Statistical Analysis of an Automated In-Situ Frequency Response Optimisation Algorithm for Active Loudspeakers

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ABSTRACT

This paper presents a novel method for automatically selecting the optimal in-situ acoustical frequency response of active loudspeakers within a discrete-valued set of responses offered by room response controls on active loudspeakers. The rationale of the room response controls for the active loudspeakers is explained. The frequency response, calculated from the acquired impulse response, is used as the input for the optimisation algorithm to select the most favourable combination of room response controls. The optimisation algorithm is described. The performance of the algorithm is analysed and discussed. This algorithm has been implemented and is currently in active use by specialist loudspeaker system calibrators 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 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 active loudspeakers. Even with experienced system calibrators, significant variance between calibrations can be seen. Furthermore, with a number of different people calibrating loudspeaker systems, additional variance in results will occur. For these reasons an automated calibration method was developed to ensure consistency of calibrations.

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. The goal of in-room 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].

An approach to improve the performance of a loudspeaker 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 an optimisation may not be practically possible in all cases and that this is only half of the installation process, as the loudspeaker should be corrected for problems caused by the loudspeaker-room interaction too.

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 sound reproduction applications such as recording studios, mixing rooms and sound reinforcement.

In-situ response equalisation is typically implemented using a separate equaliser, although equalisers are increasingly built into active loudspeakers. 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 [5] 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 applying an equalisation designed using only a single point measurement. 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. Examples of spatial averaging have been described in the automotive industry [6] and cinema in the SMPTE Standard 202M [7]. Spatial averaging can reduce local variance in midrange to high frequencies and can also 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 to be 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 equalisation filters, 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 have a link to the perception of sound being 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 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 causing timber changes in the perceived audio.

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).

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

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.

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). Figure 5 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.

	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,

$$\min_m E = \left| \frac{\max\limits_{f_a}\left(\dfrac{a_m(f)x(f)}{x_0(f)}\right)}{\max\limits_{f_b}\left(\dfrac{a_m(f)x(f)}{x_0(f)}\right)} \right|,$$

$$f_a = [f_1, f_2],\ f_b = [f_2, f_3] \qquad (2)$$

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,

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

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.

Table 9. Number of filter evaluations needed 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 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%.

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 strong cancellation in the frequency response around f_{LF}, or the high frequency level is decreased significantly 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.

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 frequen-

cies 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 loudspeaker and optimise this to the optimised left loudspeaker response. The result will be the closest match possible between the left and right loudspeaker 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 [7]. 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 [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.[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 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.

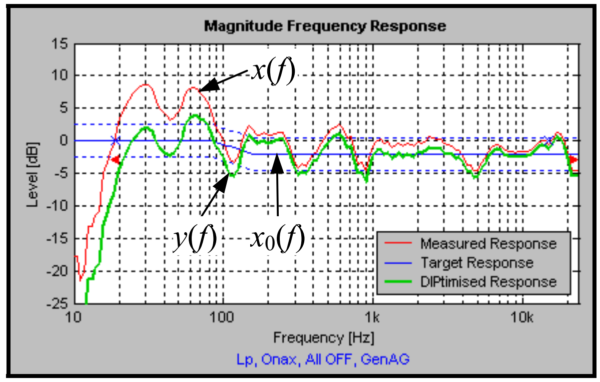

Figure 1. Typical graphical output of the optimiser software. Original response $x(f)$, target response $x_0(f)$ and final response $y(f)$. Also, –3 dB cut-off frequencies (triangles), optimisation range (crosses) and target tolerance (dotted).

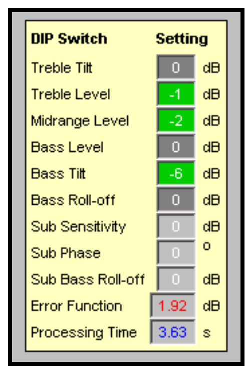

Figure 2. Output section displays all settings and values to be changed (green background) as well as the value of the error function and processing time.

4. PERFORMANCE OF THE OPTIMISATION ALGORITHM

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 algo-

[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.

rithm 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

A further statistical analysis was conducted on all of the loudspeakers in the study. The bandwidths of the frequency bands used are shown in Table 11. The bandwidths '*LF*', '*MF*' and '*HF*' are later referred to collectively as the '*subbands'* and correspond roughly to the bandwidths for each driver in the three-way systems.

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

For each loudspeaker, the broadband (Table 11) magnitude response data median value is standardised to 0 dB.

The statistical descriptors recorded before and after equalisation for each loudspeaker and in each frequency band defined in Table 11 are the minimum, maximum and range of the magnitude dB values. Also for the magnitude pressure values in each bandwidth (Table 11), the median, 5% & 95% percentiles and quartiles are recorded. In addition, the root-mean-square (RMS) deviation of the pressure from the median in each bandwidth is calculated: the value is expressed in dB.

These statistical descriptors are compared for each subband to study the in-band flatness improvement due to equalisation. The median values for each subband are compared to study the broadband tonal balance improvement. This is indicated by a reduction of the median value differences.

4.2. Example of Statistical Data Analysis

Figure 7 in Appendix C shows a case example where room response control settings are calculated according to the optimisation algorithm. The equalisation target is a flat magnitude response (straight line at 0 dB 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.

Figure 8 in Appendix C 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 distribution 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.

4.3. Results

A total of 63 loudspeakers were measured before and after equalisation. Of these, 12 were small two-way systems, 22 were two-way systems, 30 were three-way systems and three were 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 the controls were used when available on the loudspeaker. The midrange level control is used 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 gives the 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. Both the quartile difference and RMS deviation values represent two slightly different ways to look at the deviation from the median value of the distribution. The quartile limits are more robust to outlier values while the RMS values include these effects.

For small two-way systems (Figure 9-10), the main improvement is seen at low frequencies in four out of 12 cases. In only 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) 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 and there are no changes in the HF subband. The flatness in the broadband and LF subband of the 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 a clear reduction in most cases of both the quartile difference (Figure 13) and RMS deviation (Figure 14) for the broadband and LF subband. Slight, and equal numbers of, increases and reductions are seen for MF and HF subbands.

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

Some of the responses appeared to become worse in the quartile difference and RMS deviation in the subband analysis. This was not reflected in the broadband metrics, which indicates that the arbitrary subband frequency division introduced some of the error. Also, the cases where this happened suffered from severe anomalies within the pre-equalisation response due to extremely bad room acoustic conditions. The equalisation was not designed to compensate for this.

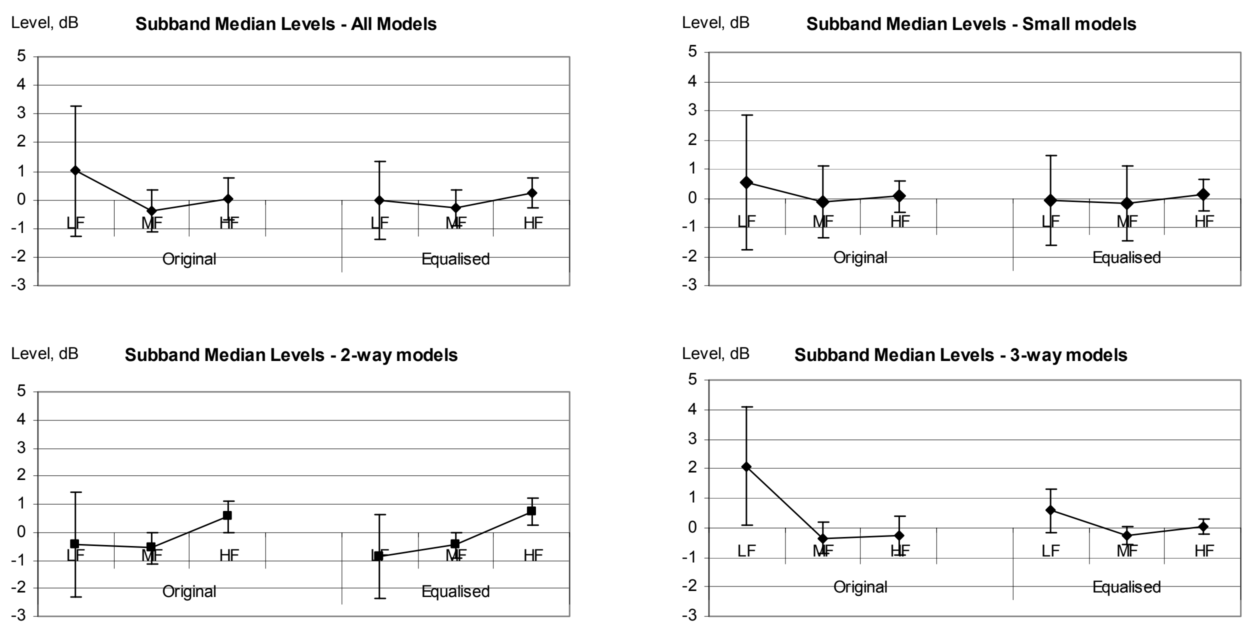

Figure 3. Mean and standard deviation of subband median levels before and after equalisation.

The subband median levels (Figure 3) illustrate the broadband frequency balance between the subbands. Loudspeaker loading from nearby boundaries is reflected in the LF subband median level before equalisation, especially in the often flush mounted three-way models. The median level in the LF subband is reduced after equalisation, which indicates that equalisation compensates well for the loudspeaker loading. A better match across subbands of the average subband median level demonstrates that equalisation has improved the broadband flatness. The largest improvement is seen in the three-way loudspeakers. In the two-way systems the equalisation has improved broadband flatness only marginally as the subband median levels do not show major signs of change. The broadband flatness improvement is mainly the result of better alignment of the LF subband with the MF and HF subbands. This indicates that the equalisation has not only reduced the variation inside individual subbands but also improved the broadband flatness of the acoustical response, translating to a reduced colouration of the audio at the listening position.

For all loudspeakers pooled together (Figure 3), the equalisation reduces the variance in the median level for the LF subband. A similar outcome is noted separately for each loudspeaker type. However, only in the three-way systems is an improvement seen also in the MF and HF subband variance.

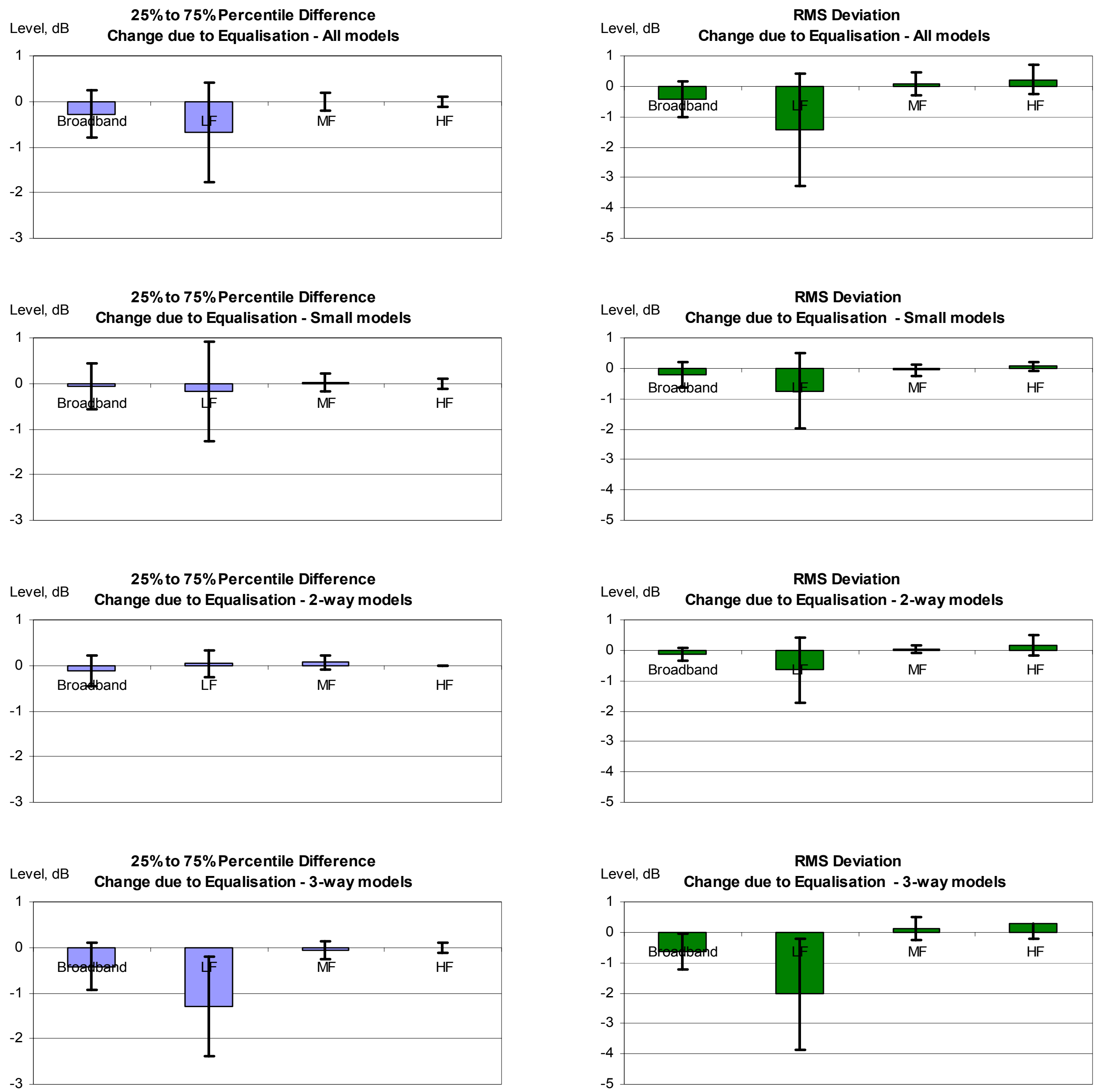

Figure 4. Change in sound level deviation due to equalisation. For each subband, quartile difference and RMS deviation from the median. The error bar indicates the standard deviation.

In Figure 4 the results are pooled for all products and for each product type, excluding the main monitors where there were only three cases. The change in quartile difference and RMS deviation for the broadband and the subbands is illustrated. For all models, the broadband flatness is improved by 0.4 dB and the mean reduction in the LF subband RMS deviation is 1.4 dB. The RMS deviation for all models pooled together has been reduced by equalisation and the largest reduction is seen for the three-way systems. To some extent this is similar for the quartile difference but the small two-way and two-way systems do not see such large improvements due to equalisation. This indicates that the improvement is mainly a reduction of extreme magnitude values (heights of peaks and notches) in the low frequency response.

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 show that it is effective.

The room response controls in active loudspeakers implement discrete filter parameter values rather than a continuous parameter value range. The number of possible filter parameter value combinations can be quite large and so 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 filter parameter 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 the 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 relative to a full exhaustive search.

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 [41]. The optimisation is efficient and so the software is 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 which are not intended to correct for narrow-band deviations in the frequency response. 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 the broadband frequency balance relative to the target response and by reducing the variability in the response, particularly in the low frequencies. Across all loudspeaker groups the main improvement is in the reduction of extreme (outlier) values in the low frequency band of the response.

It is interesting to note that the most commonly used room response control was the midrange level, followed closely by the treble level and bass tilt control. This is explained by the fact that the algorithm in most stages minimises the RMS deviation, and in so doing affects most efficiently the extreme deviations from the median level.

For all models pooled together, the broadband flatness was improved by equalisation. This improvement is mainly due to a reduction of the extreme magnitude values (heights of peaks and notches) in the low frequency response (LF subband).

The lack of improvement in the quartile values and RMS deviation in the midrange and high frequencies (MF and HF subbands) is because the room related response variation becomes narrow band. Some improvement in the equalisation could be obtained with room response controls offering a tilting or shaping of the response within the mid-to-high frequency range.

The largest variability of the improvement in the low frequency range can be explained by the acoustics found in listening rooms [42]. At low frequencies the radiation from the loudspeaker can be considered omnidirectional and the sound field in the room is usually not very diffuse. This results in strong room effects and hence large variations in the magnitude response at these frequencies.

The largest improvement is seen for the three-way systems and can be explained by two main factors. Firstly, the rooms in which this type of loudspeaker is typically installed are of a higher quality acoustical design, so the sound field in them is well controlled. Conversely, smaller loudspeakers are often installed in rooms with little or no acoustical design, making corrections to the response by equalisation a very challenging task. Secondly, the three-way systems contain more room response controls than the two-way systems, which gives a higher capability for the equaliser to compensate for room problems. It should also be noted that 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, therefore the characteristics of the room and the quality of its acoustical design will 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 the optimal combination of settings for loudspeakers where the room response equaliser is implemented as a filter set with discrete parameter values. The algorithm proves to be useful because it performs systematically with widely varying types of loudspeakers, with slightly differing filter sets and loudspeakers found in multiple types of 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 filters, and performs 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 FLOW CHART

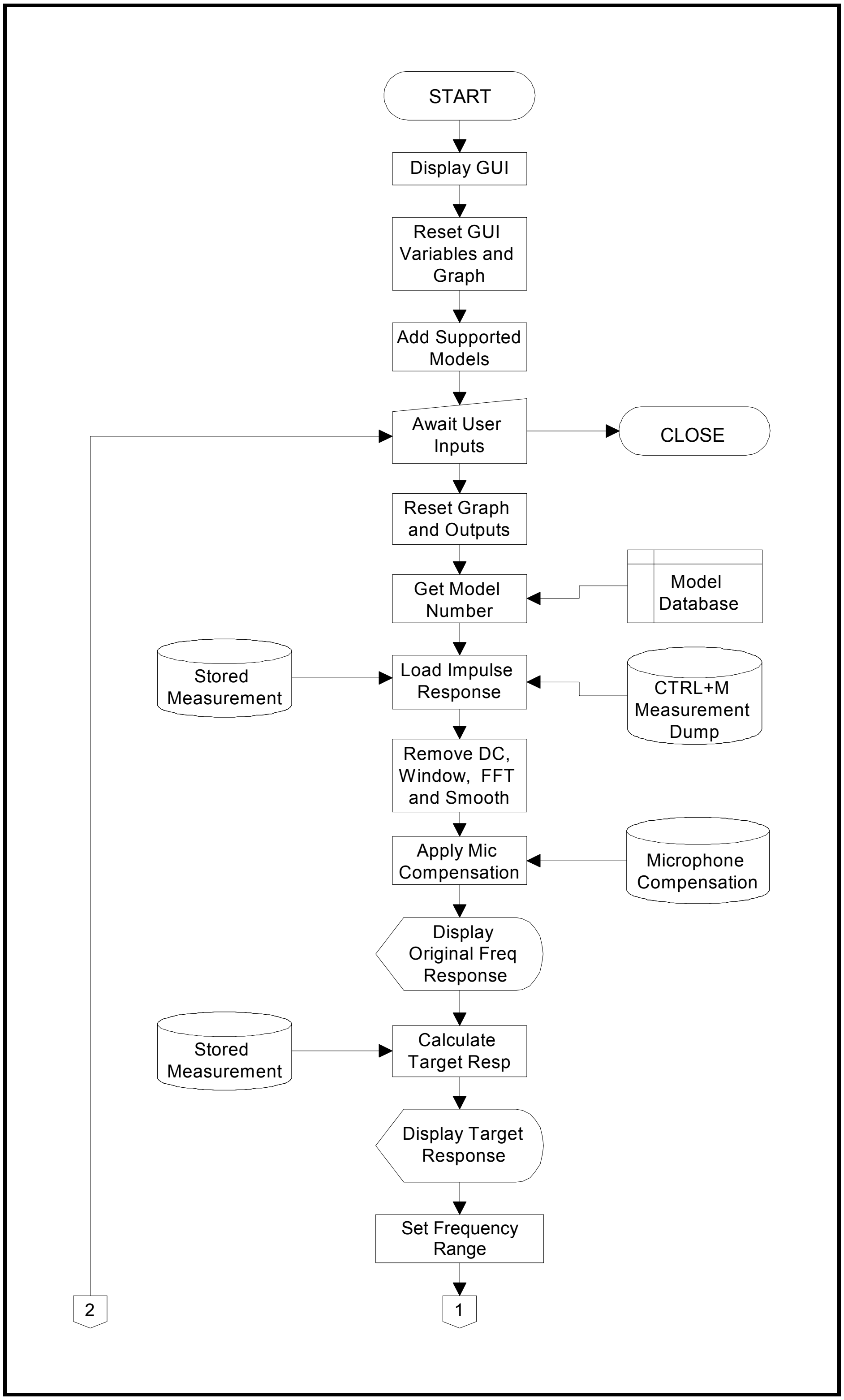

Figure 5. Software flow chart, part 1.

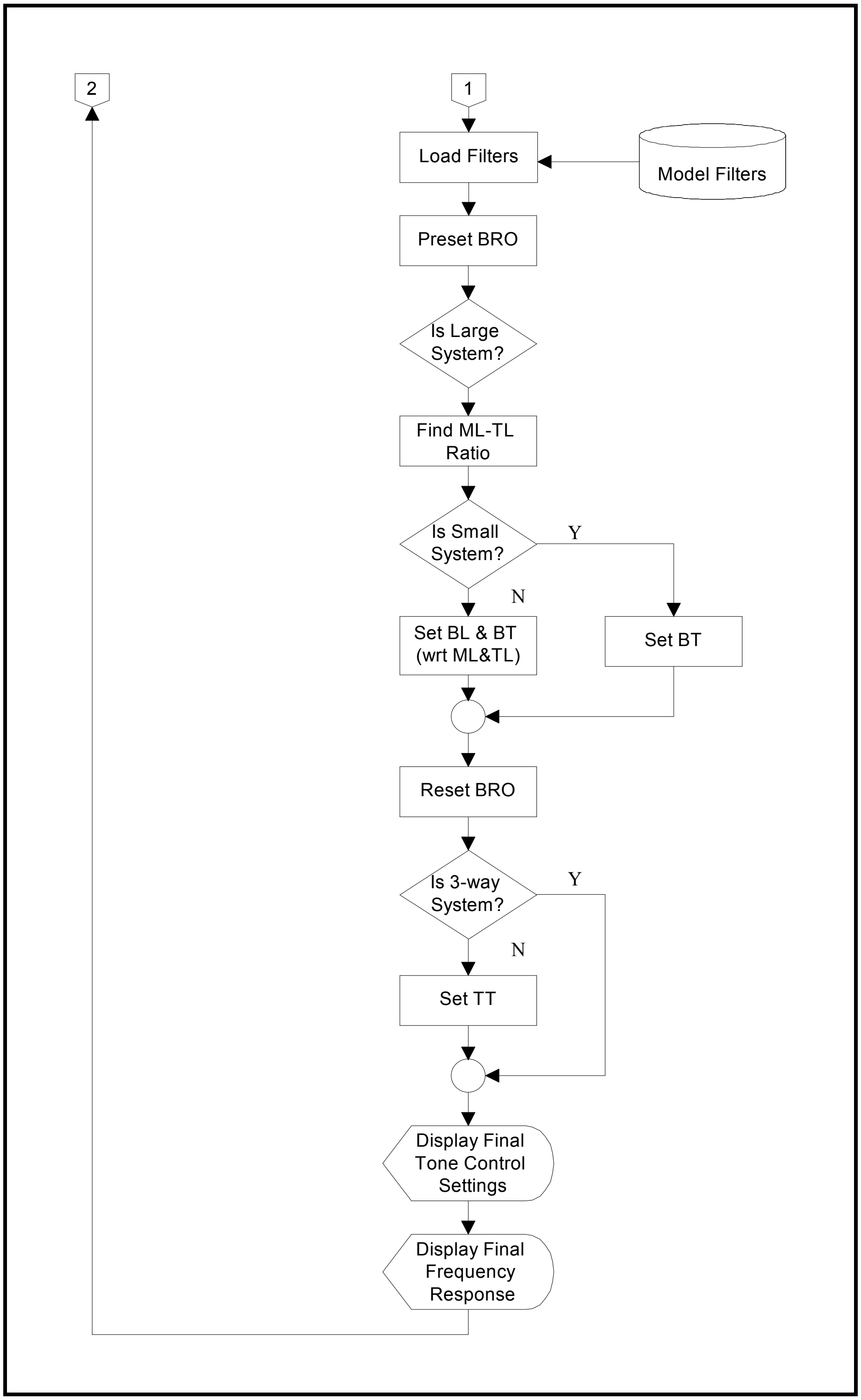

Figure 5 continued. Software flow chart, part 2.

APPENDIX B – SOFTWARE GRAPHICAL USER INTERFACE

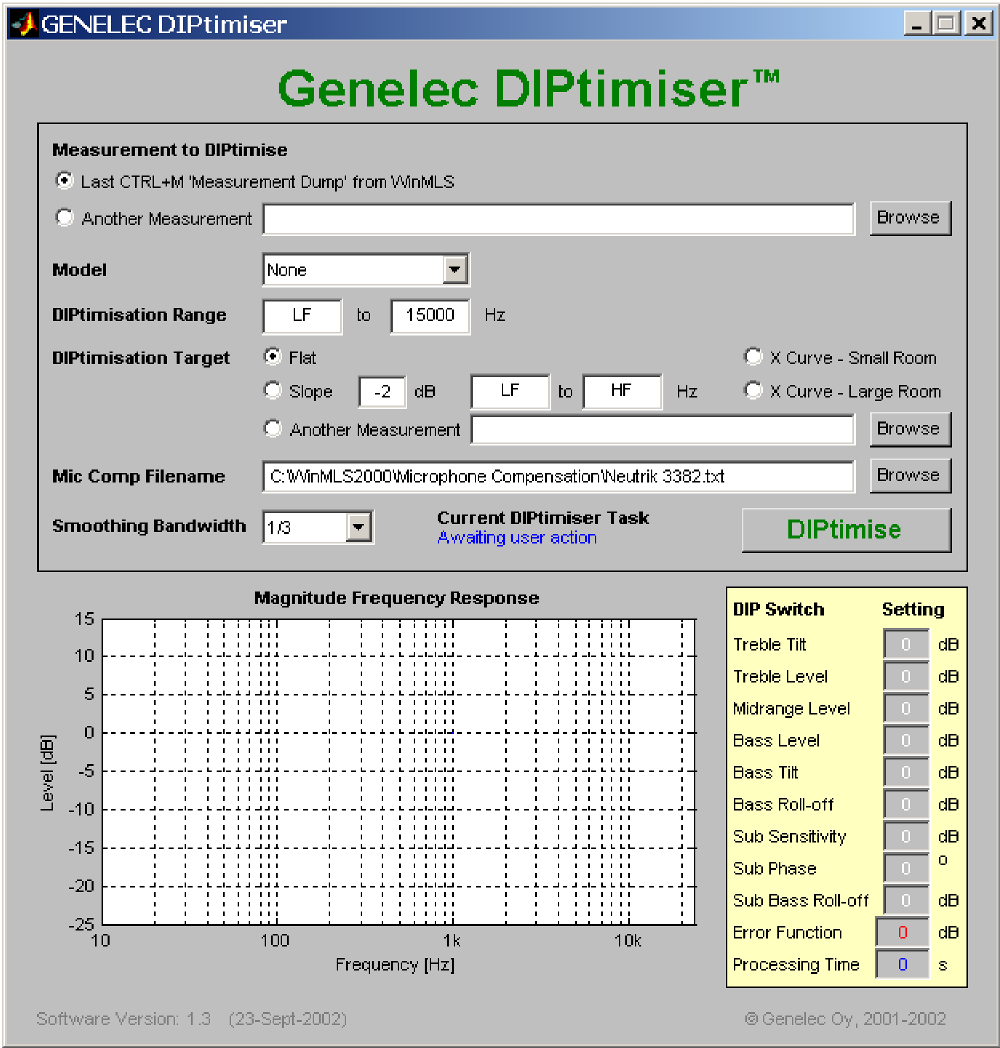

Figure 6. Software graphical user interface at start up.

APPENDIX C – CASE EXAMPLE, STATISTICAL GRAPHS

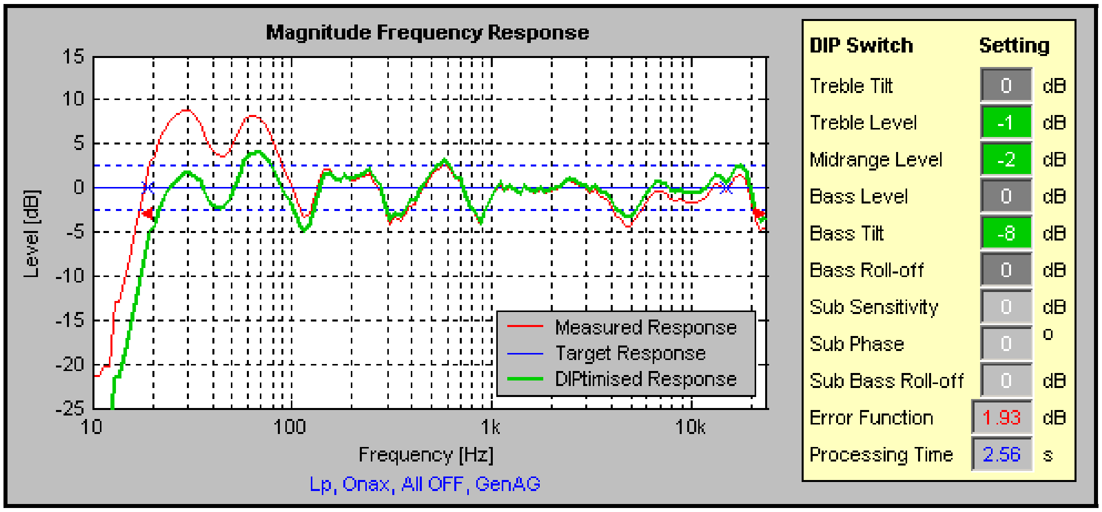

Figure 7. Case example, optimisation results.

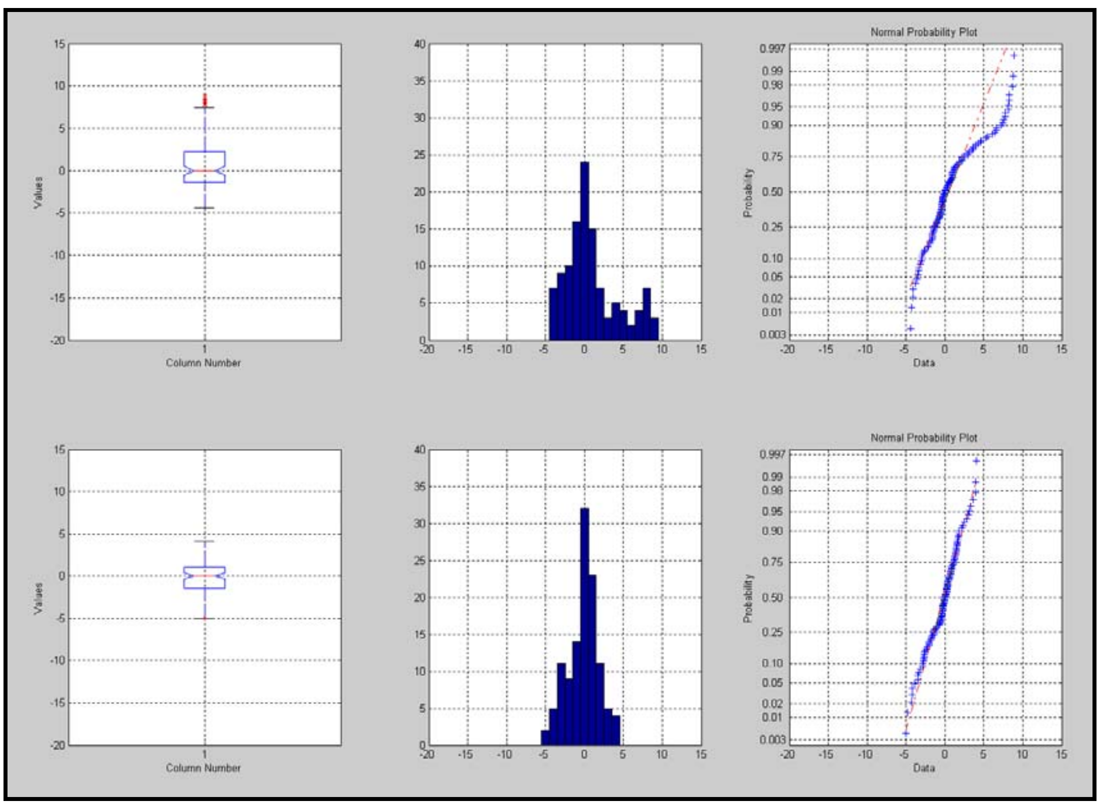

Figure 8. Case example, statistical output.

APPENDIX D – STATISTICAL GRAPHS

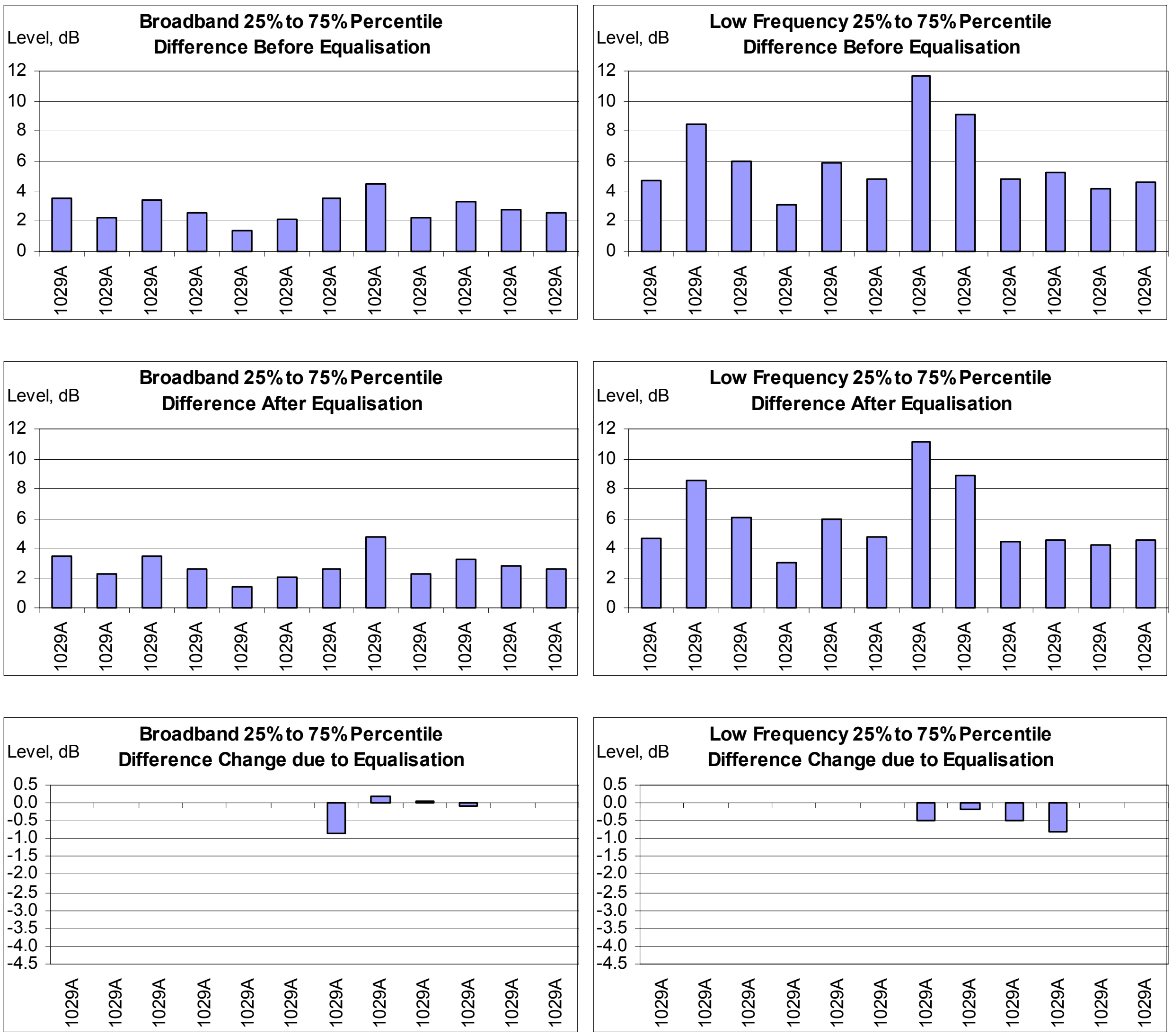

Figure 9a. 25% to 75% percentile differences for small 2-way systems.

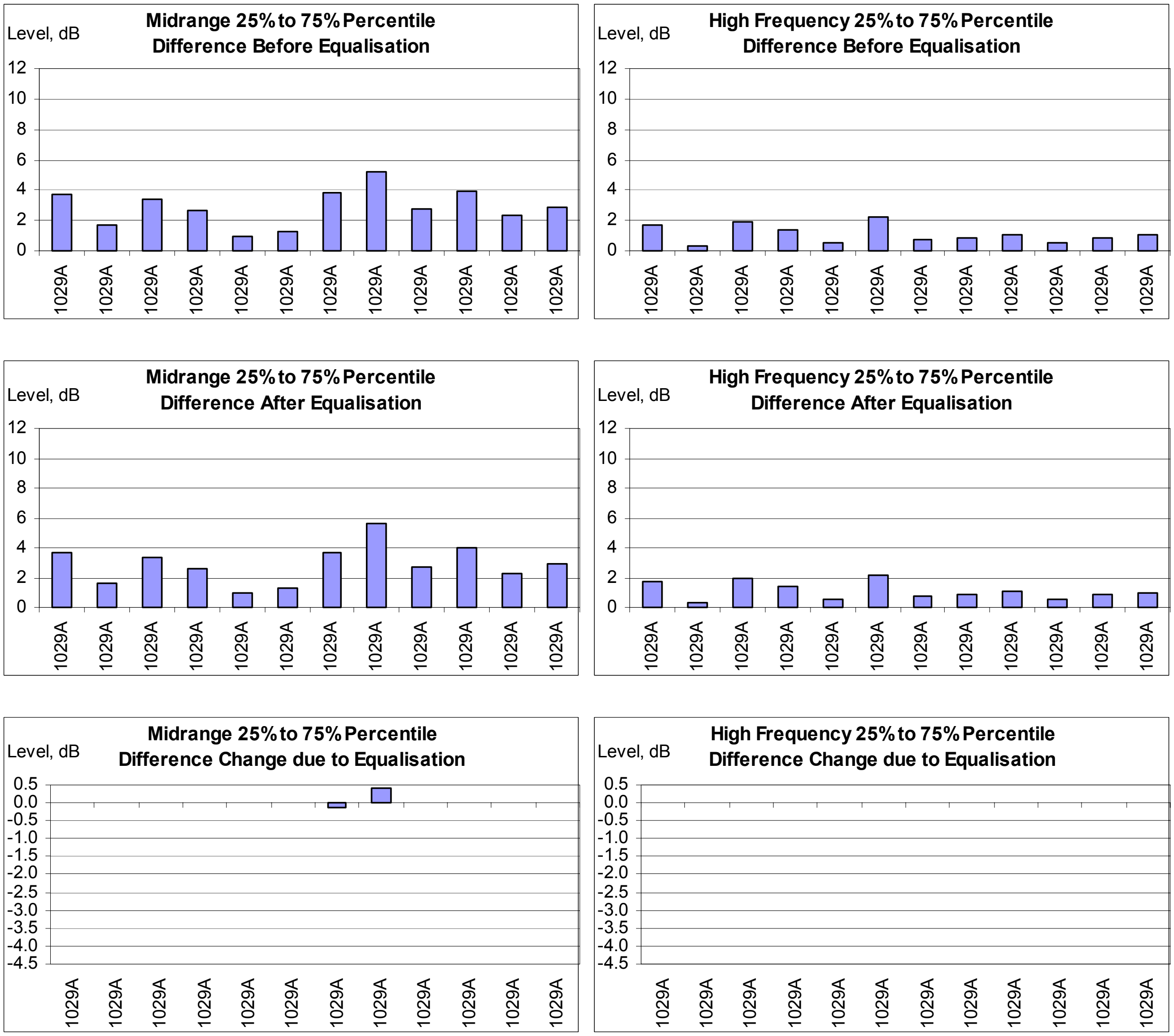

Figure 9b. 25% to 75% percentile differences for small 2-way systems.

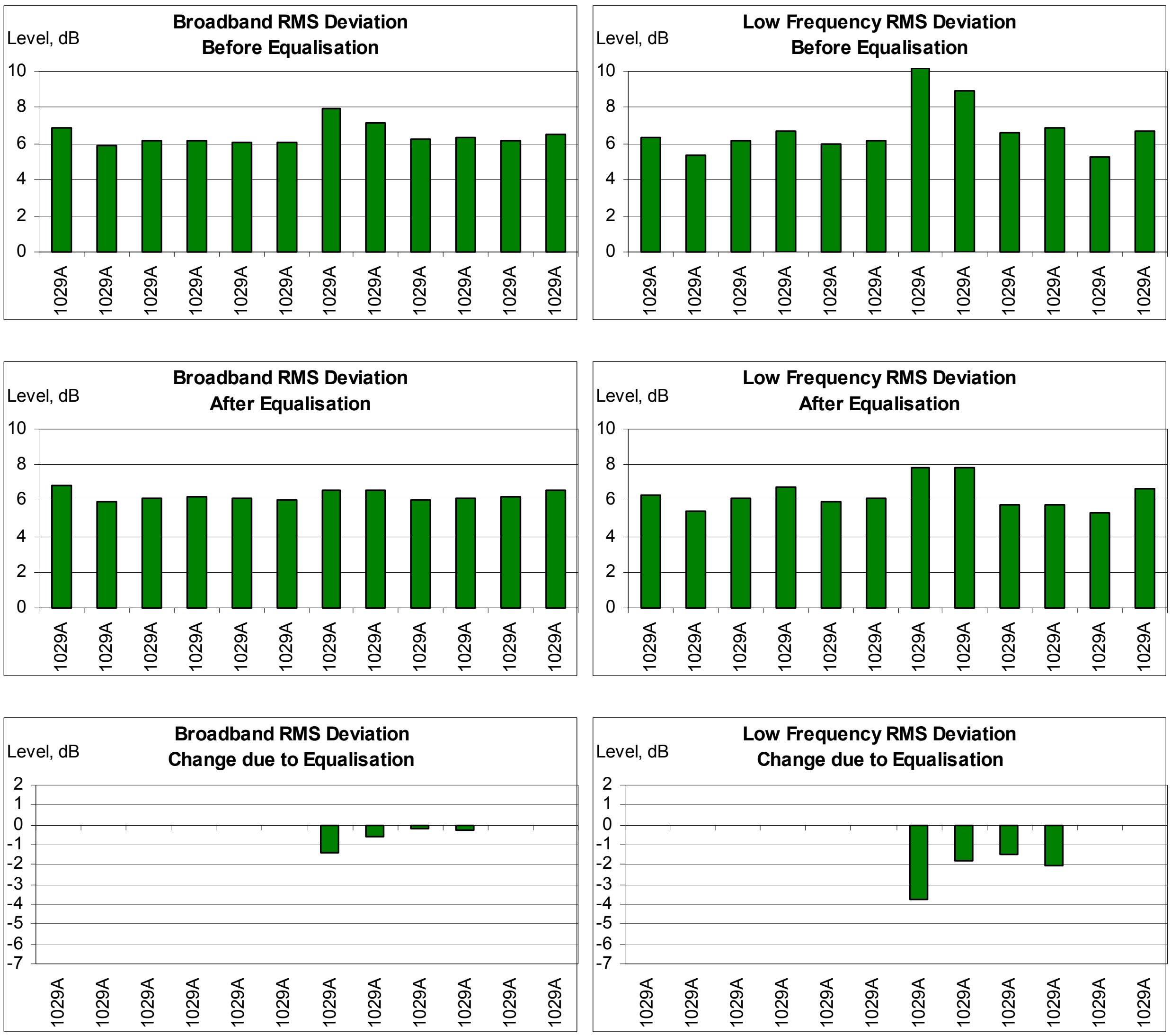

Figure 10a. RMS deviation for small 2-way systems.

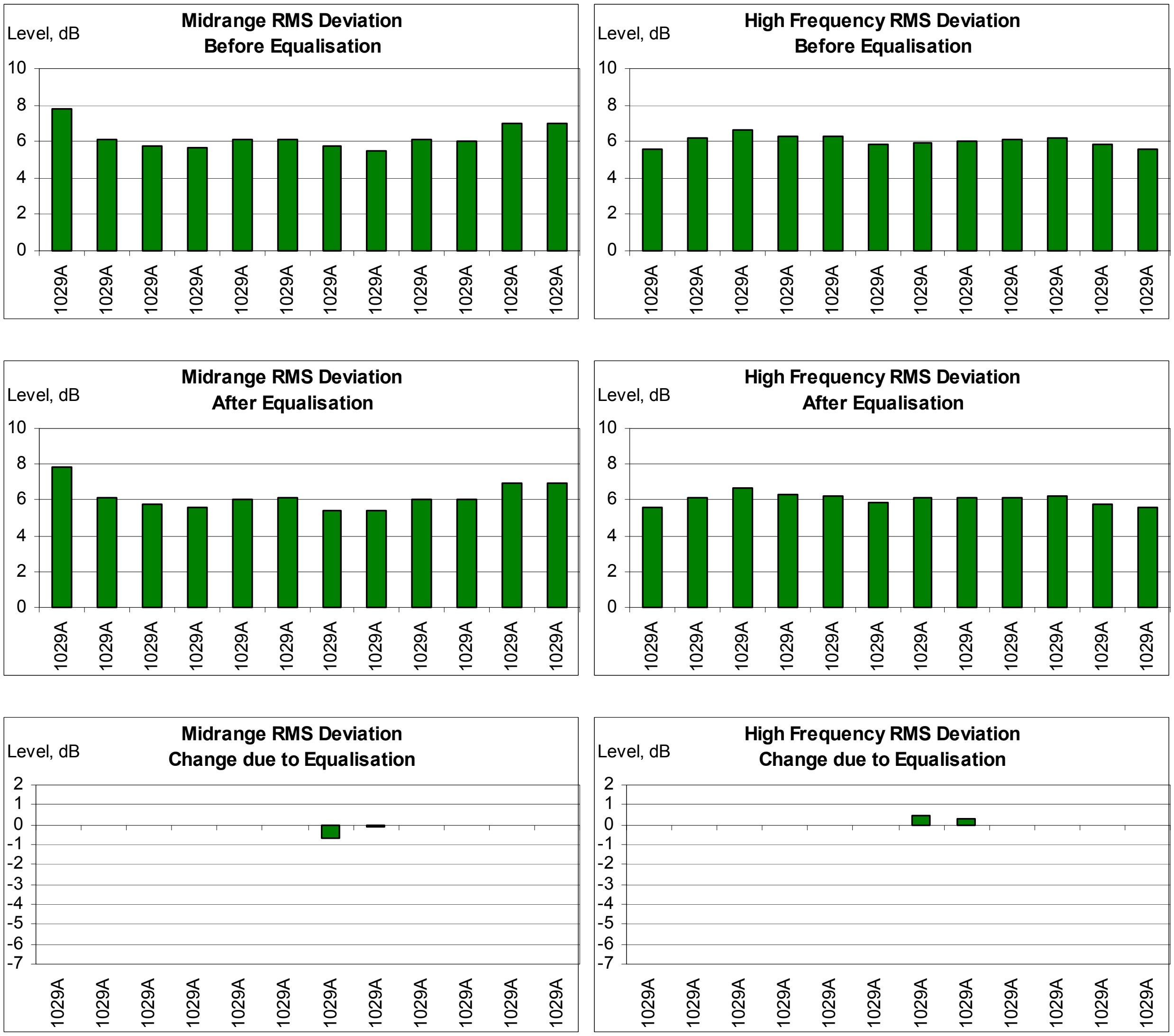

Figure 10b. RMS deviation for small 2-way systems.

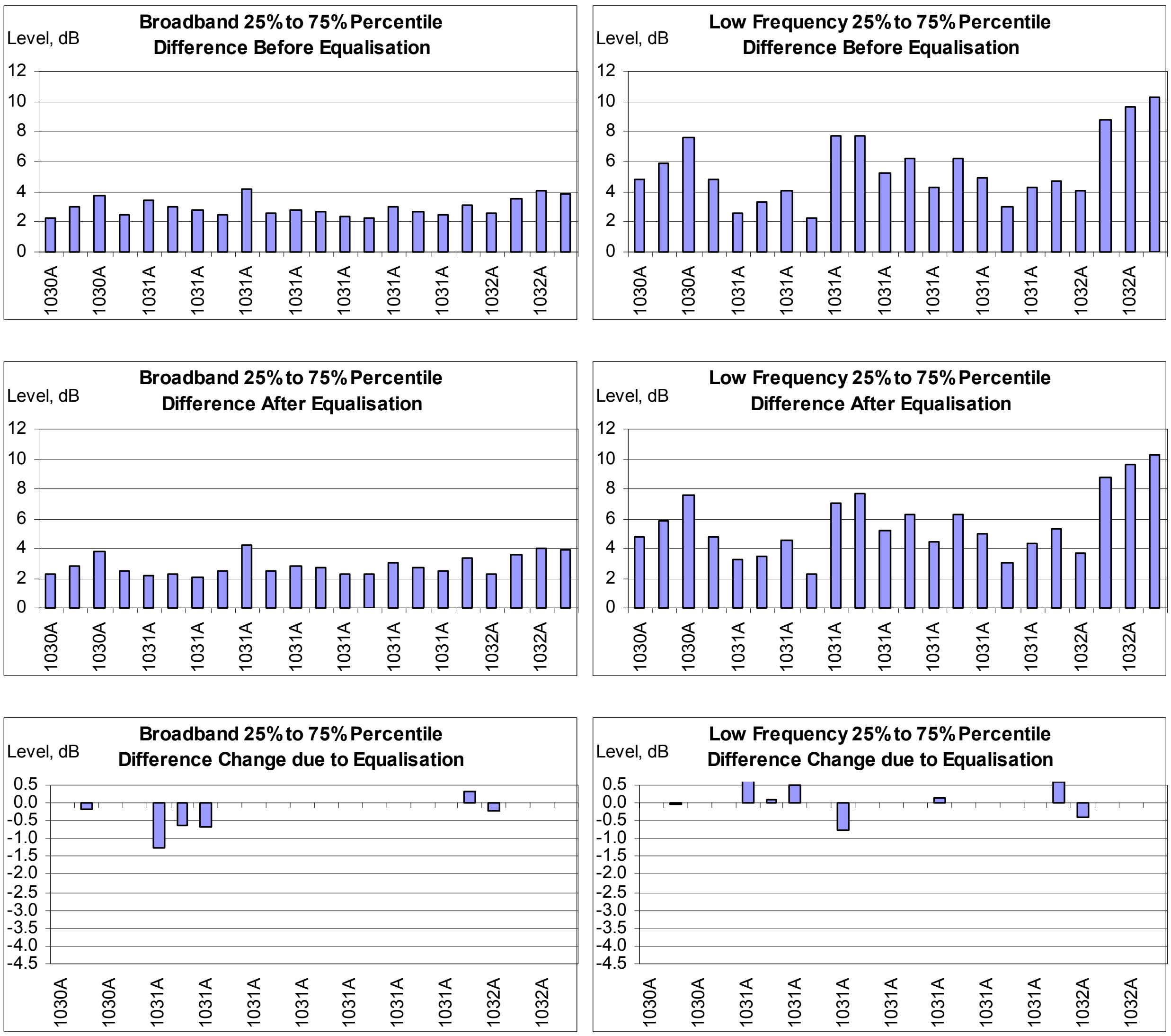

Figure 11a. 25% to 75% percentile differences for 2-way systems.

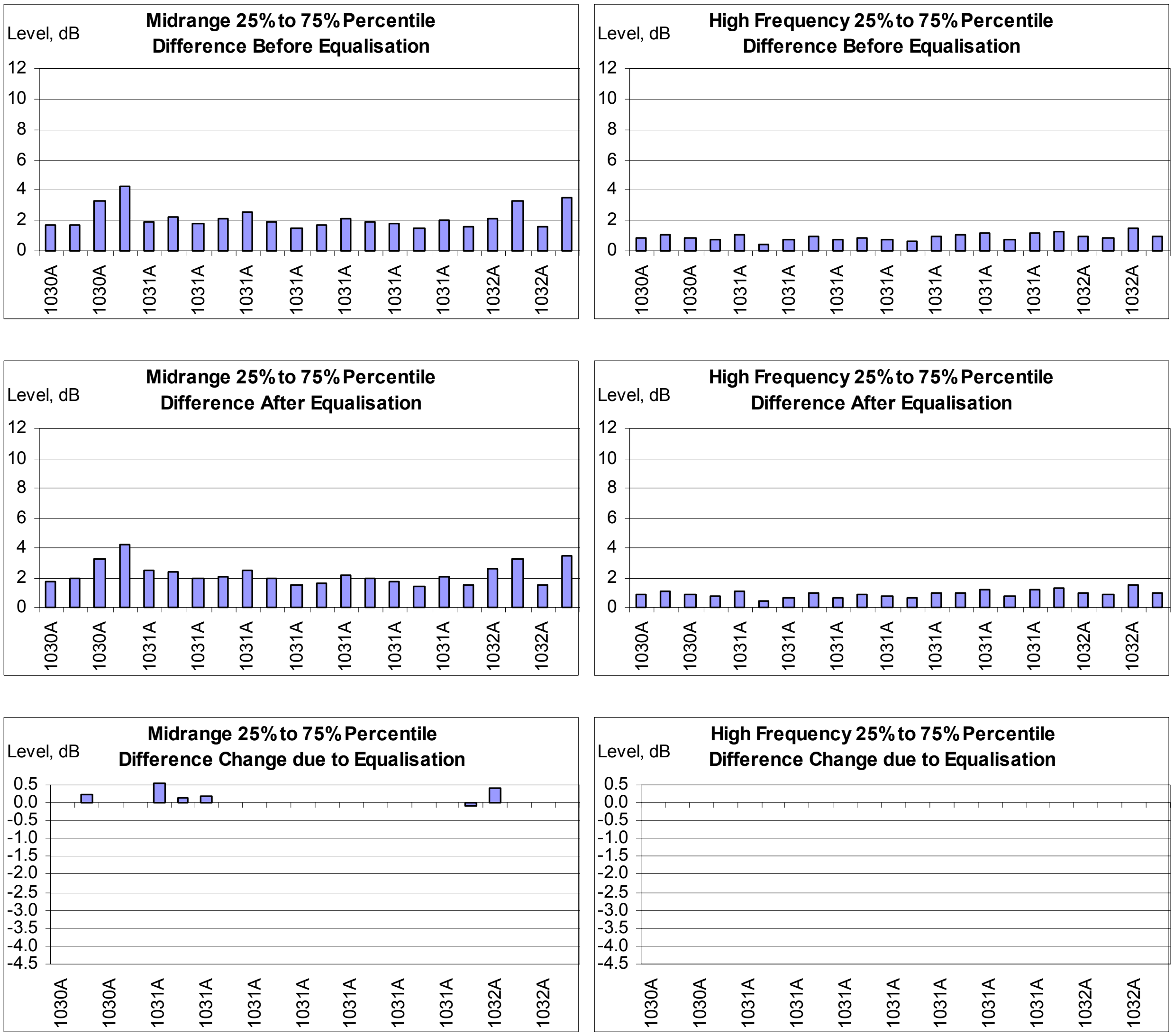

Figure 11b. 25% to 75% percentile differences for 2-way systems.

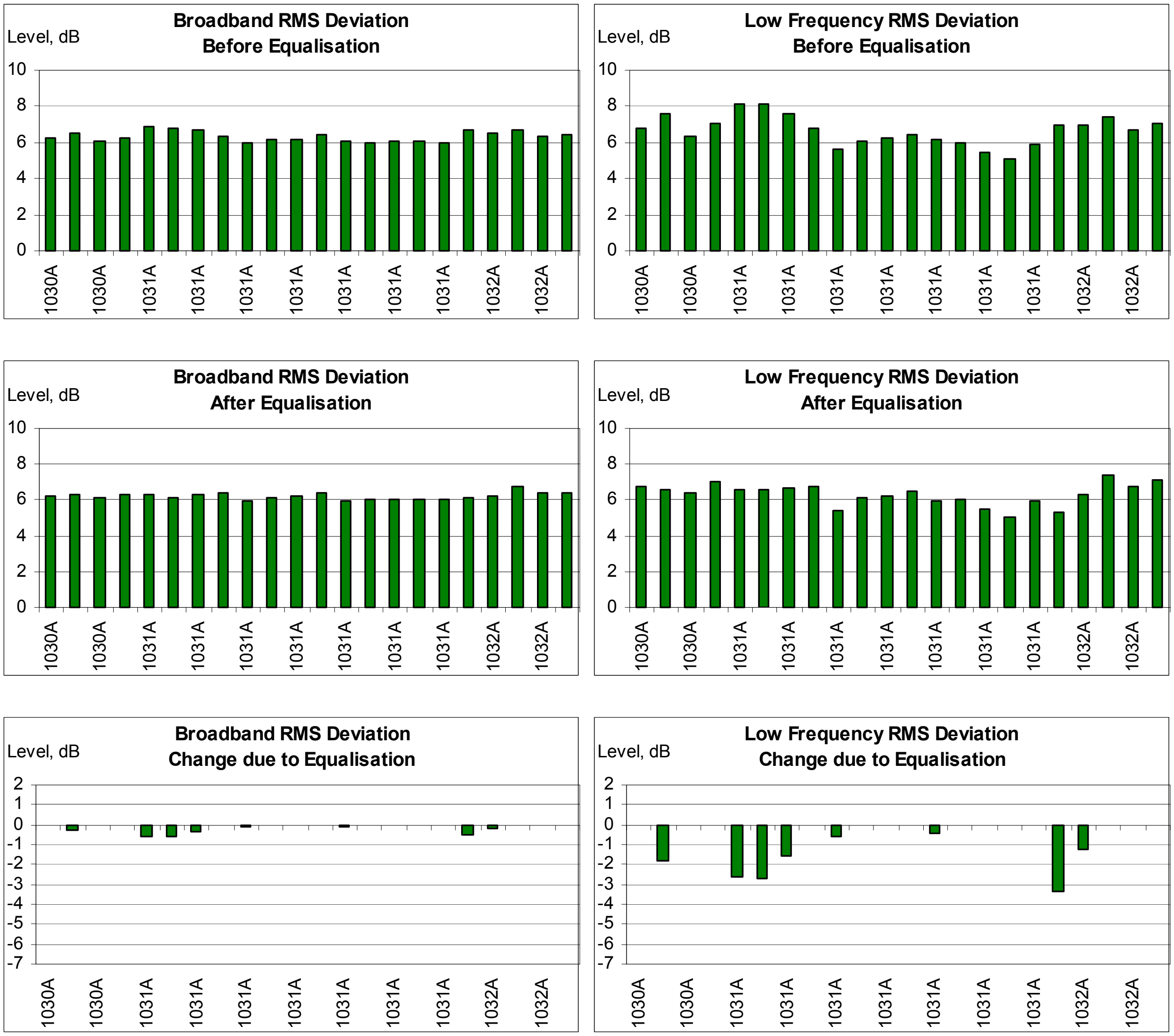

Figure 12a. RMS deviation for 2-way systems.

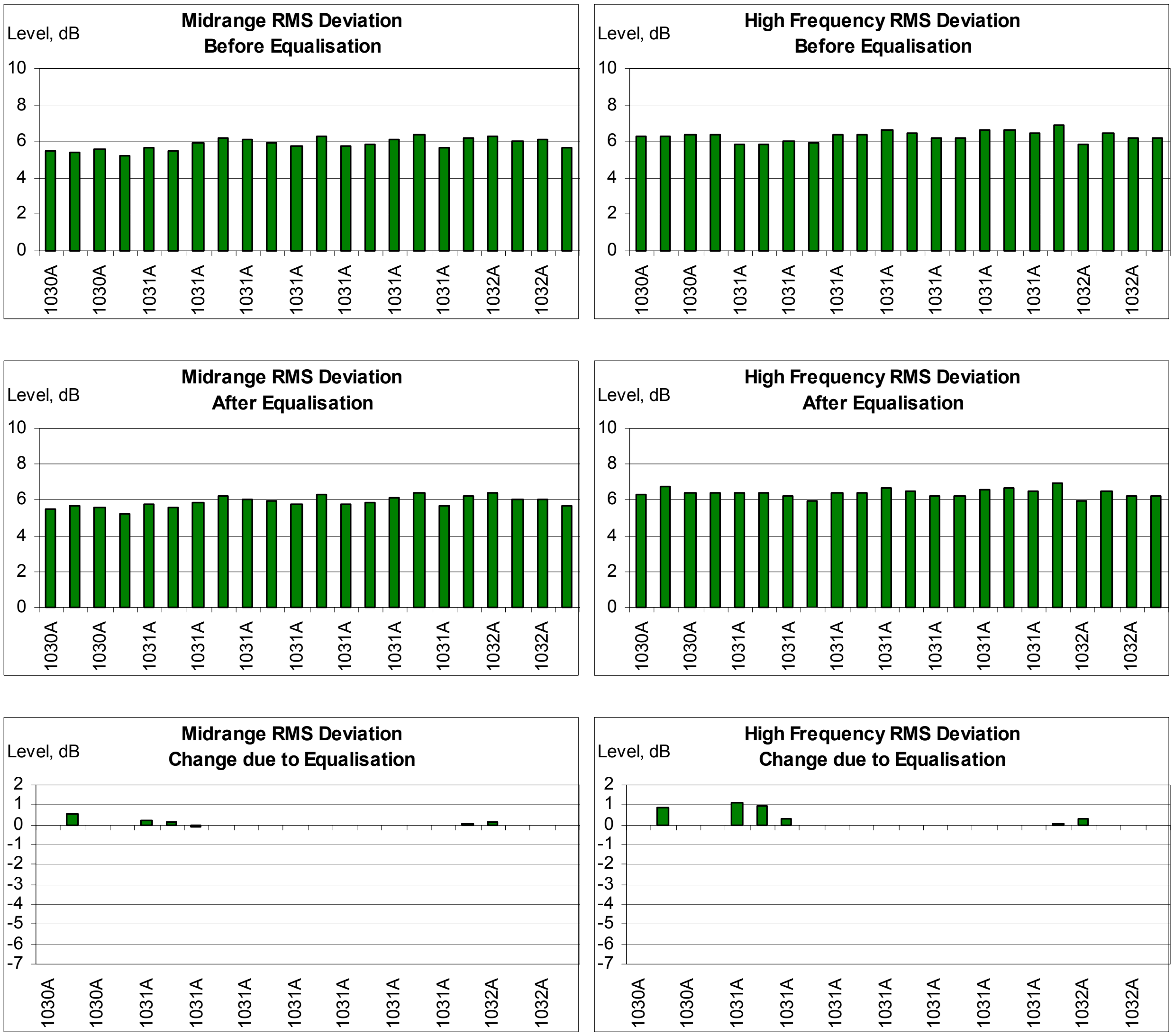

Figure 12b. RMS deviation for 2-way systems.

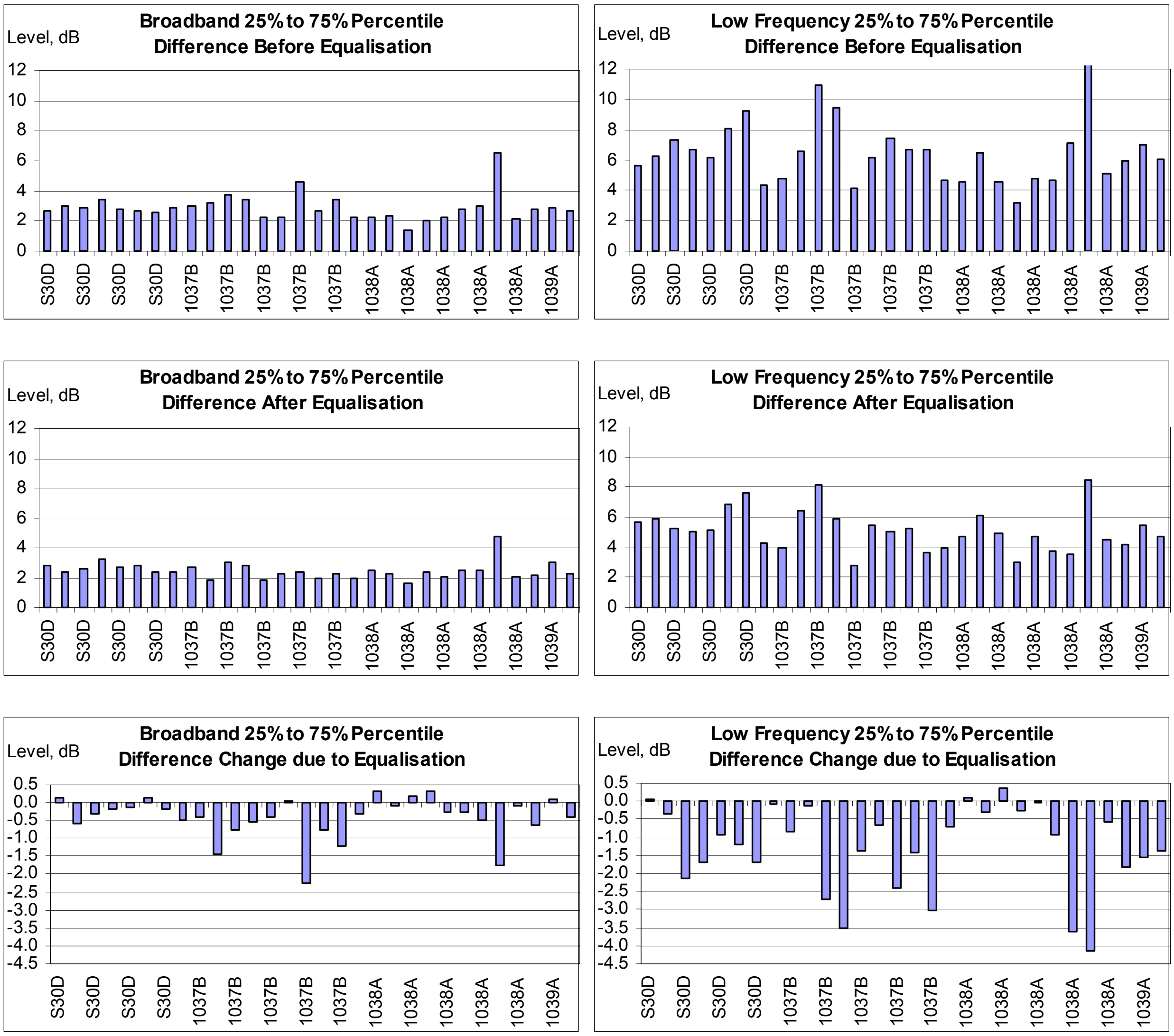

Figure 13a. 25% to 75% percentile differences for 3-way systems.

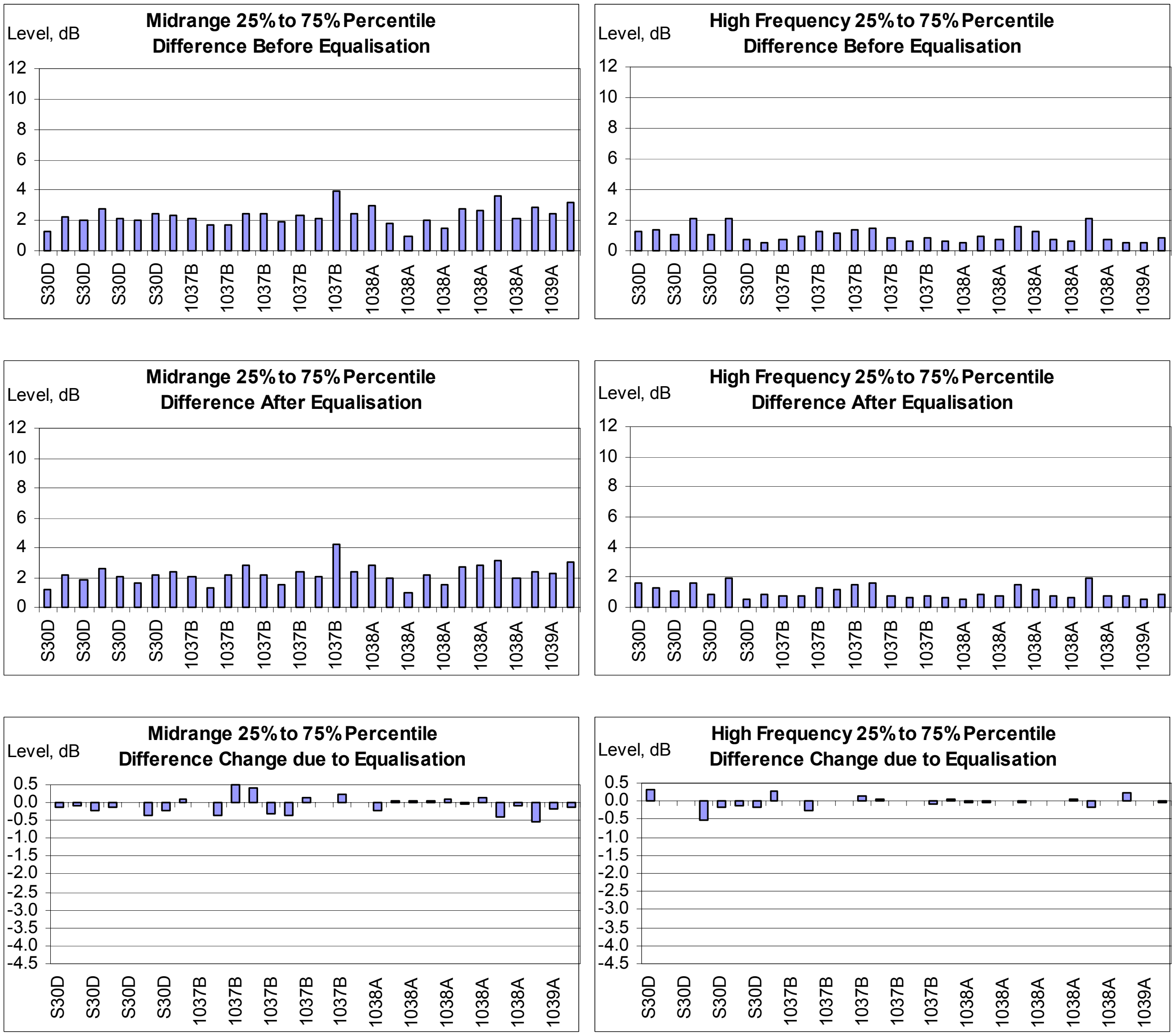

Figure 13b. 25% to 75% percentile differences for 3-way systems.

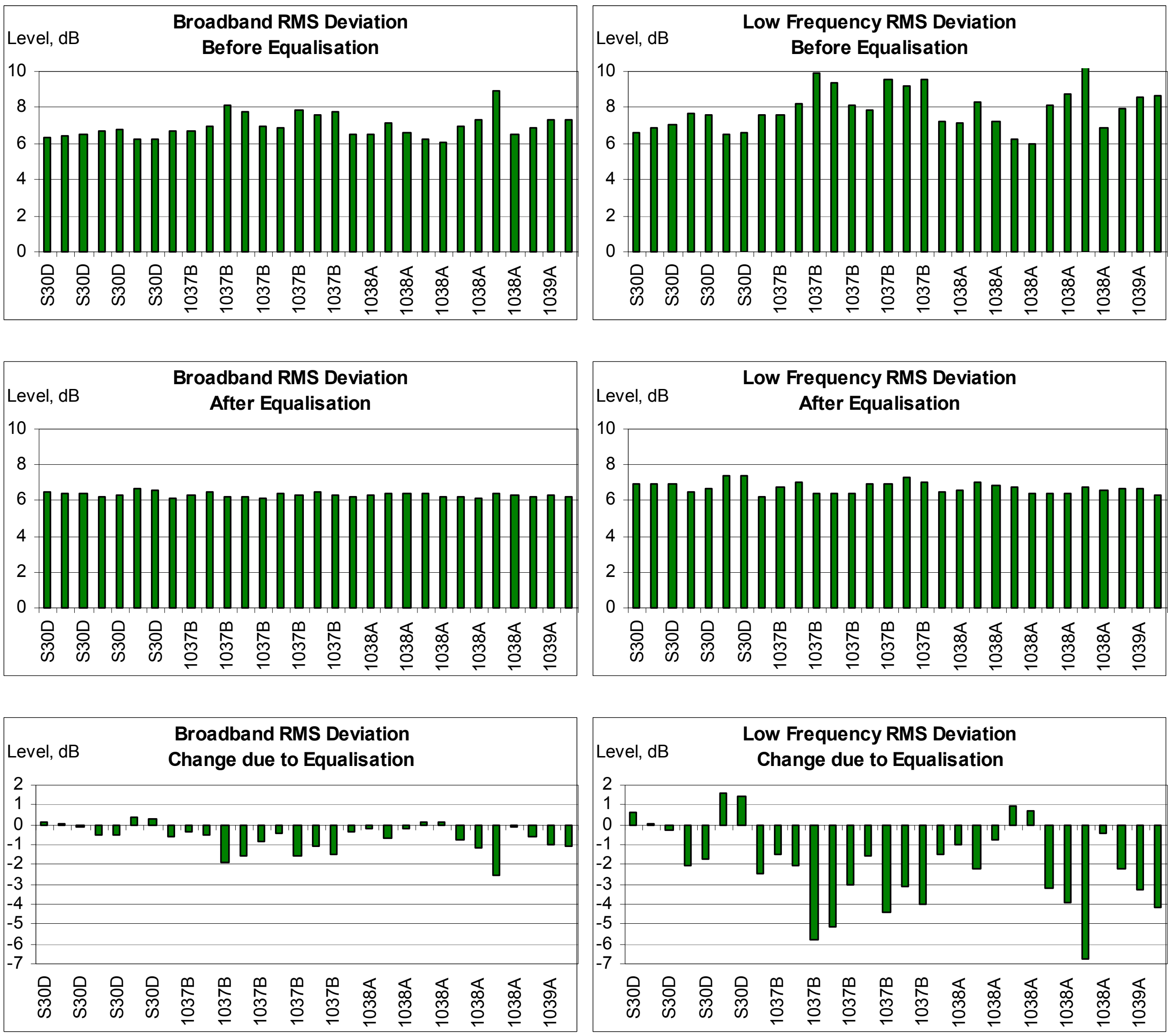

Figure 14a. RMS deviation for 3-way systems.

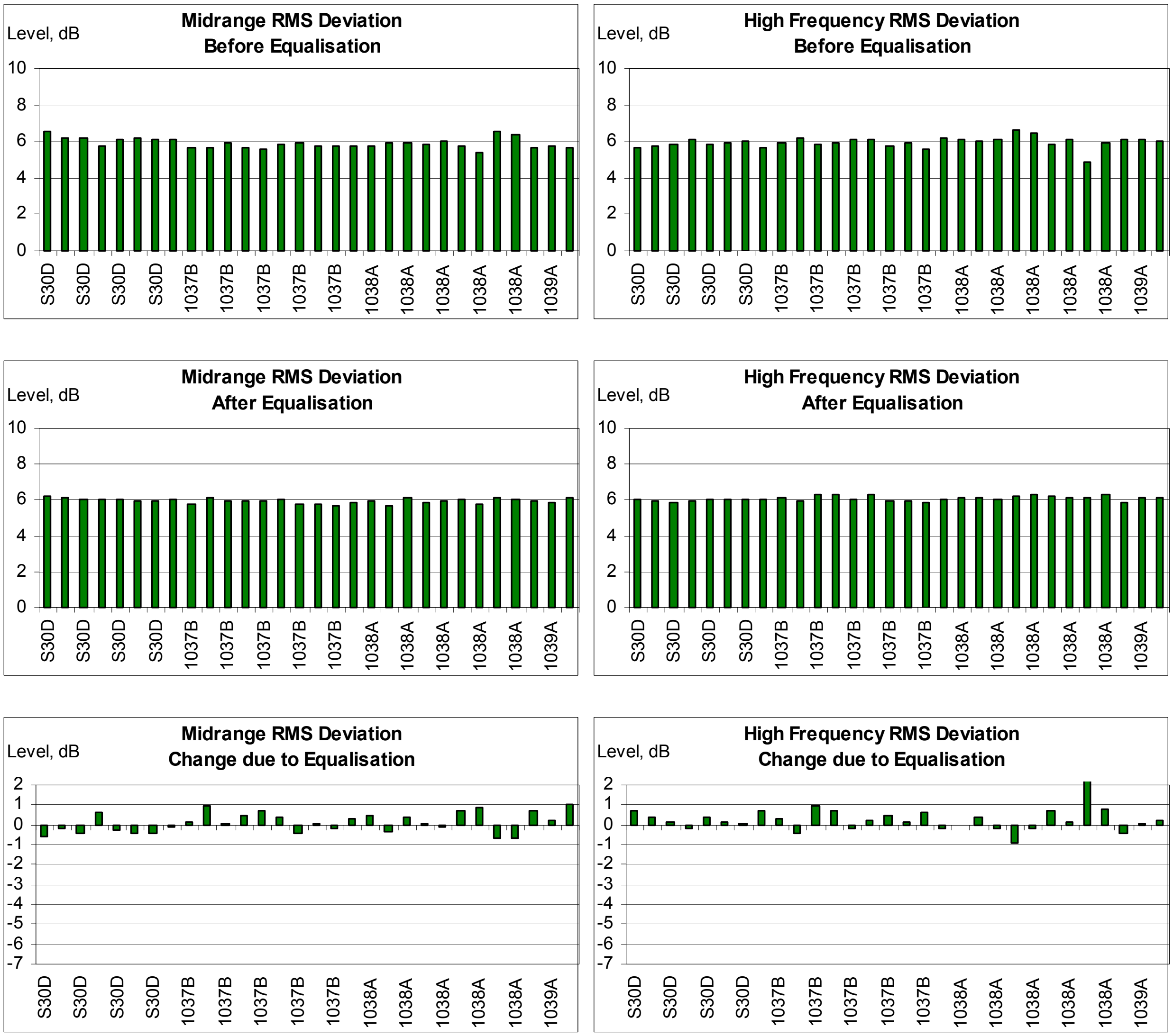

Figure 14b. RMS deviation for 3-way systems.

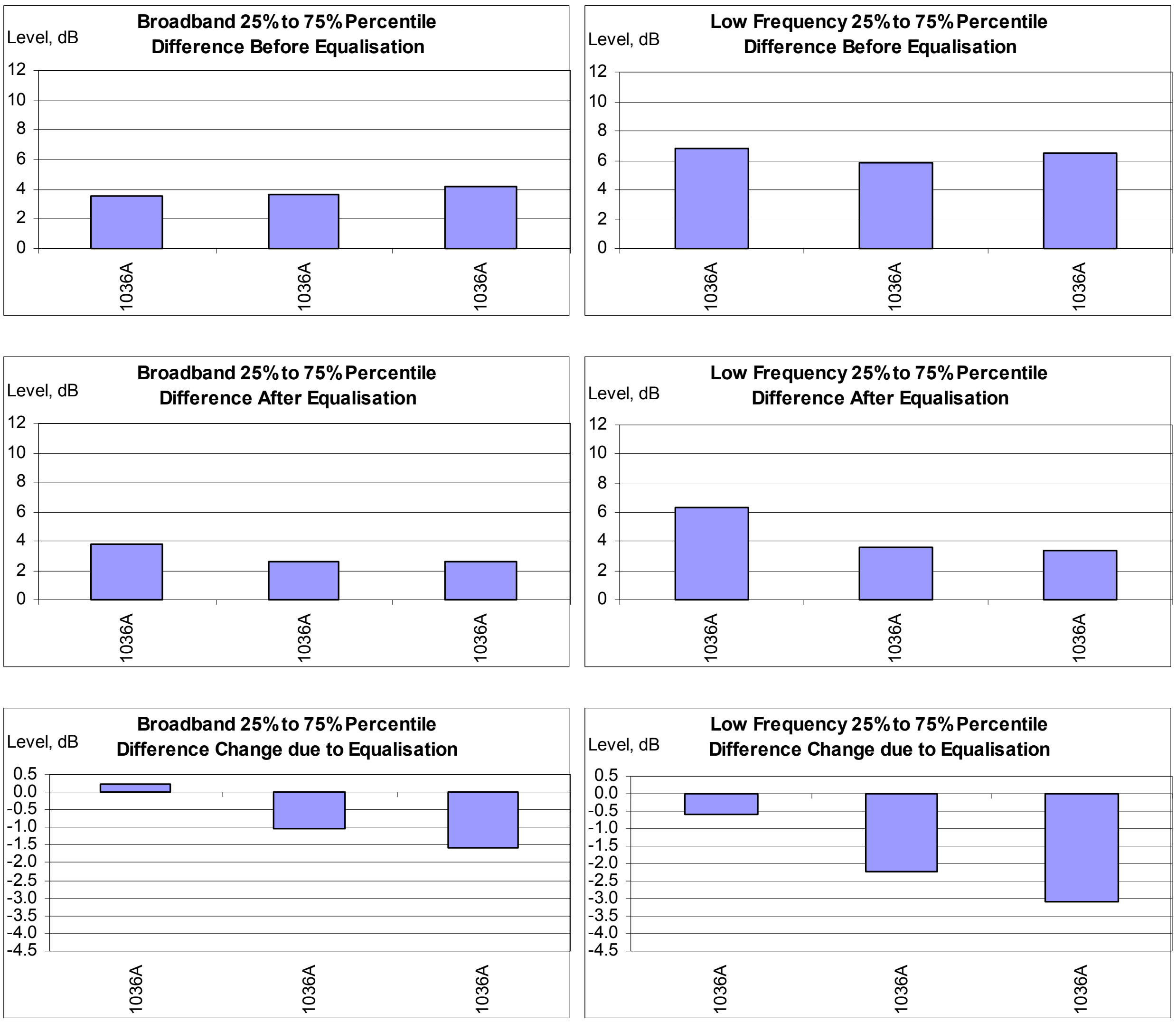

Figure 15a. 25% to 75% percentile differences for large systems.

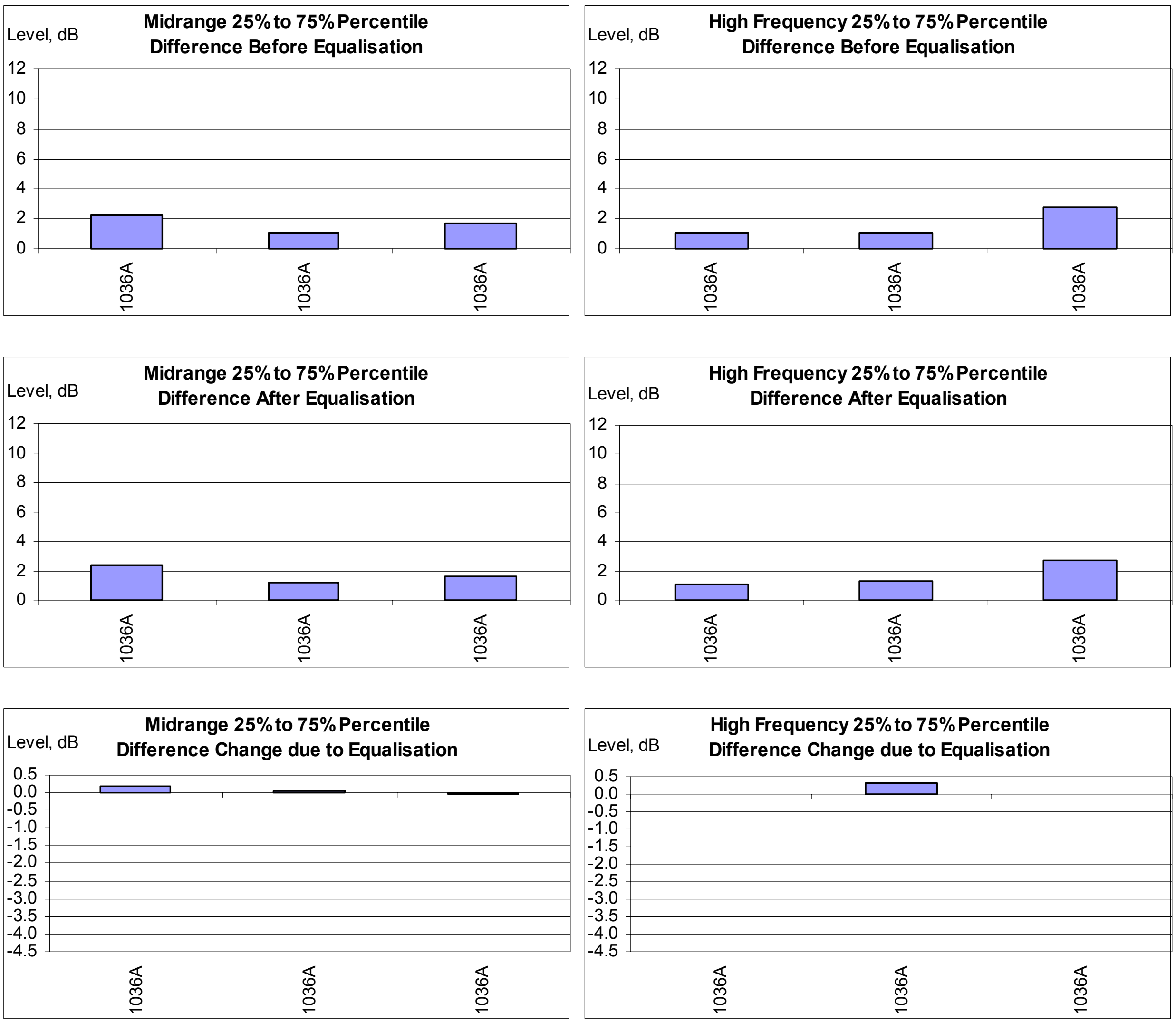

Figure 15b. 25% to 75% percentile differences for large systems.

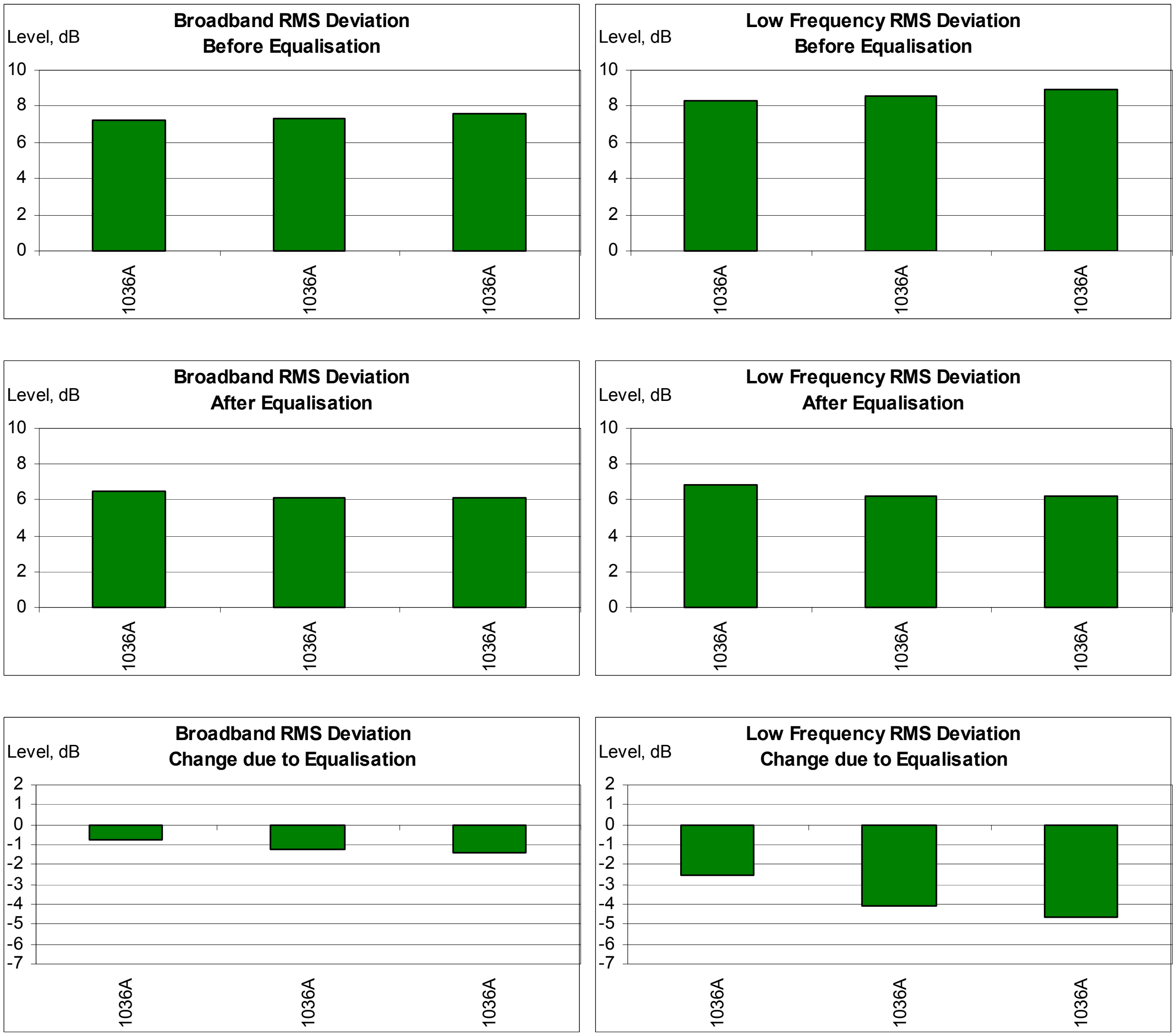

Figure 16a. RMS deviation for large systems.

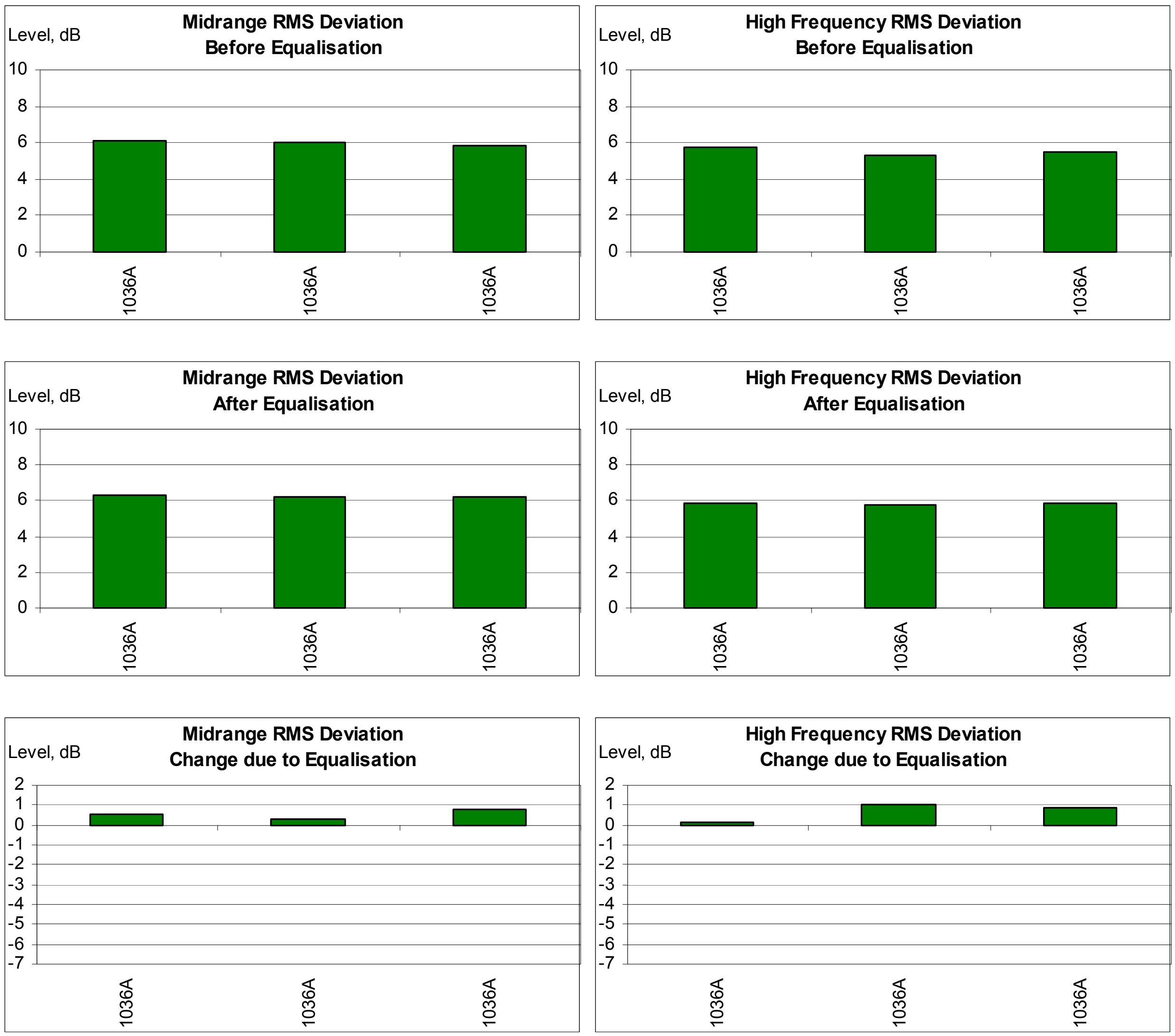

Figure 16b. RMS deviation for large systems.