronic equalisation to improve the subjective sound quality has been widespread for at least 40 years; see Boner & Boner [3] for an early example. Equalisation is particularly prevalent in professional sound reproduction applications such as mixing rooms and sound reinforcement.

The room transfer function is position dependent, which poses major problems for all equalisation techniques. Perfect equalisation within a reasonably large listening area appears not to be possible, and even an acceptable equalisation is typically a compromise. Cox and D'Antonio [4] (Room Optimiser) use a computer model of the room to find optimal loudspeaker positions and acoustical treatment location to give an optimally flat in-situ frequency response magnitude. Positional areas for the loudspeaker and listening locations can be given as constraints to limit the final solution. Despite advances in psychoacoustics, it is difficult to quantify how good the listener actually perceives the sound quality to be, and to optimise equalisation based on that evaluation [5-7]. Also, despite the widespread use of equalisation, it is still difficult to provide exact timbre matching between different environments.

In-situ response equalisation is typically implemented using a separate equaliser. Some equalisers on the market play a test signal and then alter their response

according to the in-situ transfer function measured in this way [8] but the process can be so sensitive that a simple 'press the button and everything will be OK' approach proves hard to achieve with reliability, consistency and robustness.

It is possible that equalisation becomes skewed if it is based only on a single point measurement. The frequency response in nearby positions can actually become worse after the equalisation designed using only a single point measurement is applied. A classical method to avoid this is to use a weighted average of responses measured within the listening area. Such spatial averaging is often required when the listening area is large. Spatial averaging can reduce local variance seen in the midrange to high frequencies and can also reduce problems caused by the fact that a listener perceives sound differently to a microphone. Examples of spatial averaging have been described in the automotive industry [9] and cinema in the SMPTE Standard 202M [10].

When using one loudspeaker, no correction filter is capable of reducing the difference between responses measured at two separate receiver points. At high frequencies a high-resolution correction can be very position sensitive. Frequency dependent resolution change becomes preferable and is typically applied [11,12].

Traditionally, electronic equalisation uses arrangements of analogue low order minimum phase filters [13-15]. Since the loudspeaker-room transfer function is of substantially higher order than such equalisation filters, the effect of filtering is to gently shape the response. Several methods have been proposed for more exact inversion of the frequency response to achieve a close approximation of unity transfer function (no change to magnitude or phase) within a certain bandwidth of interest [16-23]. Some researchers have also shown an interest to control selectively the temporal decay characteristics of a listening space by active absorption or modification of the primary sound [24-29]. If realisable, these are extremely attractive ideas because they imply that the perceived sound could be modified with precision, to different target responses. One of the major problems is that spatial variations in the frequency response can become far more difficult to handle than with low-order methods because the correction depends strongly on an exact match between the acoustic and equalization transfer functions, and can therefore be highly local in space [30].

2.2. Room Acoustic Considerations

In small to medium sized listening environments, the sound field in the frequency range up to a critical

frequency f_c (typically 70...200 Hz in small spaces) is often dominated by room modes and comb filtering caused by low-order discrete reflections from room boundaries. Sound reproduction can be problematic because of this. For a room with a reverberation time T_{60} of 0.3 s the room mode bandwidth is approximately $2.2/T_{60} = 7.3$ Hz [23]. However, this does not predict accurately what the decay rate of an individual mode is as reverberation time represents the total decay rate in diffuse field whereas modal decay rate may vary.

Above f_c modal density becomes sufficiently high to be described statistically. An unsmoothed room transfer function shows a large number of high Q notches. When frequency smoothing due to human hearing is taken into account [31], the resulting sensation is a rather smooth room transfer function (Figure 3 and Figure 6).

In the time domain, early reflections before about 25 ms combine with the direct sound to produce tone colouration (comb filtering effect). Reflections arriving later than about 25 ms are less problematic as they typically combine to produce the reverberation of the room and are perceived as separate sound events (echoes and reverberation) rather than tone colouration. This part of the time domain response contributes to the sensations of envelopment and spaciousness.

2.3. Room Response Controls

The loudspeakers to be optimised have room response controls [1,32]. The smaller loudspeakers have simpler controls than the larger systems but the philosophy of filtering is consistent across the range (Tables 1-4).

The **treble tilt control** is used to reduce the high frequency energy. In the small two-way systems and two way systems it is a level control of the treble driver and has an effect down to about 4 kHz. In large systems it has a noticeable effect only above 10 kHz and has a roll-off character.

The **driver level controls** can be used to shape the broadband response of a loudspeaker. They control the output level of each driver with frequency ranges that are determined by the crossover filters.

The **bass tilt control** compensates for a bass boost seen when the loudspeaker is loaded by large nearby boundaries [33-36]. This typically happens when a loudspeaker is placed next to, or mounted into, an acoustically hard wall. This filter is a first order shelving filter.

The **bass roll-off control** compensates for a bass boost often seen at the very lowest frequencies the

loudspeaker can reproduce. This typically happens when the loudspeaker is mounted in the corner of a room where the loudspeaker is able to couple very efficiently to the room thereby exacerbating room mode effects that dominate this region of the frequency response. It is a notch filter with a centre frequency set close to the low frequency cut-off of the loudspeaker.

Table 1. Small two way room response controls.

Control type	Room response control settings, dB
Treble tilt	0, -2
Bass tilt	0, -2, -4, -6
Bass roll-off	0, -2

Table 2. Two way room response controls.

Control type	Room response control settings, dB
Treble tilt	+2, 0, -2, -4, driver mute
Bass tilt	0, -2, -4, -6, driver mute
Bass roll-off	0, -2, -4, -6, -8

Table 3. Three way room response controls.

Control type	Room response control settings, dB
Treble level	0, -1, -2, -3, -4, -5, -6, driver mute
Midrange level	0, -1, -2, -3, -4, -5, -6, driver mute
Bass level	0, -1, -2, -3, -4, -5, -6, driver mute
Bass tilt	0, -2, -4, -6, -8
Bass roll-off	0, -2, -4, -6, -8

Table 4. Large system room response controls.

Control type	Room response control settings, dB
Treble tilt	+1, 0, -1, -2, -3
Treble level	0, -1, -2, -3, -4, -5, -6, driver mute
Midrange level	0, -1, -2, -3, -4, -5, -6, driver mute
Bass level	0, -1, -2, -3, -4, -5, -6, driver mute
Bass tilt	0, -2, -4, -6, -8
Bass roll-off	0, -2, -4, -6, -8

3. ROOM EQUALISATION OPTIMISER

Optimisation involves the minimisation or maximisation of a scalar-valued objective function $E(x)$,

$$\min E(x) \quad (1)$$

where, x is the vector of design parameters, $x \in \mathcal{X}^n$. Multi-objective optimisation is concerned with the minimisation of a vector of objectives $E(x)$ that may be subject to constraints or bounds. Several robust methods exist for optimising functions with design parameters x having a continuous value range [37].

3.1. Efficiency of Direct Search

The room response controls of an active loudspeaker form a discrete-valued set of frequency responses. If the optimum is found by trying every possible combination of room response controls then the number of processing steps becomes prohibitively high (Table 5).

Table 5. Number of setting combinations.

Room Response Control	Type of loudspeaker			
	Large	3-way	2-way	Small 2-way
Treble tilt	5	-	4	2
Treble level	7	7	-	-
Midrange level	7	7	-	-
Bass level	7	7	-	-
Bass tilt	5	5	4	4
Bass roll-off	5	5	5	2
Total	42875	8575	80	16

3.2. The Algorithm

The algorithm exploits the heuristics of experienced system calibration engineers by dividing the optimisation into five main stages (Table 6), which will be described in detail. The optimiser considers certain frequency ranges in each stage (Table 7). Figure 9 in Appendix A shows a flow chart of the software. A screenshot of the software graphic user interface can be seen in Appendix B.

Table 6. Optimisation stages.

Optimisation stage	Type of loudspeaker			
	Large	3-way	2-way	Small 2-way
Preset bass roll-off	✓	✓	✓	✓
Find midrange/treble ratio	✓	✓	-	-
Set bass tilt and level	✓	✓	-	-
Reset bass roll-off	✓	✓	✓	✓
Set treble tilt	✓	-	✓	✓

Table 7. Optimiser frequency ranges; $f_{HF} = 15$ kHz; f_{LF} is the frequency of the lower -3 dB limit of the frequency range.

	Frequency Range Limit	
	Low	High
Loudspeaker pass band	f_{LF}	f_{HF}
Midrange and treble driver band	500 Hz	f_{HF}
Bass roll-off region	f_{LF}	$1.5 f_{LF}$
Bass region	$1.5 f_{LF}$	$6 f_{LF}$

3.2.1. Pre-set Bass Roll-off

In this stage, the bass roll-off control is set to keep the maximum level found in the ‘bass roll-off region’ as close to the maximum level found in the ‘bass region’. Once found the bass roll-off control is reset to one position higher, for example, -4 dB is changed to -2 dB. The reason for this is to leave some very low bass energy for the bass tilt to filter. It is possible that the bass tilt alone is sufficient to optimise the response and less or no bass roll-off is eventually required. The min-max type objective function to be minimized is given by Equation 2,

$$\min_m E = \left| \frac{\max_{f_a} \left(\frac{a_m(f)x(f)}{x_0(f)} \right)}{\max_{f_b} \left(\frac{a_m(f)x(f)}{x_0(f)} \right)} \right|, \quad (2)$$

$$f_a = [f_1, f_2], \quad f_b = [f_2, f_3]$$

where $x(f)$ is the smoothed magnitude of the in-situ frequency response of the system, $a_m(f)$ is the bass roll-off setting m currently being tested, $x_0(f)$ is the target response, f_a defines the ‘bass roll-off region’ (Table 7) and f_b defines the ‘bass region’ (Table 7). User selected frequency ranges are not permitted.

The reason for this arrangement rather than using a least squares type objective function is that the bass roll-off tends to assume maximum attenuation to minimise the rms deviation. This type of objective function does not yield the best setting, as subjectively a loss of bass extension is perceived.

This stage of the optimiser algorithm takes six filtering steps (three for small two-way models).

3.2.2. Midrange Level to Treble Level Ratio

The aim of this stage is to find the relative levels of the midrange level and treble level controls required to get closest to the target response. The least squares

type objective function to be minimised is given in Equation 3,

$$\min_m E = \int_{f=f_1}^{f_2} \left| \frac{a_m(f)x(f)}{x_0(f)} \right|^2 df \quad (3)$$

where $x(f)$ is the smoothed magnitude of the in-situ frequency response of the system, $a_m(f)$ is the midrange and treble level control combination m currently being tested, $x_0(f)$ is the target response, f_1 and f_2 define the ‘midrange and treble driver band’ (Table 7). The lower frequency bound is fixed at 500 Hz but a user selectable high frequency value is permitted. The default value is 15 kHz.

The midrange-to-treble level ratio is saved for performing the third stage of the optimisation process. The reason for this is to reduce the number of room response control combinations to be tested in the next stage.

This stage of the optimisation algorithm takes 49 filtering steps and is not required for two-way models or small two-way models.

3.2.3. Bass Tilt and Bass Level

This stage of the optimiser algorithm filters using all possible combinations of bass tilt and bass level controls for a given midrange/treble level difference. By fixing this difference the total number of filter combinations can be reduced substantially.

A constraint imposed in this stage is that only two of the driver level controls can be set at any one time. If three of the level controls are simultaneously set the net effect is a loss of overall system sensitivity. Table 8 shows an example of incorrect and correct setting of the driver level controls.

Table 8. Driver level control settings.

Control	Incorrect Setting	Correct Setting
Bass level	-4 dB	-2 dB
Midrange level	-3 dB	-1 dB
Treble level	-2 dB	0 dB
Input sensitivity	-6 dBu	-4 dBu

The least squares type objective function to be minimised is the same as shown in Equation 3. However, $a_m(f)$ is the bass tilt and bass level combination m currently being tested together with the fixed midrange and treble level ratio setting found in the previous stage. Also, f_1 and f_2 now define the ‘loudspeaker pass band’ (Table 7). High and low user selected frequency values are permitted. The default

values are the -3 dB lower cut-off frequency of the loudspeaker and 15 kHz.

This part of the optimisation algorithm takes 35 filtering steps. There are no driver level controls in two-way or small two way systems so these virtual controls are set to 0 dB. The bass tilt control can then be optimised using the same objective function. Only five filtering steps are required for two-way and small two-way systems.

3.2.4. Reset Bass Roll-off

Firstly, the bass roll-off control is reset to 0 dB. Then the same method used to set the bass roll-off earlier is repeated, but without modifying upwards the final setting. The same objective function is used as presented in Section 3.2.1.

3.2.5. Set Treble Tilt

The least squares type objective function to be minimised is the same as shown in Equation 3. However, f_1 and f_2 now define the ‘loudspeaker pass band’ (Table 7). High and low user selected frequency values are permitted. The default values are the -3 dB lower cut-off frequency of the loudspeaker and 15 kHz.

This part of the algorithm requires five filtering steps for two way and large models (three for small two way models) and is skipped for three ways because they do not have this control.

3.3. Reduction of Computational Load

The optimiser algorithm has been designed to reduce the computational load by exploiting the heuristics of experienced calibration engineers. The resulting number of filtering steps has been dramatically reduced for the larger systems (Table 9) and even the relatively simple two-way systems show a substantial improvement when compared to the number of filtering steps needed by direct search method as summarised in Table 5. There are two main reasons for the improvement; the constraint of not allowing the setting of all three of the driver level settings simultaneously and the breaking up of the optimisation into stages.

The run time on a PII 366 MHz computer for a three-way system is about 15 s (direct search 3 minutes). Large systems now take about the same time as a three-way system (predicted direct search time was 15 minutes). The processing time is directly proportional to the processor speed as a PIII 1200 MHz based computer takes about 4 s to perform the same optimisation. Further changes in the software have improved these run times by about 30%.

Table 9. Number of filter evaluations needed by the optimisation algorithm.

Optimisation stage	Type of loudspeaker			
	Large	3-way	2-way	Small 2-way
Preset bass roll-off	6	6	6	3
Find midrange/treble ratio	49	49	-	-
Set bass tilt and level	35	35	5	5
Reset bass roll-off	6	6	6	3
Set treble tilt	5	-	4	2
Total	101	96	21	13
Total re. direct search	0.2%	1.1%	26%	81%

3.4. Algorithm Features

3.4.1. Frequency Range of Equalisation

The default frequency range of equalisation is from the low frequency -3 dB cut-off of the loudspeaker f_{LF} to 15 kHz. If there is a wide band cancellation in the frequency response around f_{LF} , or the high frequency level is decreased strongly due to an off-axis location or the loudspeaker is positioned behind a screen or due to very long measuring distance, manual readjustment of the design frequency range (indicated on the graphical output by the blue crosses, Figure 1) is needed. Naturally it is preferable to remove the causes of such problems, if possible.

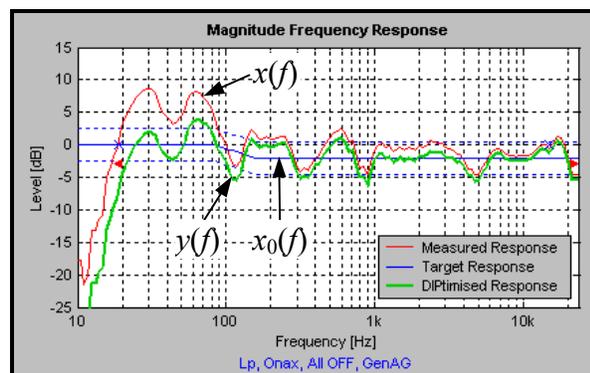


Figure 1. Typical graphical output of the optimiser software. Original response $x(f)$, target response $x_0(f)$ and final response $y(f)$. Also, -3 dB cut-off frequencies (triangles), optimisation range (crosses) and target tolerance (dotted).

3.4.2. Target for Optimisation

There are five target curves from which to select:

1. ‘Flat’ is the default setting for a studio monitor. The tolerance lines are set to ± 2.5 dB.

2. ‘Slope’ gives a user defined sloping target response. There are two user defined knee frequencies and a dB drop/lift value. A positive slope can also be set but is generally not desirable. The tolerance lines are set to ± 2.5 dB. Some relevant slope settings include:
 - -2 dB slope from low frequency -3 dB cut-off to 15 kHz for the large systems to reduce the aggressiveness of sound at very high output levels
 - -2 dB slope from 4 kHz to 15 kHz to reduce long-term usage listening fatigue
 - -3 dB slope from 100 Hz to 200 Hz for Home Theatre installations to increase low frequency impact without affecting midrange intelligibility
3. ‘Another Measurement’ allows the user to optimise a loudspeaker’s frequency response magnitude to that of another loudspeaker. For example, measure the left loudspeaker and optimise it, then measure the right speaker and optimise this to the optimised left speaker response. The result will be the closest match possible between the left and right speaker pair ensuring a good stereo pair match and phantom imaging. Tolerance lines are set at ± 2.5 dB.
4. ‘X Curve – Small Room’ will give the closest approximation to the X Curve for a small room as defined in ANSI/SMPTE 202M-1998 [10]. This is a target response commonly used in the movie industry. A small room is defined as having a volume less than 5300 cubic feet or 150 cubic meters. The curve is flat up to 2 kHz and rolls off 1.5 dB per octave above 2 kHz. Tolerance lines are set to ± 3 dB.¹
5. ‘X Curve – Large Room’ will give the closest approximation to the X Curve for a large room as defined in ANSI/SMPTE 202M-1998 [10]. The curve is flat from 63 Hz to 2 kHz and then rolls off at 3 dB per octave above 2 kHz. Below 63 Hz there is also a 3 dB roll off, with 50 Hz being down by 1 dB and 40 Hz by 2 dB. Tolerance lines are set to ± 3 dB with additional leeway at low and high frequencies.¹

An example of the room equaliser settings output for the large system optimised in Figure 1 is shown in Figure 2. The optimised result is displayed in green and dark grey boxes. The green boxes are room

¹ The room response controls do not directly support the X Curves but it may be possible to achieve X Curves in a room due to particular acoustic circumstances. This is also a good way to check how close the response is to the selected X Curve.

response controls that should be set on the loudspeaker. The light grey boxes are room response controls that are not present on the loudspeaker. Also displayed in this area is the error function, which is an rms of the optimised frequency response pass band.

DIP Switch	Setting
Treble Tilt	0 dB
Treble Level	-1 dB
Midrange Level	-2 dB
Bass Level	0 dB
Bass Tilt	-6 dB
Bass Roll-off	0 dB
Sub Sensitivity	0 dB
Sub Phase	0
Sub Bass Roll-off	0 dB
Error Function	1.92 dB
Processing Time	3.63 s

Figure 2. Output section displays all settings and values to be changed (green background) as well as the value of the error function and processing time.

4. PERFORMANCE OF THE OPTIMISATION ALGORITHM

The method for acquiring the in-situ impulse responses in both case studies is shown in Table 10.

4.1. Case Study 1

This is a case study using a two-way loudspeaker² placed next to a wall. This loudspeaker is a two-way design with a one-inch tweeter and an eight-inch bass driver with low frequency cut-off at 47 Hz. The loudspeaker was placed on a stand and positioned next to a wall and in the standard 0° centre position for 5.1 channel surround sound [40]. The measuring distance was 1.40 m.

Room acoustic measurements are shown in Appendix C (Figures 11-16). The octave and third octave reverberation times of the room were calculated according to ISO3382 [41]. The room is well damped having a broadband average T_{30} of 0.2 s. The lower mid-band frequencies (100-500 Hz) T_{30} is about 0.15 s and parts of the high frequency region (1.5-3 kHz) T_{30} is about 0.2 s. The spurious result at 50 Hz in the third octave band T_{30} seems to be an algorithm calculation error, as it does not appear in this way in the octave plot or Schroeder plot. The Schroeder curves indicate a smooth, linear and short decay in all octave bands except 62.5 Hz, which shows some stepping caused by reflections. There is a high level (-5 dB) reflection about 3 ms after the direct sound

due to the floor reflection. The loudspeaker suffers from a strong cancellation, due to the floor reflection, at 160 Hz. There is some ripple in the bass and midrange, however the high frequencies are very flat.

Table 10. Measurement settings in the acoustic measurement system.

Parameter	Setting
Measurement System	WinMLS2000 [38]
Microphone	Neutrik 3382 [39]
Sample rate, f_s	48 kHz
MLS sequence order	14
Averages	1
Impulse response length	0.341 s
Time window	Half-cosine
FFT size	16384
Frequency resolution	2.93 Hz

To determine room equalizer settings the default frequency range was considered (-3 dB low frequency cut-off to 15 kHz). The settings chosen by the optimisation algorithm are shown in Table 11 and the response of resulting room equaliser filter in Figure 4. Figure 5 shows the in-situ third octave smoothed frequency response of the loudspeaker after equalisation.

Table 11. Case study 1, room response control settings.

Control	Setting
Treble tilt	0 dB
Bass tilt	-2 dB
Bass roll-off	0 dB

The broadband rms deviation between the original responses (2.3 dB) and the optimised responses (2.0 dB) shows a reduction of 15% (0.36 dB).

The bass tilt control was set to reduce the effect of the boundary loading on the loudspeaker. No other controls were set as there was no gain increase at very low frequencies and high frequencies were quite flat. At these high frequencies the loudspeaker is sufficiently directional so there is little room influence, especially as the room was well damped. The remaining ripple cannot be corrected using the available room response controls.

² Genelec 1031A [1]

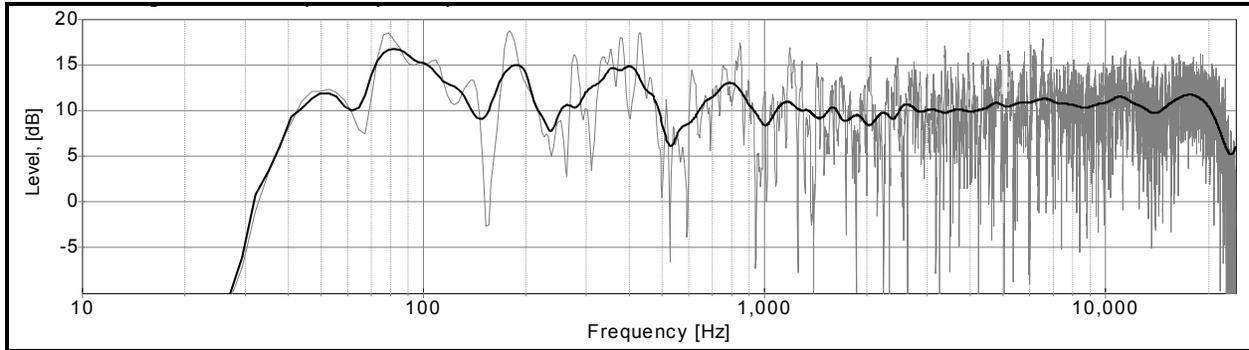


Figure 3. Case study 1, original frequency response (centre loudspeaker), (light curve) unsmoothed magnitude response, (dark curve) the third octave smoothed response

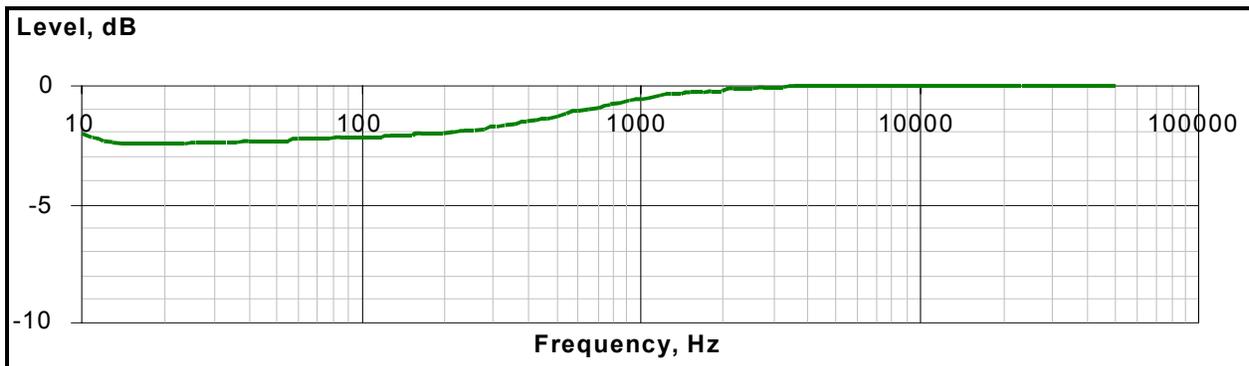


Figure 4. Case study 1, room response control filter shapes.

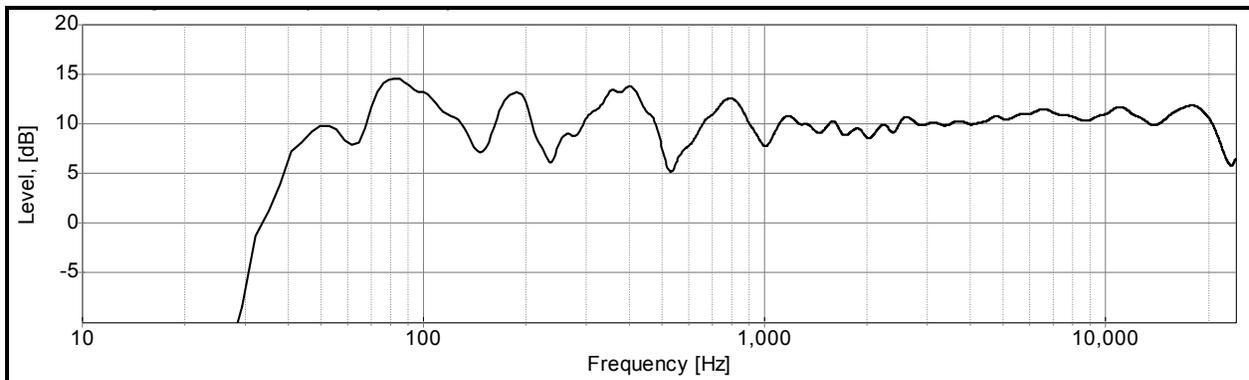


Figure 5. Case study 1, equalised magnitude of the frequency response. Third octave smoothing.

4.2. Case Study 2

This is a case study of a compact three-way loudspeaker³ placed in a very well damped listening room. In this case a reasonable improvement is shown to the in-situ response, however the fundamental acoustic problems in the room are not solved and can still be seen in the final response.

³ Genelec S30D [1]

The loudspeaker has an eight-inch bass driver with a low frequency cut-off of 35 Hz. There is also a 3.5-inch midrange driver and a ribbon tweeter having the upper cut-off frequency of 50 kHz. The loudspeaker was mounted on a stand and placed next to a hard wall in the front right position of a 5.1 surround sound system [40]. The measuring distance was 2.95 m.

Room acoustic measurements are shown in Appendix D (Figures 17-22). The octave and third octave reverberation times of the room were calculated according to ISO3382 [41]. The room is extremely

well damped having a T_{30} of about 0.2 s across the whole frequency range. The Schroeder curves indicate a smooth, linear and very short decay in all octave bands except at 62.5 Hz where low frequency room effects frequently appear even in good rooms. There is a high level (−6 dB) reflection about 3 ms after the direct sound due to the floor reflection. Another high level reflection (−9 dB) is seen about 5 ms after the direct sound, and this is from the sidewall.

The frequency response before equalisation (Figure 6) shows relatively flat mid and high frequencies above 1 kHz. There is a gain increase around 35 Hz. Between 100 Hz and 1 kHz large notches and gain increases affect the magnitude response to produce a very non-flat response even after third octave smoothing. Wide notches can be seen in the smoothed response around 100 Hz, 200 Hz and 600 Hz. Gain increases can be seen between these notches. The rapid roll-off in measurements above 23 kHz is due to the measurement system's anti-aliasing filter, not the loudspeaker's natural response.

To determine room equalizer settings the default frequency range was considered (−3 dB low frequency cut-off to 15 kHz). Settings recommended by the optimisation algorithm are shown in Table 12 and the resulting room equaliser filter response in Figure 7.

Table 12. Case study 2, room response control settings.

Control	Setting
Treble Level	−3 dB
Midrange Level	−4 dB
Bass Level	0 dB
Bass Tilt	−6 dB
Bass Roll-off	0 dB

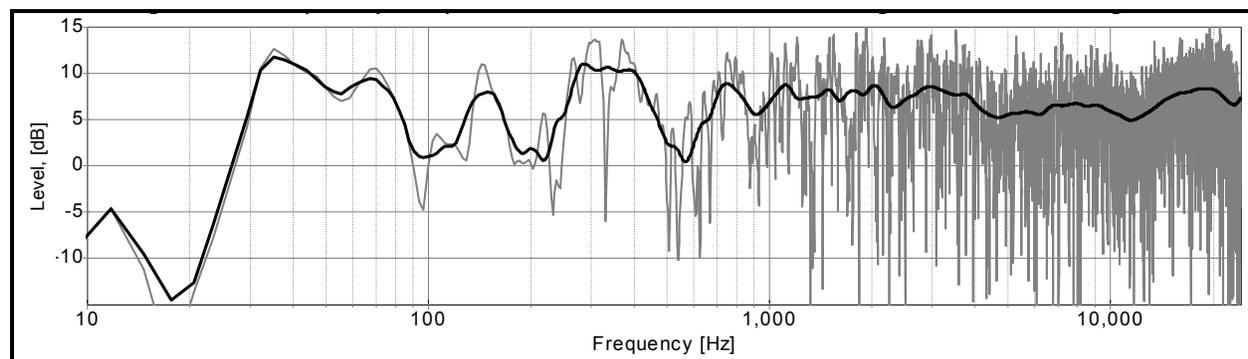


Figure 6. Case study 2, original frequency response, (light curve) unsmoothed magnitude response, (dark curve) the third octave smoothed response.

After equalisation, the response (Figure 8) is closer to the target of a flat response. The broadband rms deviation between the original responses (2.8 dB) and the optimised responses (2.6 dB) shows a reduction of 7% (0.2 dB).

The gain increase at 35 Hz has been flattened due to the bass tilt setting. The midrange/treble balance is flatter due to the 1 dB decrease in the midrange level relative to the treble level. The equalisation was unable to improve the bass and midrange notches and gain increases because the room response controls are not designed to compensate for this type of acoustic problem.

After equalisation the midrange gain increase around 300-400 Hz has become more prominent because the bass response has been flattened. The subjective impact of this change should be tested to see if the resulting balance is disturbing.

In the 100 Hz to 1 kHz region there are cancellations that should be removed by damping reflection sources. The gain increase around 500 Hz is caused by the floor reflection and should also be damped or the loudspeaker cabinet raised up to reduced the level of the floor reflection. The sidewall reflections should also be damped to reduce their effects. Such narrow-band problems remain because they cannot be fixed using the room response controls.

4.3. Other Case Studies

Further detailed examples of the optimisation algorithm performance in different acoustic conditions, with smaller and larger loudspeaker systems and with differing target responses can be found in [42].

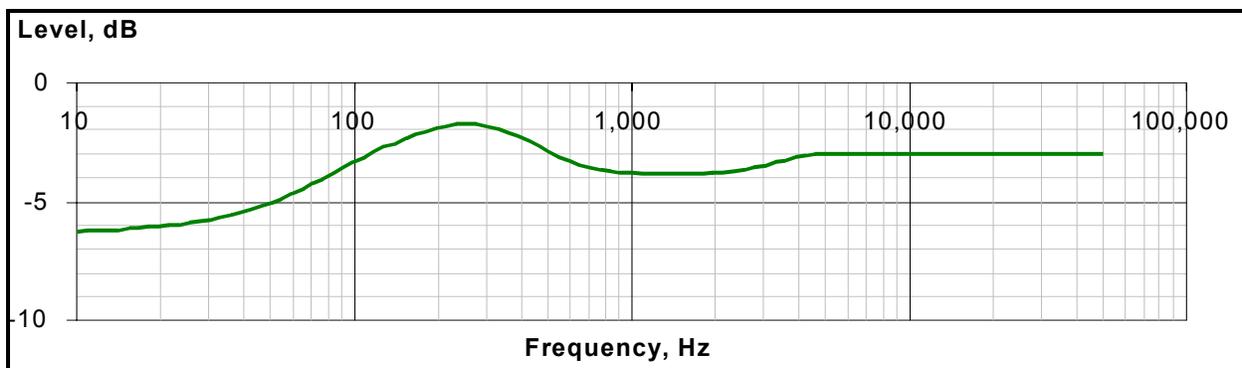


Figure 7. Case study 2, room response control filter shape.

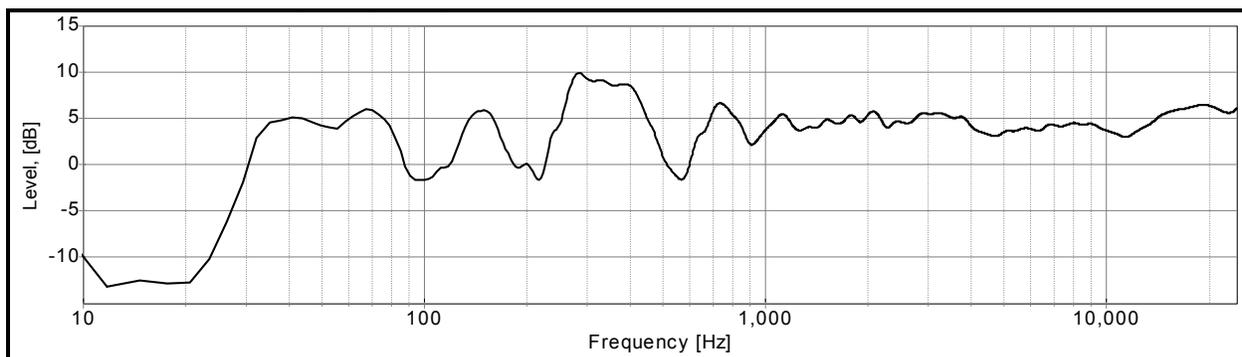


Figure 8. Case study 2, equalised magnitude of the frequency response. Third octave smoothing.

5. DISCUSSION

The objective of this paper is to introduce an automated system for choosing appropriate room response control settings once an in-situ frequency response measurement has been made.

The room response controls in active loudspeakers implement discrete filter values instead of a continuous value range. However, the number of possible combinations available can be quite large. Even an experienced operator can find it difficult to choose the optimal settings.

The task of the automated optimiser is to find the optimal combination from the possible combinations of discrete values. The cost of performing a brute force search of all value combinations and then choosing the best among them is prohibitive in terms of computer processing time. The approach chosen is to exploit heuristics of experienced calibration engineers, and to reduce the number of alternatives by dividing the task into subsections that can reliably be solved independently. A significant part of the heuristics is the order in which these choices should be taken.

A considerable improvement in the speed of optimisation was achieved. The optimisation algorithm is

relatively robust to a wide variety of situations, such as varying room acoustics, different sized loudspeakers with differing anechoic responses, and varying in-situ responses [42]. The optimisation is efficient and so the software is fast enough to be used routinely at in-situ loudspeaker calibrations.

Two case studies demonstrate the system. The recommended settings achieve improved equalisation in the form of a smaller rms deviation from the target. The degree of improvement is not limited by the optimisation method but by the fact that room response controls are not intended to correct narrow-band deviations in the loudspeaker frequency response. Most of these variations in measured responses result from acoustic issues such as cancellations and comb filtering due to reflections. These should be solved acoustically rather than electronically.

If the measured magnitude of the frequency response has a loss of energy around the low frequency cut-off due to a cancellation in the room, the optimisation algorithm will attempt to compensate for this unless the user prevents it by raising the low corner of the optimisation frequency range.

6. CONCLUSIONS

An optimisation algorithm was described to efficiently select the optimal values for discrete-valued room response equaliser filters used in professional active monitoring loudspeakers. The efficiency and reliability has been achieved by exploiting heuristics of experienced sound system calibration engineers. The advantage of the automated algorithm is that it performs consistently and fast, irrespective of the operator. The algorithm has been implemented in a loudspeaker calibration tool, which is used by specialists who set up and tune studios and listening rooms.

7. ACKNOWLEDGEMENTS

The authors would like to thank Steve Fisher (SCV London) for the original inspirational idea, Lars Morset (Morset Sound Development) and Genelec Oy. This work is also presented in more detail as an MSc Thesis at the Helsinki University of Technology [42].

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APPENDIX A – SOFTWARE FLOW CHART

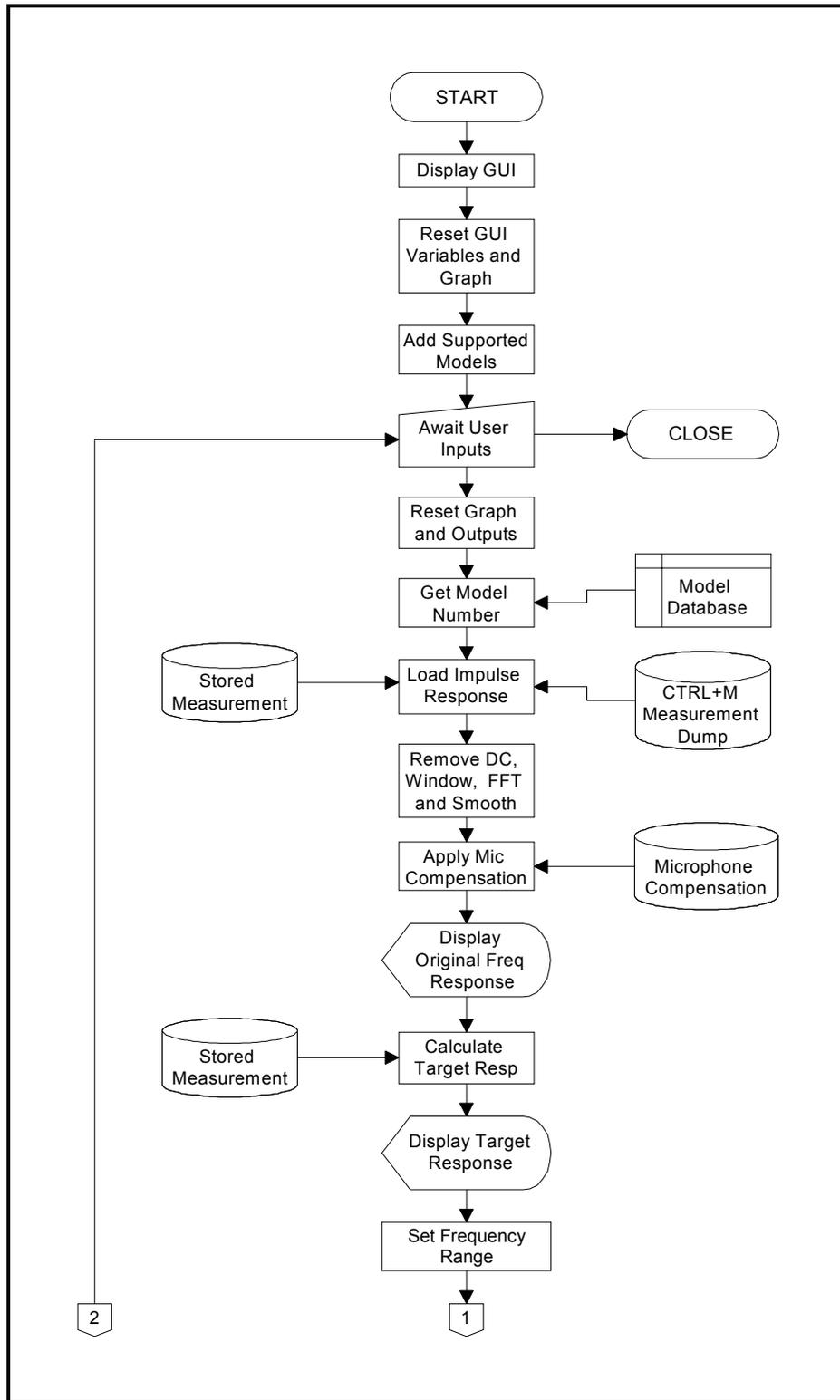


Figure 9. Software flow chart, part 1.

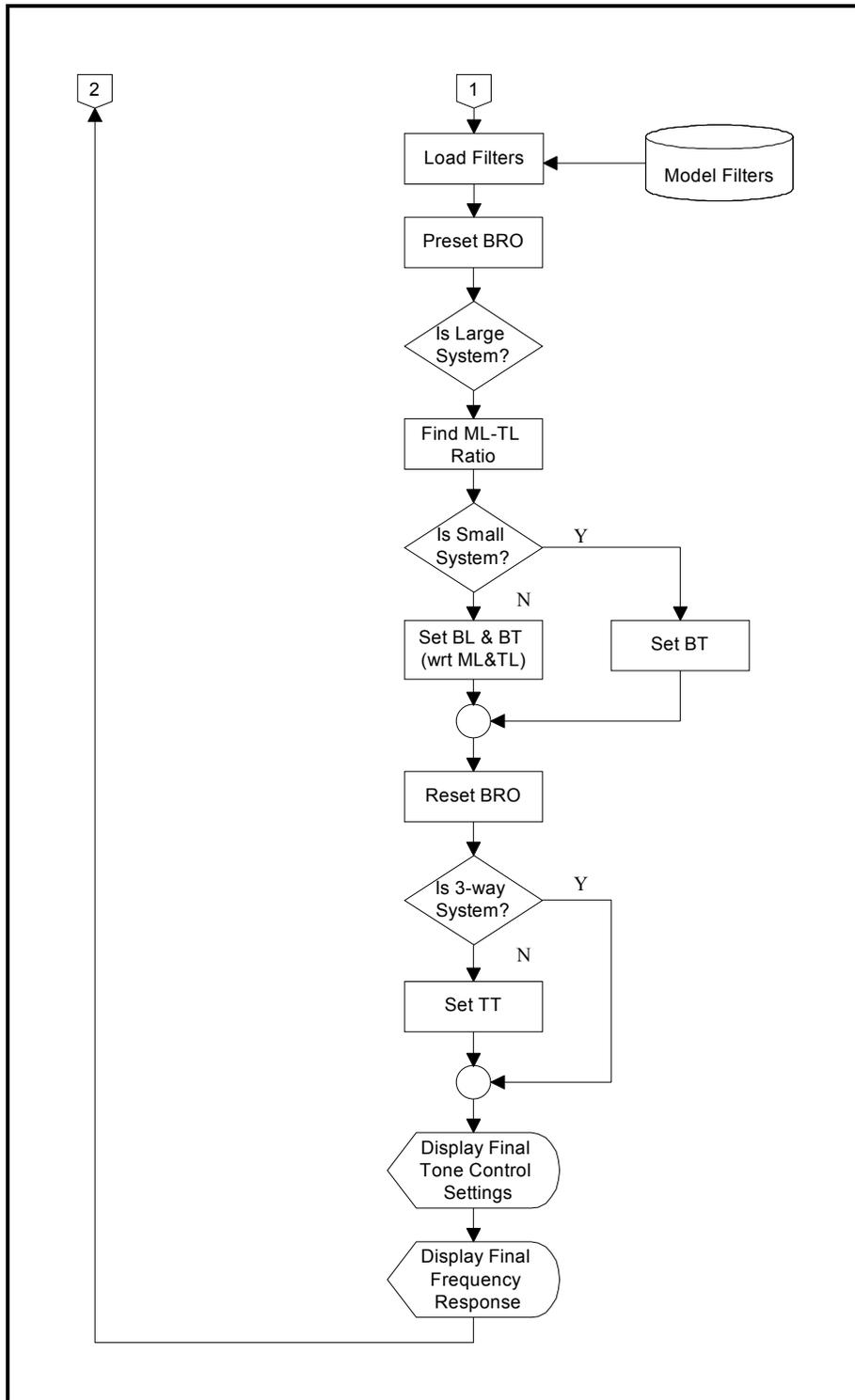


Figure 9 cont'd. Software flow chart, part 2.

APPENDIX B – SOFTWARE GRAPHICAL USER INTERFACE

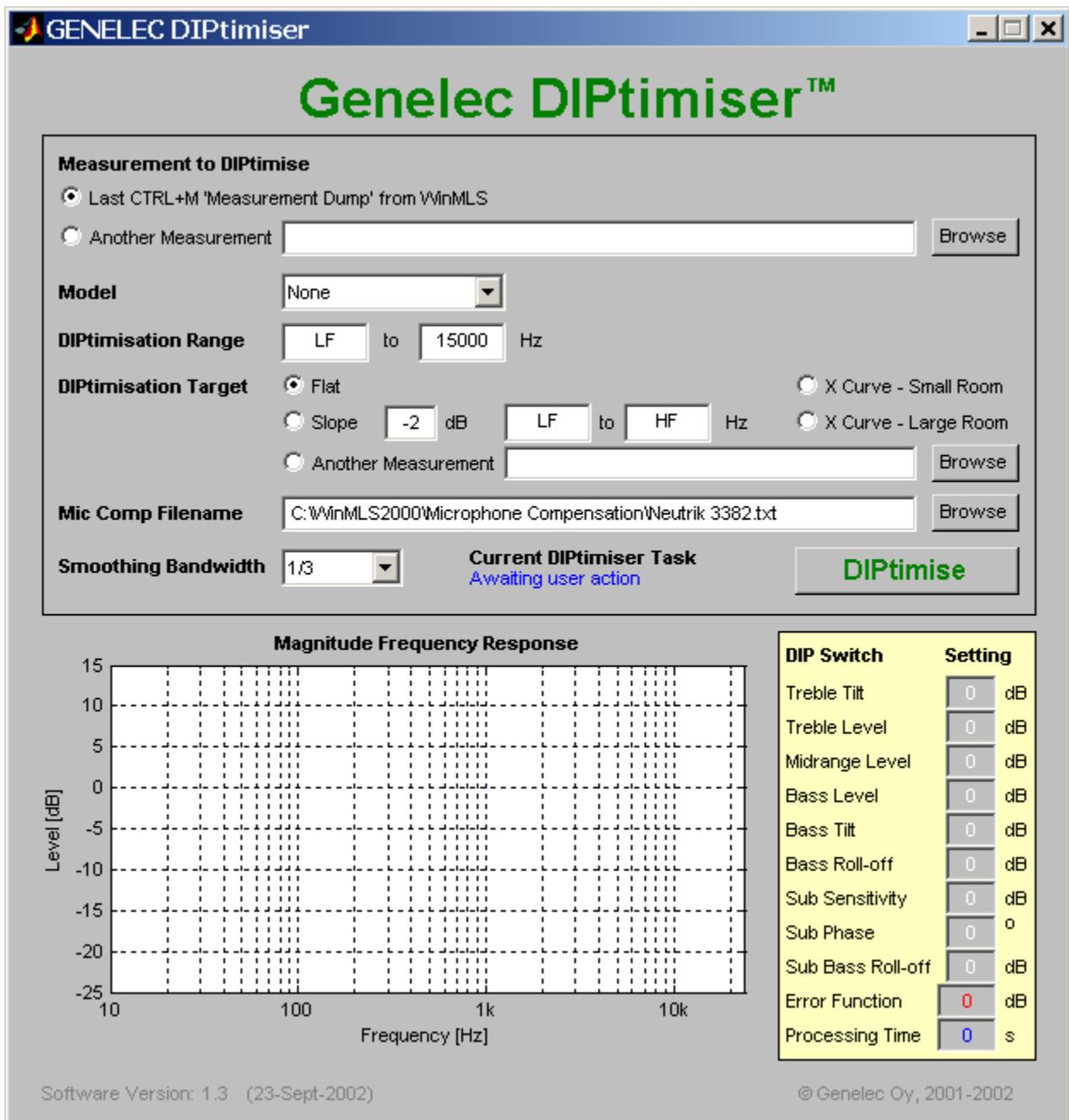


Figure 10. Software graphical user interface at start up.

APPENDIX C – CASE 1, ROOM ACOUSTICS

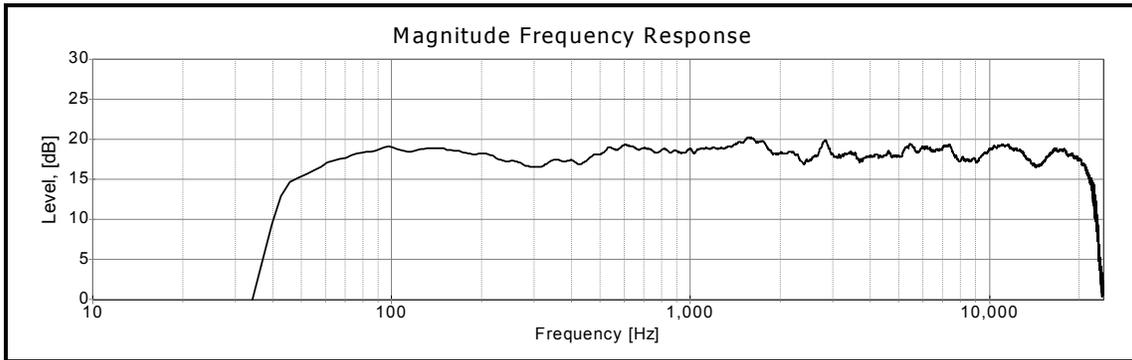


Figure 11. Case 1, in-situ magnitude response.

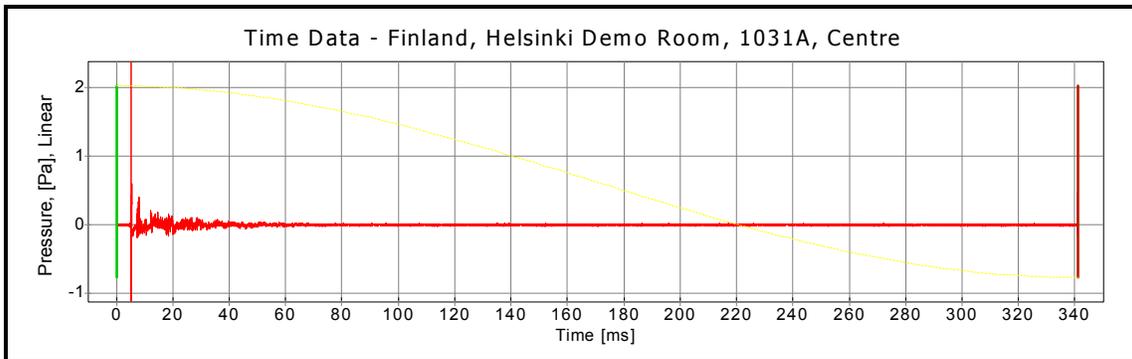


Figure 12. Case 1, impulse response data (solid line), time window (dotted line).

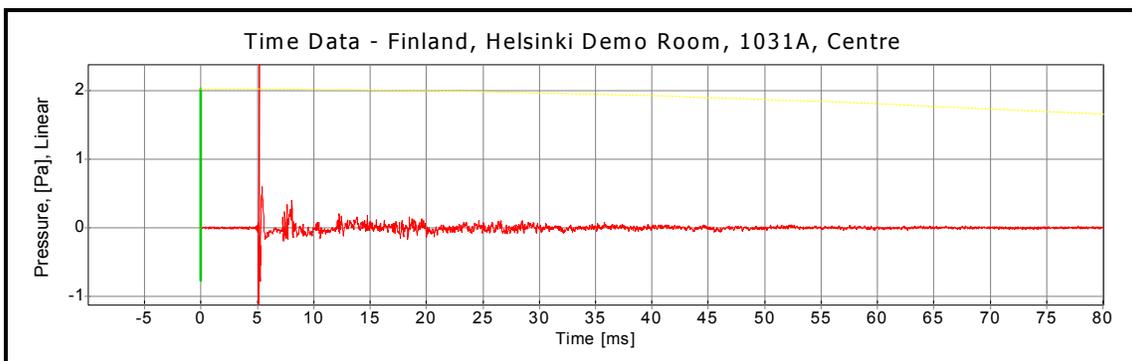


Figure 13. Case 1, zoomed impulse response data (solid line), time window (dotted line).

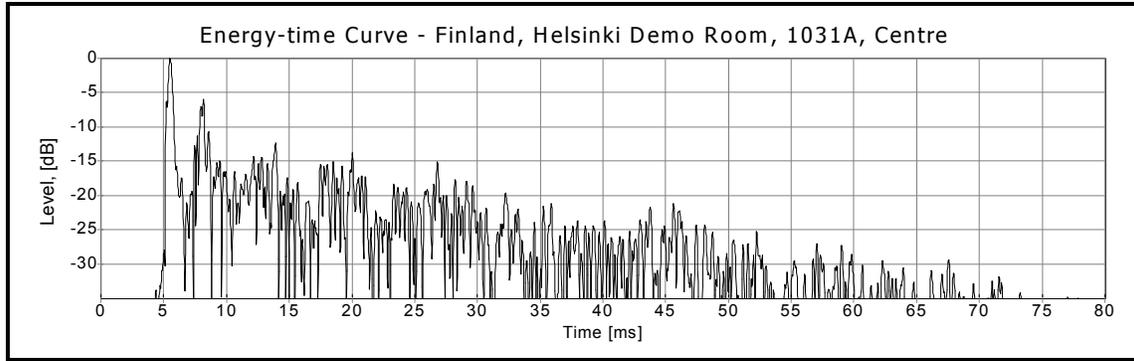


Figure 14. Case 1, energy-time curve. This data is low pass filtered at 4 kHz.

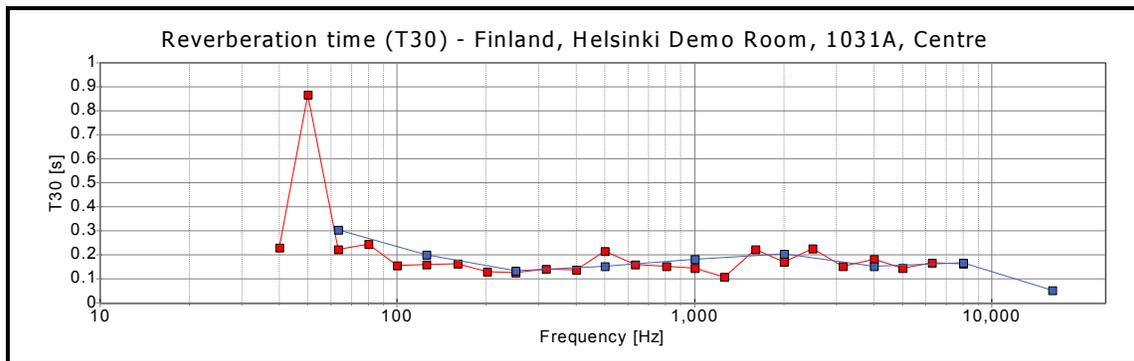


Figure 15. Case 1, octave and 1/3 octave reverberation time.

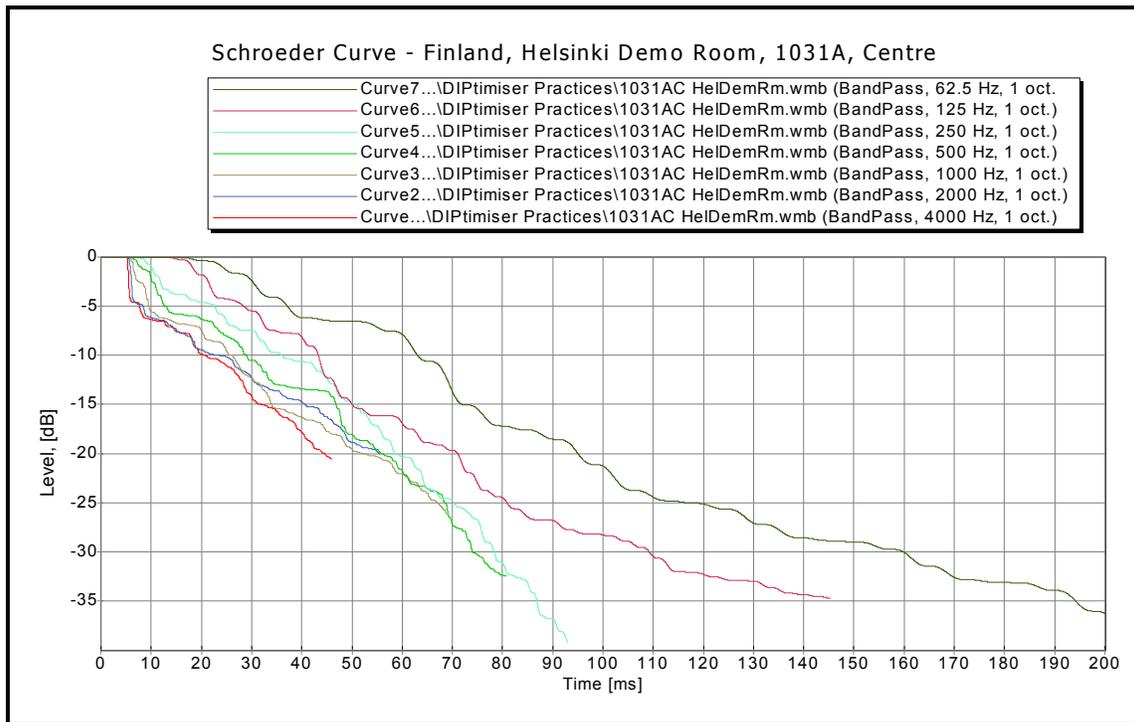


Figure 16. Case 1, octave filtered Schroeder plot.

APPENDIX D – CASE 2, ROOM ACOUSTICS

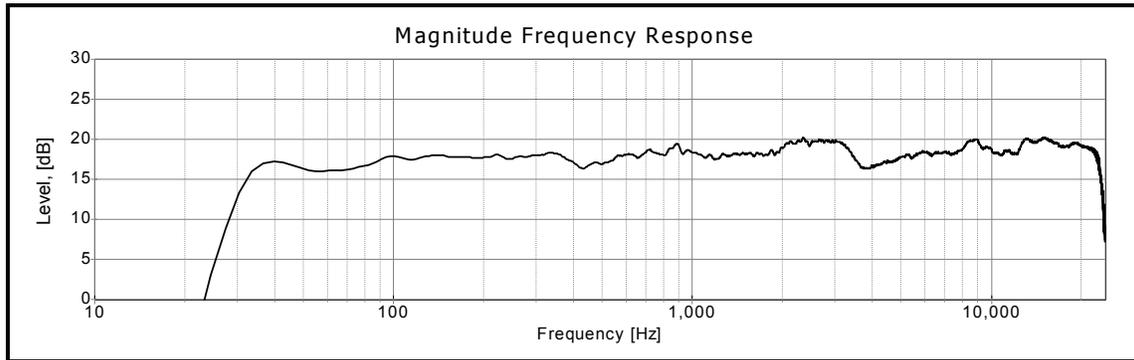


Figure 17. Case 2, anechoic frequency response.

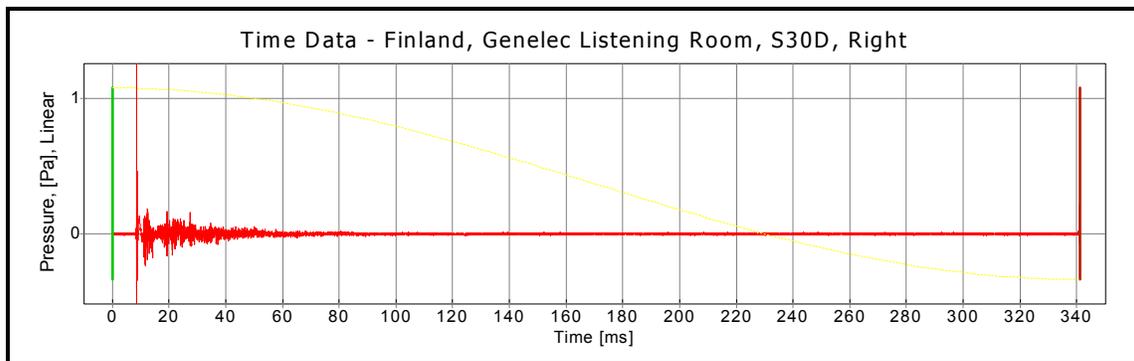


Figure 18. Case 2, impulse response data (solid line), time window (dotted line).

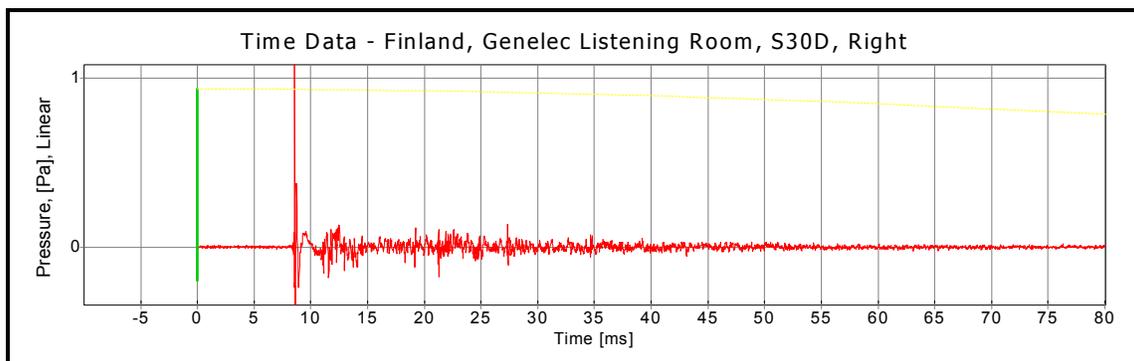


Figure 19. Case 2, zoomed impulse response data (solid line), time window (dotted line).

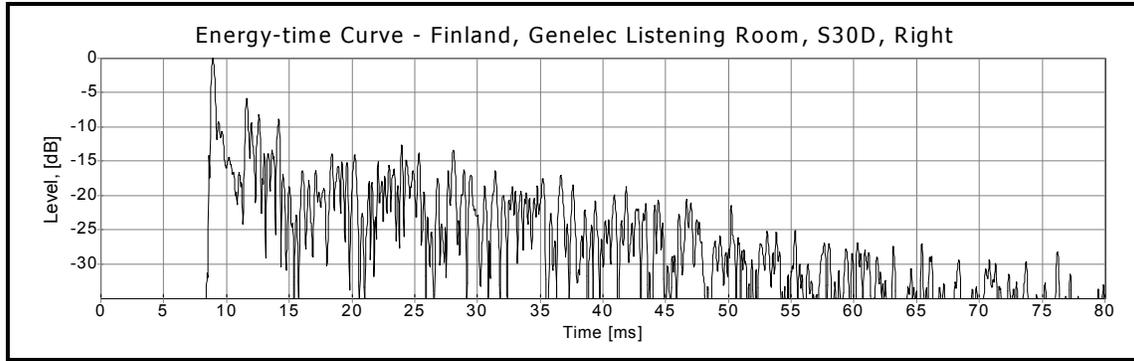


Figure 20. Case 2, energy-time curve. This data is low pass filtered at 4 kHz.

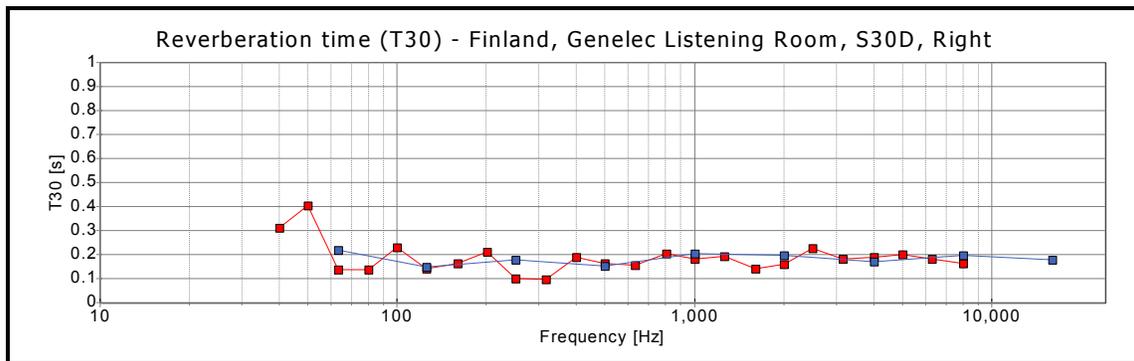


Figure 21. Case 2, octave and 1/3 octave reverberation time.

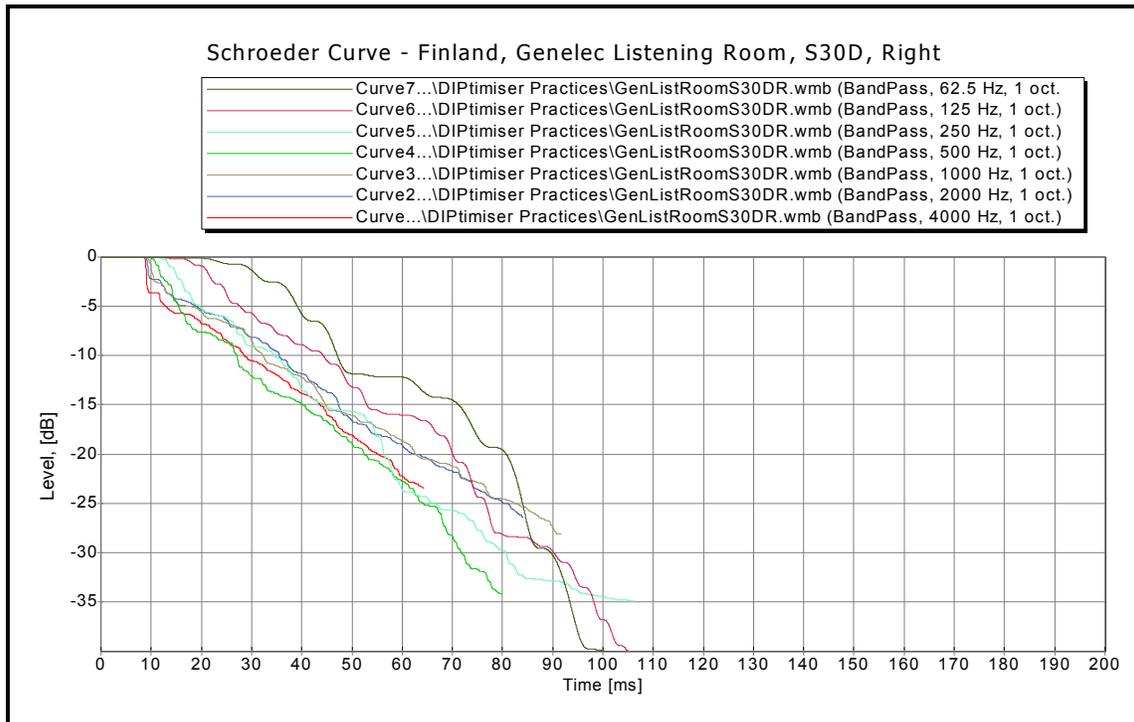


Figure 22. Case 2, octave filtered Schroeder plot.