AN AUTOMATED IN-SITU FREQUENCY RESPONSE OPTIMISATION ALGORITHM FOR ACTIVE LOUDSPEAKERS, INCLUDING A STATISTICAL ANALYSIS OF ITS PERFORMANCE

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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 individually calibrated in anechoic conditions to have a flat frequency response magnitude within design limits of ±2.5 dB.
When a loudspeaker is placed into the listening environment, the frequency response changes due to loudspeaker-room interaction. To help alleviate this, the active loudspeakers incorporate a pragmatic set of room response controls, which account for 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 loudspeakers. Significant variance between calibrations can be seen even with experienced system calibrators. Additional variance will occur with different people calibrating loudspeaker systems. An automated calibration method was developed to ensure consistency of calibrations because of these reasons.
Presented first in this paper is the discrete-valued room response equaliser employed in the active loudspeakers. Then, the algorithm for automated value selection is explained including the software structure, algorithm, features and operation. The performance of the optimisation algorithm is then investigated by studying the statistical properties of frequency responses before and after equalisation.

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, not to convert the listening room anechoic. In fact, listeners prefer to hear some room response in the form of liveliness creating a spatial impression and some envelopment[2].
An approach to improve the loudspeaker performance in a room is to choose an optimal location for the loudspeaker. Cox and D'Antonio[3] (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. Problems with this approach are that optimisation may not be practically possible in all cases.
Electronic equalisation to improve the subjective sound quality has been widespread for at least 40 years; see Boner & Boner[4] for an early example. Equalisation is particularly prevalent in professional applications such as recording studios, mixing rooms and sound reinforcement, typically using a separate equaliser, although equalisers are increasingly built into active loudspeakers. Some equalisers play a test signal and alter their response according to the in-situ transfer function measured[5]. This process can be sensitive and simple 'press the button and everything will be OK' approach proves hard to achieve with reliability, consistency and robustness.
Equalisation may become skewed if based only on a single point measurement. The frequency response in nearby positions can actually become worse after equalisation designed using only a sin-

gle point measurement. A classical method to avoid this is to use a weighted average of responses measured within a listening area. Such spatial averaging is often required when the listening area is large; see examples described in the automotive industry[6] and cinema in the SMPTE Standard 202M[7]. Spatial averaging can reduce local variance in mid to high frequencies and can reduce problems caused by the fact that a listener perceives sound differently to a microphone, but typically reduces the accuracy of equalisation obtained at the primary listening location.

The room transfer function is position dependent, and this poses major problems for all equalisation techniques. For a single loudspeaker in diffuse field no correction filter is capable of removing differences between responses measured at two separate receiver points. At high frequencies a required high-resolution correction can become very position sensitive. Frequency dependent resolution change is then preferable and is typically applied[8,9] but with the expense of reduced equalisation accuracy. Perfect equalisation able to achieve precisely flat frequency response in a listening room, even within a reasonably small listening area, appears not possible. An acceptable equalisation is typically a compromise to minimise the subjective coloration in audio due to room effects.

Typically electronic equalisation in active loudspeakers uses low order analogue minimum phase filters[10-12]. Since the loudspeaker-room transfer function is of substantially higher order than such equaliser, the effect of filtering is to gently shape the response. Even with this limitation, in-situ equalisers have the potential to significantly improve perceived sound quality. The practical challenge is the selection of the best settings for the low-order in-situ equaliser.

Despite advances in psychoacoustics, it is difficult to quantify what the listener actually perceives the sound quality to be, or to optimise equalisation based on that evaluation[13-15]. Because of this, in-situ equalisation typically attempts to obtain the best fit to some objectively measurable target, such as a flat third-octave smoothed response, known to link to the perception of sound free from coloration. Also, despite the widespread use of equalisation, it is still hard to provide exact timbre matching between different environments.

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[16-24]. 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[25-30]. If realisable, these are extremely attractive ideas because they imply that the perceived sound could be modified with precision, to different target responses. Then, 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 equalisation transfer functions, and can therefore be highly local in space[25].

2.2 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 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. Three way 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 3. Two way 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 4. Large system 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 $E(x)$,

$$\min E(x) \tag{1}$$

where, x is the vector of design parameters, $x \in \Re n$. 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.

	Type of loudspeaker			
Room Response Control	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[38] 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). A screenshot of the software graphic user interface can be seen in Appendix A and a flow chart of the software can be seen in Appendix B.

Table 6. Optimisation stages.

	Type of loudspeaker			
Optimisation stage	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 minimised is given by Equation 2,

$$\min_m E = \left| \frac{\max\limits_{f_a}\left(\dfrac{a_m(f)x(f)}{x_0(f)}\right)}{\max\limits_{f_b}\left(\dfrac{a_m(f)x(f)}{x_0(f)}\right)} \right|, \qquad f_a = [f_1, f_2],\ f_b = [f_2, f_3] \tag{2}$$

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,

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

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). The user can select both values. 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 method used to set the bass roll-off earlier is repeated, but without modifying upwards the final setting. The same objective function as presented in Section 3.2.1 is used.

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). The user can select both values. The defaults 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 it is skipped for three ways because having no such control.

3.3 Reduction of Computational Load

The optimiser algorithm reduces the computational load by exploiting the heuristics of experienced calibration engineers. As a result, the number of filtering steps has dramatically reduced for larger systems (Table 9) and even relatively simple two-way systems show a substantial improvement compared to the number of steps needed by direct search method (Table 5). There are two main reasons for this improvement: the constraint of not allowing modification of all three driver level settings simultaneously and breaking-up of optimisation into stages.

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

	Type of loudspeaker			
Optimisation stage	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%

The run time on a PII 366 MHz computer for three-way and large systems is about 15 s (direct search for three-way systems 3 minutes, large systems 15 minutes). The processing time is directly proportional to the processor speed. A modern PIII 1200 MHz based computer takes about 4 s to perform the same optimisation. Further improvements in the software have improved run times by about 30%.

3.4 Algorithm Features

3.4.1 Frequency Range of Equalisation

The default equalisation frequency range is from the loudspeaker low frequency –3 dB cut-off f_{LF} to 15 kHz. Manual readjustment of the design frequency range (indicated on the graphical output by blue crosses, Figure 1) is needed in some special cases. Examples of these include a strong cancellation notch in the frequency response around f_{LF}, when off-axis loudspeaker location reduces significantly the high frequency level, when a loudspeaker is positioned behind a screen, or when the measuring distance is very long. It is naturally preferable to remove such causes of problems, if possible.

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’ allows the user to define a 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 normally not acoustically desirable. The tolerance lines are set to ±2.5 dB. Some relevant slope settings include:
   - for large systems a –2 dB slope across the passband up to 15 kHz 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 loudspeaker and optimise this to the optimised left loudspeaker response. This results in the closest match possible between the left-right loudspeaker pair ensuring good stereo pair match and phantom imaging. Tolerance lines are set at ±2.5 dB.
4. ‘X Curve – Small Room’ approximates the X Curve for a small room (volume less than 5300 cubic feet or 150 cubic meters) as defined in ANSI/SMPTE 202M-1998[7]. 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 – see footnote 1. This is a target response commonly used in the movie industry.

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[7]. 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 – see footnote 1.

4 PERFORMANCE OF THE OPTIMISATION ALGORITHM

To assess the performance of the combination of optimisation algorithm and equalisation in the loudspeakers, the analysis compares the unequalised in-situ frequency response to the response after equalisation. The MLS measurement technique was used to measure the in-situ acoustical frequency responses. The acquisition system parameters are shown in Table 10. The values in parentheses are the parameters used for acquiring the impulse response for models that have a bass extension below 30 Hz.

The room response control settings were calculated for each loudspeaker response according to the algorithm discussed in Section 3 and statistical data for each measurement before and after equalisation was recorded. The statistical data is analysed to study how the objective quality of the system magnitude response has been improved by using the proposed algorithm for setting the room response controls.

Table 10. Acoustic measurement system parameters.

Parameter	Equipment / Setting
Measurement System	WinMLS2000[39]
Microphone	Neutrik 3382[40]
Sample rate, f_s	48 kHz
MLS sequence order	14 (16)
Averages	1
Impulse response length	0.341 s (1.36 s)
Time window	Half-cosine
FFT size	16384 (65536)
Frequency resolution	2.93 Hz (0.733 Hz)

4.1 Statistical Data Analysis

Statistical analysis was conducted to assess the ability of the equalisation algorithm to attain a target response. The target in the study was a flat frequency response, and several statistical descriptors were employed to indicate how much the equalised response deviates from the flat target. The performance was separately studied for narrowband and wideband deviations.

The frequency bands considered in the study were the full loudspeaker passband, and its subsections called 'LF', 'MF' and 'HF' (see Table 11), collectively referred to as 'subbands'. These subbands correspond roughly to the bandwidths of each driver in a three-way system.

Several statistical descriptors of sound pressure were calculated to indicate the ability of equalisation to reduce narrow band variation of sound pressure. These were the minimum, maximum, range (of magnitude dB values), median, upper and lower quartiles, 5% and 95% percentiles calculated for the full loudspeaker passband and in each subband, as well as the root-mean-square (RMS) deviation of sound pressure from the median within each subband (value expressed in dB). In order to calculate these, the median broadband (Table 11) magnitude response for each loudspeaker was standardised to zero dB.

[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.

When a response is flat in a broadband sense, the medians calculated over various (large) frequency bands are similar. In this study, differences of median sound pressure between subbands are taken to indicate that broadband tonal balance of a response is not flat. An improvement in the broadband tonal balance due to equalisation is then indicated by a reduction of median value differences.

Table 11. Frequency band definitions the statistical data analysis; f_{LF} is the frequency of the lower –3 dB limit of the frequency range.

	Frequency Range Limit	
Bandwidth Name	Low	High
Broadband	f_{LF}	15 kHz
LF	f_{LF}	400 Hz
MF	400 Hz	3.5 kHz
HF	3.5 kHz	15 kHz

4.2 Example of Statistical Data Analysis

Figure 1 shows an example where room response control settings are calculated according to the optimisation algorithm. The equalisation target is a flat magnitude response (straight line at 0dB level). The in-situ frequency response of the loudspeaker was recorded before equalisation, i.e. when all the room response controls were set to their default position, which has no effect to the response. The appropriate room response control settings were calculated using the optimisation algorithm, applied to the loudspeaker and the corrected in-situ frequency response plotted. The loudspeaker's passband (triangles) and the frequency band of equalisation (crosses) are indicated on the graphical output. The proposed room response control settings are shown and the effect of these settings is visualised in the response plot. The treble tilt, midrange level and bass tilt controls have been set. The equalisation corrects the low frequency alignment and improves the linearity across the whole passband. The optimised result is displayed in green and dark grey boxes. The green boxes are room 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.
Figure 2 shows a statistical analysis of the same loudspeaker presented in graphical form. The upper three plots were calculated before equalisation and the lower three plots after equalisation. The plots display the values of percentiles in the magnitude value distribution (box plot), the histogram of values and the fit of the magnitude values to normal distribution before and after equalisation. These plots clearly show that the deviation in magnitude data has been reduced. This is illustrated by the reduced range in the box plot and the value histogram, as well as a better fit to a normal distribution in the normal probability plot.
Two detailed case studies can be seen in[38]. Responses before an after equalisation are shown together with room acoustic analysis to show that the algorithm performs well, even in widely varying conditions.

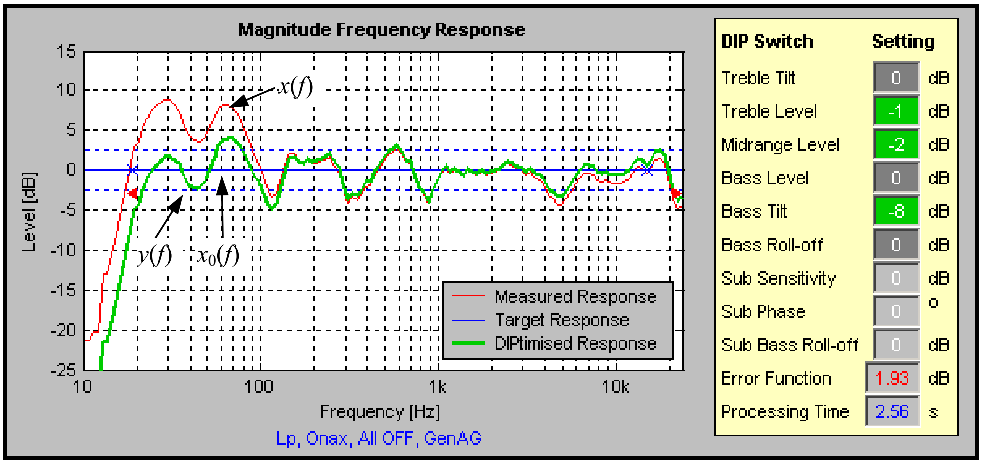

Figure 1. Graphical output of the optimiser software. Original response $x(f)$, target response $x_0(f)$ and equalised response $y(f)$, cut-off frequencies (triangles), optimisation range (crosses) and target tolerance (dotted lines). Output section on the right displays possible settings and values to be changed (green background) as well as the error function value and processing time.

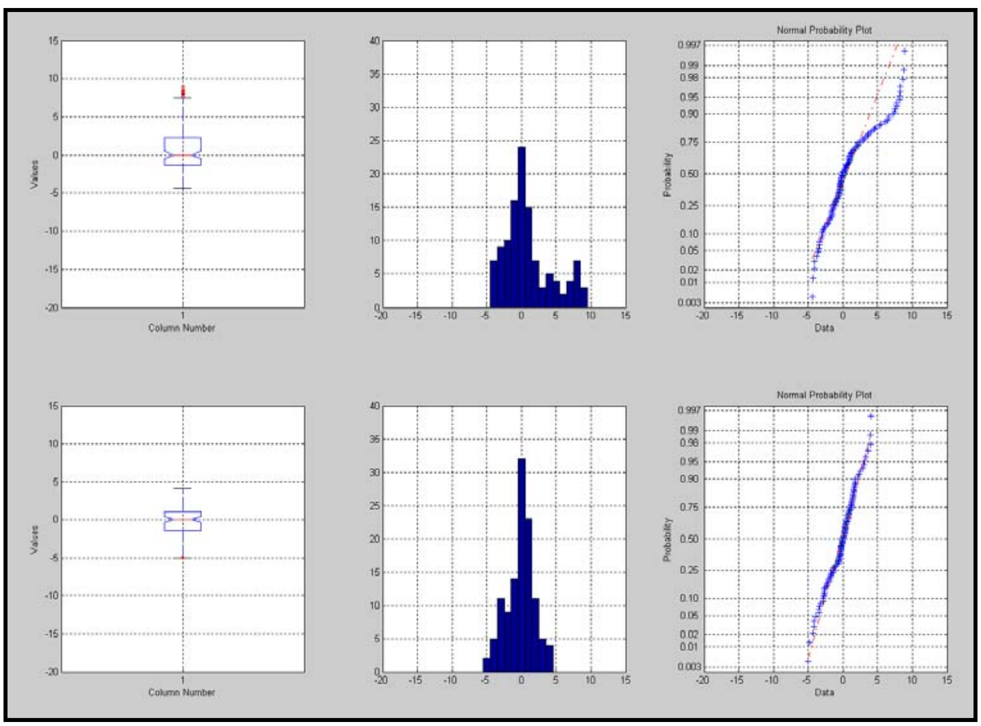

Figure 2. Case example, statistical analysis output.

4.3 Results

63 loudspeakers were measured before and after equalisation (12 small two-way, 22 two-way, 30 three-way and three large systems). Depending on the product type, not all of the room response controls are available (Tables 1–4). Table 12 shows the number times a control was used when available on a loudspeaker. The midrange level control is used the most frequently and the bass roll-off the least.

Table 12. Use of available room response controls.

Room Response Control	Usage vs. availability	% Usage
Midrange Level	27/33	82%
Treble Level	22/33	67%
Bass Tilt	37/67	55%
Treble Tilt	11/37	30%
Bass Level	8/33	24%
Bass Roll-off	10/67	15%

Appendix D shows quartile difference and RMS deviations for each loudspeaker in the study, for the broadband and each subband. The quartile difference or RMS deviation after equalisation is subtracted from the same before equalisation. An improvement will produce a negative value of difference. Quartile difference and RMS deviation values represent different ways to look at the deviation from the distribution median value. Quartile limits are more robust to outlier values while the RMS value is affected by them.

For small two-way systems (Figures 9-10[41]), the main improvement is seen at low frequencies in four out of 12 cases. Only in one case is there is a significant improvement in the broadband flatness.

The broadband flatness of the two-way systems is improved in four (quartile data, Figure 11) or eight (RMS data, Figure 12[41]) cases out of 22. An equal number of reductions and increases of low frequency quartile values can be seen. MF subband quartile values improve in one case and deteriorate in 5 cases while there are no changes in the HF subband. The flatness in the broadband and LF subband as indicated by RMS deviation data has improved, indicating a reduction of outlier values. The MF and HF subbands show no changes or a slight increase of the RMS deviation.

Three-way systems show in most cases a clear reduction of both the quartile difference (Figure 13[41]) and RMS deviation (Figure 14[41]) for the broadband and LF subband. There is no significant change in the MF and HF subbands.

A similar trend is seen for the three large systems included in this study (Figures 15-16[41]). Mainly the LF subband flatness is improved and this is reflected in broadband flatness improvement.

Some responses appear to worsen in terms of quartile difference and RMS deviation in the subband analysis. This was not evident in the broadband metrics, indicating that the arbitrary definition of subband frequency division introduced some error. The cases where this happened originally suffered from severe response anomalies due to extremely bad room acoustics. The equalisation was not designed to compensate for such problems.

Subband median level differences (Figure 3) demonstrate the broadband frequency balance. Acoustical loading of a loudspeaker by nearby boundaries is reflected in the LF subband median level before equalisation, especially for three-way models that are typically flush mounted. The median level of the LF subband is reduced by equalisation, indicating that equalisation compensates well for this loading. Smaller difference in median values across subbands shows that equalisation has improved broadband flatness. The largest improvement is seen in three-way loudspeakers. For two-way systems equalisation has improved broadband flatness only marginally. The broadband flatness improvement is mainly a result of better alignment of the LF subband with the MF and HF subbands. The equalisation has not only reduced the variation inside individual subbands but also improved the broadband flatness of the acoustical response. This should translate to a reduced audio colouration at the listening position.

All loudspeakers pooled together (Figure 3), equalisation reduces median value variance for the LF subband for all loudspeaker types. Only in three-way systems an improvement is seen also in MF and HF subband median value variances.

Figure 4 shows pooled results for all products and results for each product cathegory, excluding the three main monitors. For all models, the broadband flatness has been improved (by 0.4 dB), and the RMS deviation has been reduced. The largest reduction is seen in three-way systems. To some extent, the result is similar for the quartile difference but the small two-way and two-way systems do not experience such large improvement. This indicates that the improvement is mainly a reduction of extreme magnitude values (peak height and notch depth) in the low frequency response.

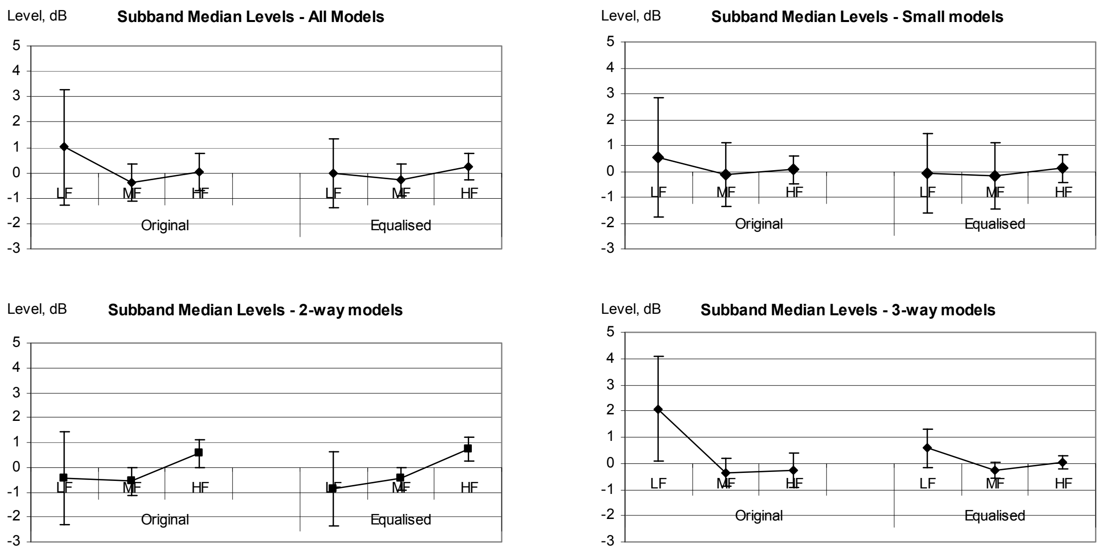

Figure 3. Mean and standard deviation of subband median levels before and after equalisation.

5 DISCUSSION

The objective of this paper is to present an automated system for choosing appropriate room response control settings once an in-situ frequency response measurement has been made and to evaluate objectively the efficacy of the proposed method of selecting the settings.

Active loudspeakers room response controls implement a discrete set of filter parameter values rather than providing a continuous value range. The number of possible setting combinations can be quite large and even an experienced operator can find it difficult to systematically choose the optimal settings.

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

A considerable improvement in the speed of optimisation was achieved relative to an exhaustive search. The optimisation algorithm is robust to a wide variety of situations, such as variations of room acoustics, differently sized loudspeakers with differing anechoic responses and varying in-situ responses[42]. The optimisation is sufficiently efficient making the software fast enough to be used routinely at in-situ loudspeaker calibrations.

A case study demonstrates the statistical changes due to the optimisation algorithm's recommended room response control settings. The settings achieve improved equalisation in the form of a smaller RMS deviation from the target response. The improvement is not limited by the optimisation method but by the room response controls not intended to correct for narrow-band deviations. Examples of these are response variations resulting from acoustic issues such as cancellations associated comb filtering due to reflections. These should be solved acoustically rather than electronically.

The statistical analysis of 63 loudspeakers shows that the automated equalisation is able to systematically reduce the variability in the equalised responses and to improve the frequency response flatness relative to the target response. It achieves this by improving both the broadband frequency balance and by reducing narrow-band variability in the response, particularly at low frequencies. The main improvement is the reduction of extreme (outlier) values at low frequencies.

It is interesting to note that when a control was available, the most commonly activated control was the midrange level, followed by the treble level and bass tilt. In addition, broadband flatness was

improved by equalisation, mainly because extreme magnitude values at low frequencies were reduced (LF subband).
The lack of improvement in midrange and high frequencies (MF and HF subbands) is because the room related response variation in these frequencies is narrow-band. Somewhat better equalisation could be obtained by using controls offering response tilting or shaping within these frequencies.
The largest variation in the improvement at low frequencies can be explained by listening rooms acoustics[43]. At low frequencies, radiation from a loudspeaker is typically omnidirectional and the sound field is usually not diffuse. This results in strong room effects and hence large variations in the magnitude response at these frequencies.

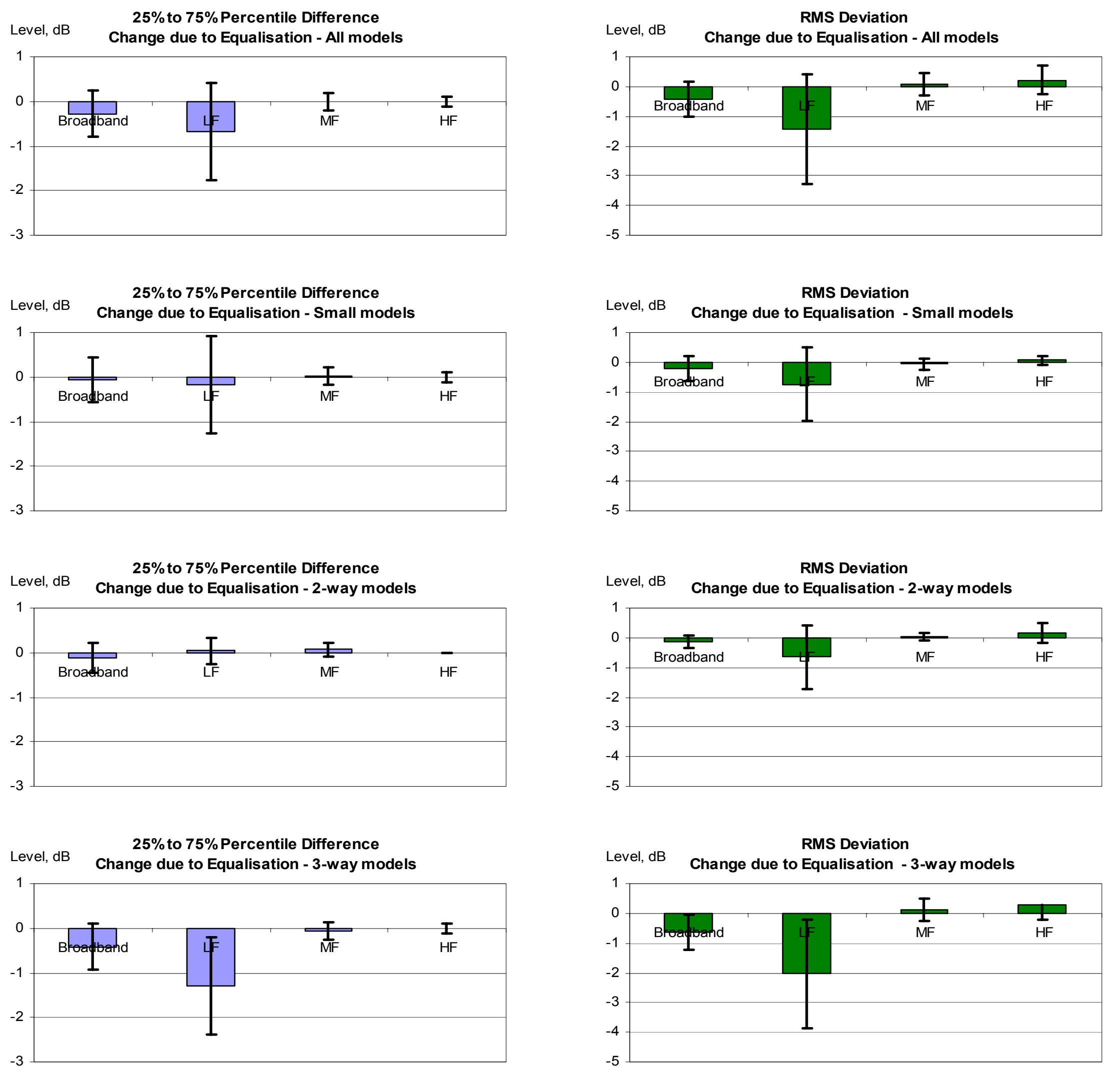

Figure 4. Change in sound level deviation due to equalisation. For each subband, quartile difference and RMS deviation from the median. The error bar indicates the standard deviation.

The largest improvement seen for three-way systems can be explained by two main factors. Firstly, rooms in which these loudspeakers are installed typically have a high quality acoustical design, producing a well-controlled sound field. Smaller loudspeakers are often installed in rooms with little or no acoustical control, making response correction by equalisation a very challenging task. Secondly, three-way systems have more room response controls than two-way systems, with a higher capability to compensate for room problems. The type of equalisation the room response controls are designed for is a gentle shaping of the response. High order narrow band corrections are not possible, and therefore room characteristics and the quality of its acoustical design will always play a major role.

6 CONCLUSIONS

The low-order room response adjustment filters in active loudspeakers can significantly improve the perceived quality of audio reproduction. The automated optimisation algorithm presented in this paper is used to select an optimal combination of filter settings in loudspeakers where a room response equaliser is implemented as a filter having discrete-valued settings. The algorithm proves to be useful because it performs systematically with varying types of loudspeakers, with differing filter sets in multiple types of acoustical installations. The efficiency and reliability of the algorithm has been achieved by exploiting heuristics of experienced sound system calibration engineers. The automated methodology obtains systematically and consistently the best combination of available filter settings, performing quickly irrespective of the operator. The algorithm has been implemented in a loudspeaker calibration tool used by specialists who set up and tune studios and listening rooms.

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APPENDIX A – SOFTWARE GRAPHICAL USER INTERFACE

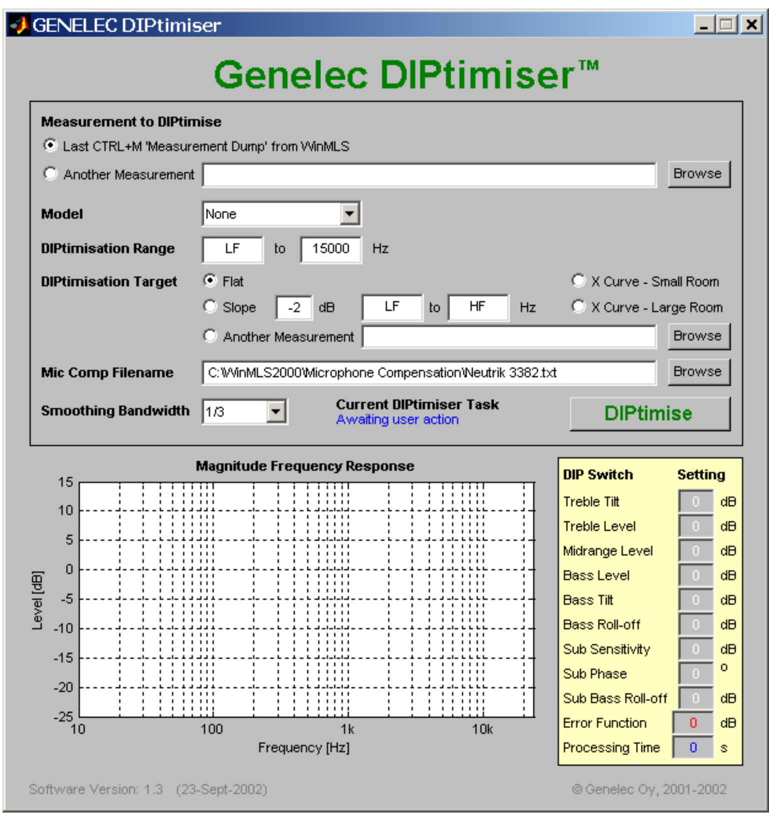

Figure 5. Software graphical user interface at start up.

APPENDIX A – SOFTWARE FLOW CHART

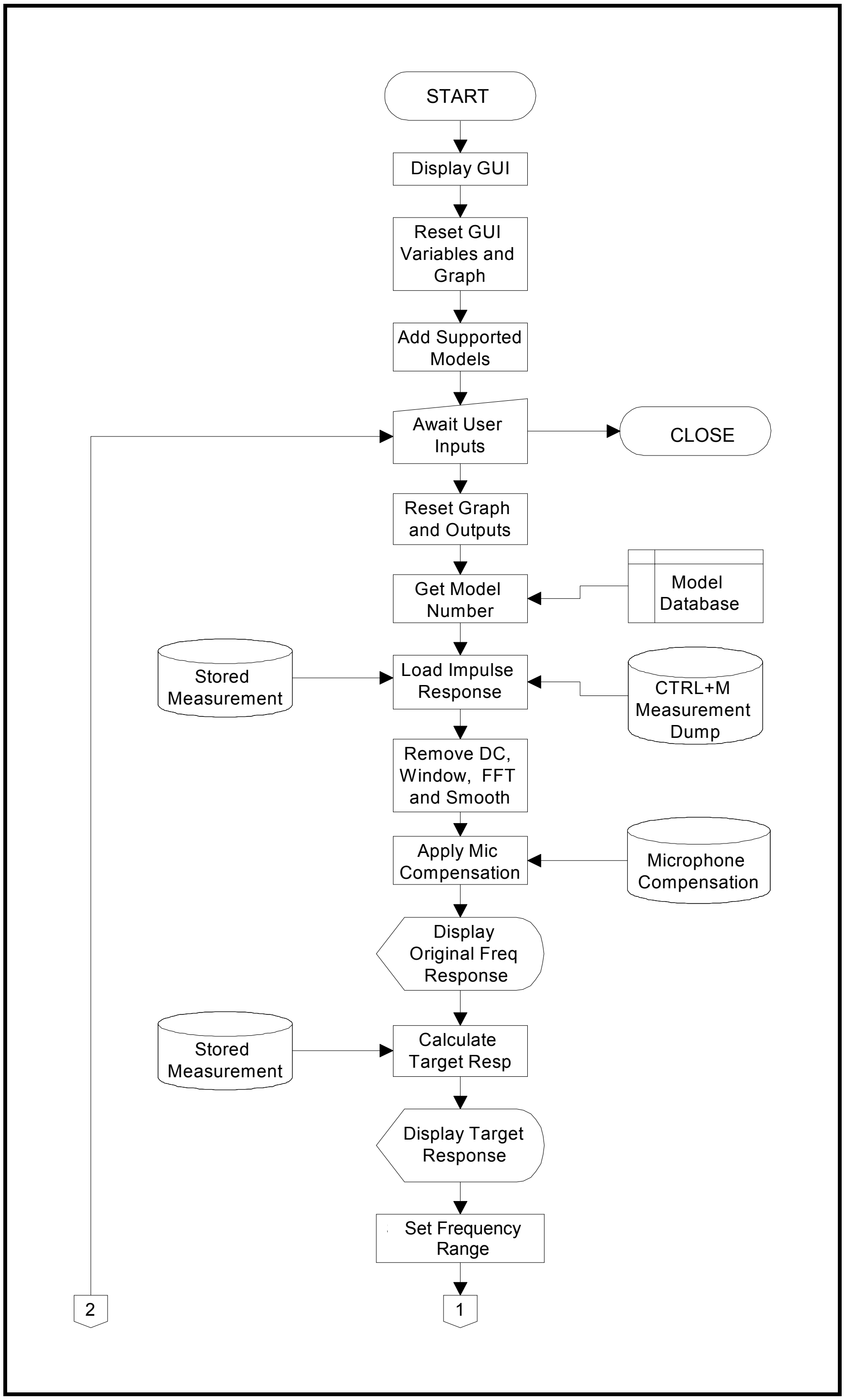

Figure 6. Software flow chart, part 1.

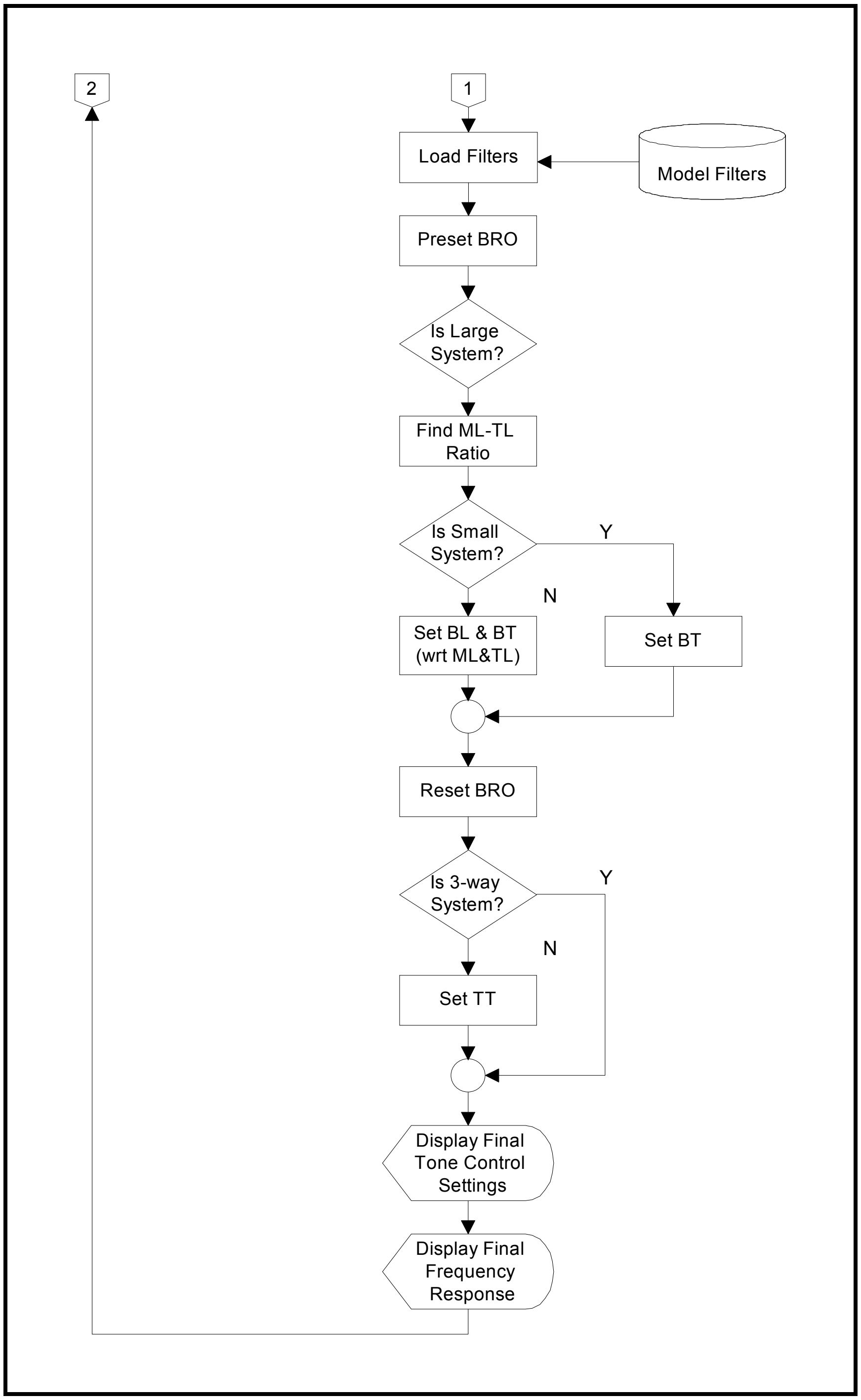

Figure 6 continued. Software flow chart, part 2.