FULLY AUTOMATIC LOUDSPEAKER-ROOM ADAPTATION – the RoomPerfect system

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Most systems for Loudspeaker-Room adaptation require the user to interact significantly with the system during the setup procedure. More specifically the user has to make crucial decisions like choosing a target function and/or editing the raw measurements before filter calculation. This is a result of systems relying on too little information about the 3D acoustic problem to be solved, e.g. by measuring sound pressure in only one position in the listening room. The presented system overcomes this problem by acquiring information both of local properties at the listening position and on the acoustic power in the 3D sound field. This enables a fully automatic calculation of target function and no interaction required from the user.

INTRODUCTION

Digital Signal Processing (DSP) has now matured, so that a high sound quality can be achieved at a cost, which is meaningful in loudspeaker systems. DSP has traditionally been used for crossover filters, delay alignment of drive units, equalization and more. A very powerful application of DSP for loudspeakers is room correction systems, where the impact of the listening room is reduced.

Room correction systems deal with the central problem of designing and using loudspeakers: the timbre is highly dependent on the listening room as well as on the position of both loudspeaker and listener [1, 2, 3, 4, 5, 6, 7, 8, 9, 10]. Many different approaches to reduce this problem have been reported [11, 12, 13, 14, 15, 16, 17, 18, 19, 20, 21, 22, 23, 24, 25, 26, 27, 28, 29, 30].

Many of these systems are based on a measurement of the transfer function from the input of the loudspeaker to an omnidirectional microphone, placed at the preferred listening position. An equalizing filter is then introduced in the signal path yielding a resulting transfer function, which approximates a target function.

One problem of such systems is their sensitivity to changes in the position of the sound source as well as the position of the listener. If the position of either the loudspeaker or the listener is changed, the effects can be severe coloration, pre-echoes, etc. The gain at a number of single frequencies have, by measurements, been found to change up to 20 dB caused by moving the listening position 10 cm even at frequencies below 100 Hz [31].

Sound pressure measured in closed rooms always feature notches at several frequencies due to reflection cancellations and room modes. This leads to a number of well-known disadvantages such as uncontrolled high gains in the inverse filter, unless a number of additional modifications are performed, e.g. significant smoothing of the acoustic measurement before inversion. Non-minimum phase characteristics of the measured transfer function present an additional challenge when designing the inverse filter as part of meeting the target function.

A common challenge of all the mentioned systems is the choice of a meaningful target function. It has been found that part of the impact of a listening room is natural to the human ear and should not be removed by a room correction system [32, 33]. A target function must be meaningful in an acoustic and in a subjective way rather than in a mathematical trivial way. E.g. a constant amplitude characteristic has been found to result in an unnatural timbre, which is lacking level at low frequencies.

Depending on the characteristics of the used loudspeaker, different target curves might be found to be optimal, which is part of the reason for having to require the user to select between a number of predefined target curves in most systems. However this is not necessary in the described system, because much more information of both listening room and loudspeakers is acquired by this system.

This paper presents how DSP can be used in a fully automatic way to adapt a loudspeaker to its acoustic environment in such a way that local phenomena at a specific listening position are addressed but under guidance of a general acoustic response [30]. This ensures not only an increased improvement at the listening position but also for listeners anywhere in the listening room.

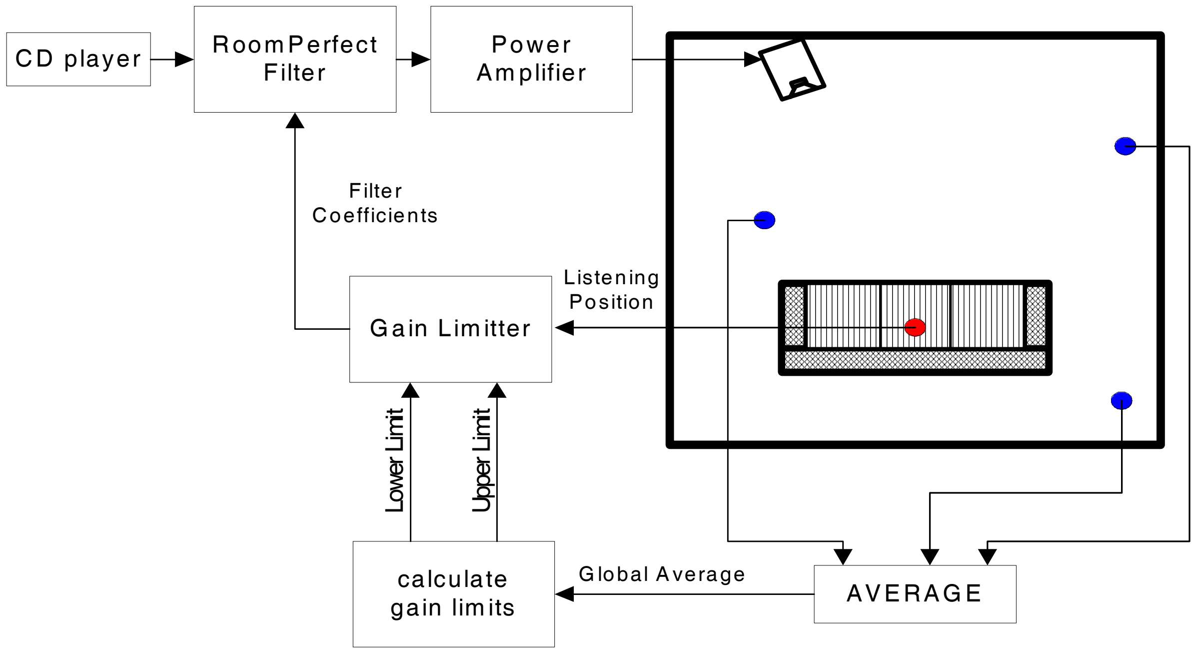

Figure 1: Basic principle of the developed room correction system, with one listening position in the sofa (light grey/red) and 3 "room positions" (black/blue) scattered randomly across the entire listening room.

1 SYSTEM PRINCIPLE

The RoomPerfect room correction system[1] is based on measuring the sound pressure in the listening position and in at least 3 randomly selected positions scattered across the entire listening room. The measurement in the listening position holds information about the listener's access to the sound field while the room positions hold information about the energy in the 3D sound field. The correction for the listening position is then bound by upper and lower gain limits calculated from the information about the 3D sound field. The basic principle of the system is shown in figure 1.
Target curves are automatically calculated based on the measurements. In fact the target curve enables the system to preserve the basic characteristics of the used loudspeaker, so that the system is not trying to make all loudspeakers sound alike; it is only removing the influence of the listening room. This is achieved through estimation of the main characteristics of the used loudspeaker: lower cut off frequency and slope, sensitivity, directivity index and upper cut off for the treble drive unit. More information about the system can be found in [30].

[1] The presented technology is known also commercially by the name "RoomPerfect™"

1.1 DSP structure

Figure 2 shows how different tasks in the system are handled by DSP. The line input signal is parsed through the "RoomPerfect Filter", which is an FIR filter. Then the signal is fed to an power amplifier and on to the loudspeaker.

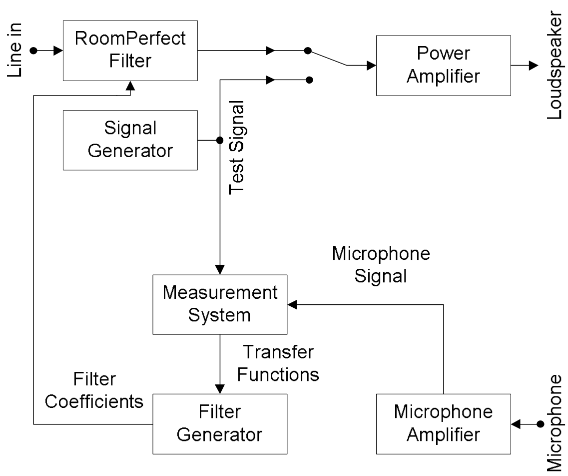

Figure 2: All major blocks in a RoomPerfect system are handled by quite different disciplines of DSP.

The signal to the power amplifier can also be taken from a signal generator during a setup procedure. The "Measurement System" then analyses the amplified

microphone signal relative to the generated signal to produce a transfer function representing the linear relationship between the measured sound pressure at the microphone and the electrical input signal to the loudspeaker.
The "Filter Generator" then calculates the actual FIR filter coefficients based on the measured transfer functions at the listening position and at least 3 random room positions. The setup procedure concludes by passing the calculated filter coefficients to the "RoomPerfect Filter".

1.2 Measurement method

The system is based on measuring sound pressure using an omnidirectional microphone first in the listening position (focus position) and then in a number of room positions.
The general acoustic response, p_{global} , is calculated as the power average of the complex valued sound pressures, p_i, measured in the room positions, see equation 1. The measured sound pressure in the listening position can also be included in this average.

$$p_{global}(f) = \sqrt{\frac{\sum_{i=1}^{N} |p_i(f)|^2}{N}} \qquad (1)$$

Equation 1 represents a spatial power average of the sound pressure in a room. Figure 3 shows the sound pressure amplitudes measured at N=9 different positions scattered across a listening room together with the power average of these 9 curves using equation 1.

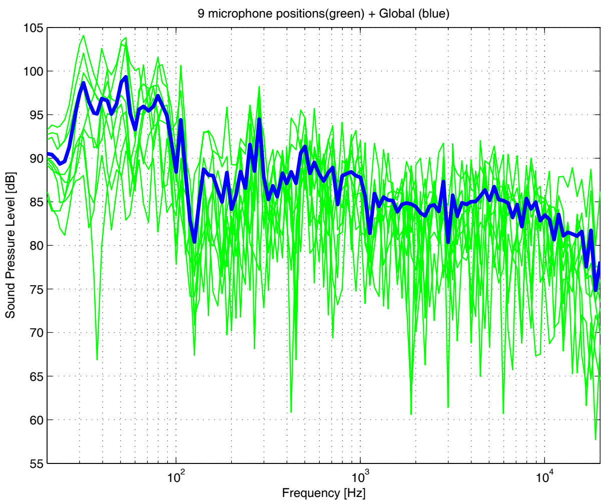

Figure 3 : Measured sound pressure amplitude [dB] in 9 different positions (gray/green) together with the power average (black/blue).

The bold curve found in figure 3 (global) and measured sound pressure at the listening position (focused) represents the fundamental input to the system: energy in the 3D sound field (global curve) and the interface to the sound field observed at the listening position.

1.3 Automatic target calculation

A critical part of room correction is to realize that the target is not the anechoic chamber, i.e. a Dirac delta impulse response. It turns out, that part of the influence of a room is positive and should not be removed, while other parts have to be removed/compensated [32, 33].
Also a fixed standard target curve for all loudspeaker types is not optimal, because DSP is 1 dimensional, which can not and should not be used to change 3D phenomena like the directivity of a loudspeaker, which is a pure function of the geometry of the loudspeaker and the loudspeaker drive units.
The developed system automatically calculates target curves based on the measurements – both originating from the listening position and the from the room positions. This means that the target curves attempt to preserve the basic characteristics of the used loudspeaker, so that the system is not trying to make all loudspeakers sound alike, it is only removing the influence of the listening room.
A central point at this stage is to realize that the power averaged sound pressure across a listening room, i.e. the global curve, is approximately the power response of the loudspeaker positioned in the actual listening room, where the one measurement at the listening position is often more of the same nature as the on axis pressure response combined with a fraction of power response.
It follows from this, that the directivity index of the used loudspeaker should be removed from the global curve before this can be used to guide the focused curve (listening position) because directivity index is the ratio between pressure response and power response.
This is obtained by calculating separate target curves for global and for focused, which takes into account the following characteristics of the used loudspeaker: lower cut off frequency and slope, sensitivity, directivity index and upper cut off for the treble.
Reproduction of sound in a room always results in an increased sound pressure level towards lower frequencies. This is partly a consequence of the lower absorption found in typical rooms at low frequencies. However this is natural to the human ear as this provides the sense of being in a room. Consequently a room correction system cannot be allowed to remove this smooth increase in level at low frequencies, also referred to as the room gain [32, 33].
This can be seen as a smooth increase of the calculated target curve at lower frequencies in figure 4, which shows the power averaged curve (global) together with

the calculated target curve for global $T_{global}(f)$. Here a lower cut off frequency of 27.6 Hz was calculated from the measurements, and the sensitivity was calculated to 87.9 dB SPL. The slope above 467 Hz was calculated to be -4.1 dB/decade.

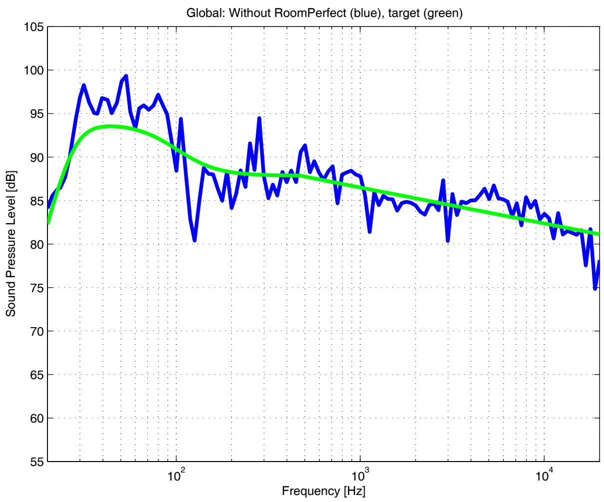

Figure 4 : Measured power averaged sound pressure amplitude (black/blue) together with the calculated target curve for global (gray/green).

Figure 13 shows a measurement at the listening position (focused) together with the calculated target curve for focused, $T_{focus}(f)$. The very same lower cut off as in figure 4 was used for the focus target curve, but here the sensitivity was 86.0 dB SPL. The directivity index is clear when comparing figure 4 and 13, where the upper roll off caused by the directivity at higher frequencies starts at 467 Hz in figure 4.

Figure 5 shows a different example of calculated global target together with the power averaged sound pressure measured in a different room using a different loudspeaker system. Here a lower cut off frequency of 36.4 Hz was calculated from the measurements, and the sensitivity was calculated to 86.4 dB SPL. The slope above 4195 Hz was calculated to be -18.4 dB/decade.

Figure 15 shows in a similar way as figure 13 a measurement at the listening position (focused) together with a calculated target curve for focused, $T_{focus}(f)$. This measurement corresponds to the global curve in figure 5, and the sensitivity was calculated to be 87.1 dB SPL.

Equation 2 and 3 gives the intermediate correction filters for focused, $H_{focus}(f)$, and for global, $H_{global}(f)$. These intermediate correction filters are the ratio between target curves, $T_{focus}(f)$ and $T_{global}(f)$, and measured sound pressures, $p_{focus}(f)$ and $p_{global}(f)$ respectively. This effectively removes the directivity index from the global curve and removes the loudspeaker characteristics leaving the influence of the listening room.

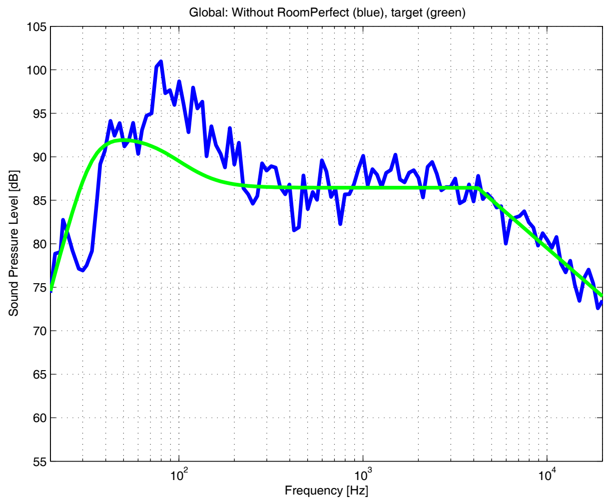

Figure 5 : Measured power averaged sound pressure amplitude (black/blue) together with the calculated target curve for global (gray/green).

$$H_{focus}(f) = \frac{T_{focus}(f)}{p_{focus}(f)} \tag{2}$$

$$H_{global}(f) = \frac{T_{global}(f)}{p_{global}(f)} \tag{3}$$

1.4 Gain limits

The lower and upper gain limits, $gain_{lower}(f)$ and $gain_{upper}(f)$, are calculated as a function of frequency from the information about the 3D sound field, i.e. the intermediate correction filter for global, $H_{global}(f)$..

Equation 4 and 5 gives one example of upper and lower gain limits, where the lower gain limit is simply the intermediate correction filter for global, $H_{global}(f)$, lowered 3 dB, see equation 4. The upper gain limit, given by equation 5, is set to 0 dB if the intermediate correction filter for global, $H_{global}(f)$, is below 0 dB indicating too high energy in the 3D sound field. In this situation no positive gain is allowed in the system.

However if the intermediate correction filter for global, $H_{global}(f)$, is above 0 dB indicating a lack of energy in the 3D sound field, then the upper gain limit is set to be equal to the intermediate correction filter for global, $H_{global}(f)$. This is the case at 126 Hz in figure 6. This means that if there is a general lack of 2 dB at a certain frequency, then up to +2 dB of gain is allowed by the system.

$$gain_{lower}(f) = \frac{H_{global}(f)}{\sqrt{2}} \tag{4}$$

$$gain_{upper}(f) = \begin{cases} 1 | \; H_{global}(f) < 1 \\ H_{global}(f) | H_{global}(f) \geq 1 \end{cases} \quad (5)$$

Figure 6 shows an example of an intermediate correction filter for focus together with upper and lower gain limits according to equation 4 and 5. Figure 7 shows the resulting curve after the gain limiting, G(f), as given in equation 6.

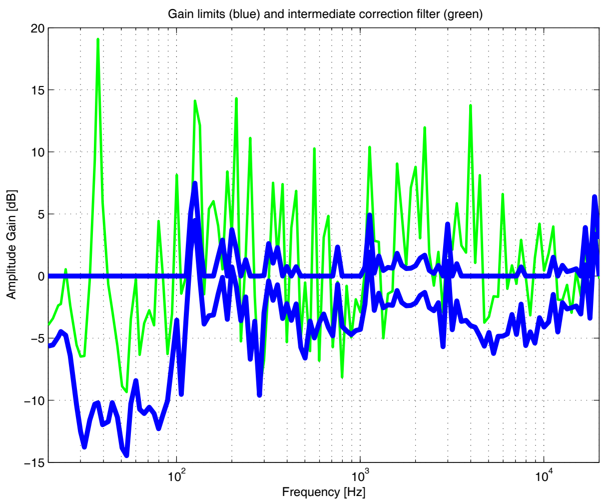

Figure 6 : Intermediate correction filter for focus (gray/green), upper and lower gain limits (black/blue).

$$G(f) = \begin{cases} gain_{lower}(f) | \; H_{focus}(f) < gain_{lower}(f) \\ H_{focus}(f) | \; gain_{lower}(f) < H_{focus}(f) < gain_{upper} \\ gain_{upper}(f) | \; gain_{upper}(f) < H_{focus}(f) \end{cases} \quad (6)$$

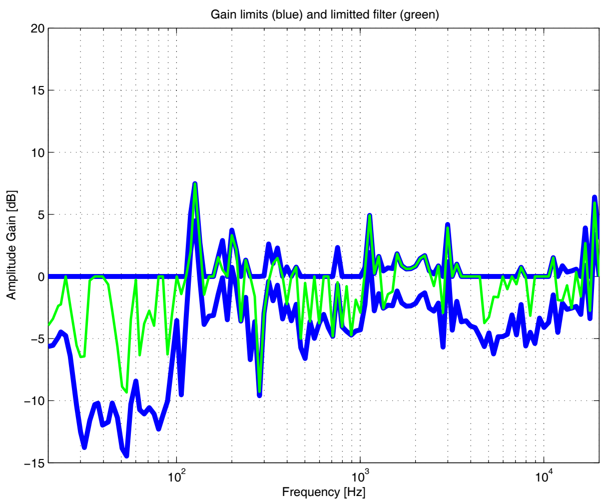

Figure 7 : Upper and lower gain limits (black/blue) and the resulting curve after gain limiting (gray/green).

It should be noted that the intermediate correction filter for focus in figure 6 corresponds to the traditional single point room adaptation with no smoothing applied.
The developed system does not rely on smoothing to try to overcome the problem of uncontrolled high gains like at 38 Hz in figure 6. When the intermediate correction filter for focus falls between the upper an lower gain limit, then the resulting curve is exactly equal to the intermediate correction filter for focus with no smearing or loss of accuracy caused by a gross smoothing, which is necessary to try to control the intermediate correction filter gain at 38 Hz: +19 dB.

1.5 Filter generator

The filter target, G(f), is convolved with a smoothing function to control the amount of local details in the filter target, which in turn determines the needed FIR filter length. It is important to stress that this smoothing does not play the same role as the smoothing applied in most other reported room correction systems, where the smoothing tries to overcome the problem of uncontrolled high gains as a result of inverting a transfer function, which includes deep notches.
The smoothed filter target, $G_{smooth}(f)$, is a real valued function of frequency, i.e. zero phase at all frequencies. Minimum phase realization is chosen in order to apply the minimum natural amount of phase shift for a given change of amplitude. To the extent that an amplitude response originates from a minimum phase system, which is the case to a large extent for a loudspeaker system, then a minimum phase realization of the inverse amplitude response is indeed the correct phase, so that both amplitude and phase are optimally corrected.
The important factors for the experienced timbre in a room are the energy in the 3D sound field and the coupling to the sound field at a specific listening position and these factors are taken into account in the amplitude response.
Here it is important to note that a sound field is 3 dimensional, which means that the phase response changes differently as a function of position when moving in different directions from a listening position. It follows from this that correcting the complete phase response only makes sense in one point, i.e. 1 position. This is the reason why only the part of the phase response, which corresponds to the amplitude response, should be corrected, i.e. the minimum phase part. This is obtained by employing Homomorphic filtering of the smoothed filter target function performed in Cepstrum domain [28].

2 RESULTS

A full range "Dali Helicon 400" loudspeaker was placed in a listening room as shown in figure 8. Then the transfer function was measured from the input of the loudspeaker to the sound pressure measured by a microphone in 9 different positions: 1 measurement in a listening position and 8 measurements in room positions scattered randomly across the entire listening room.

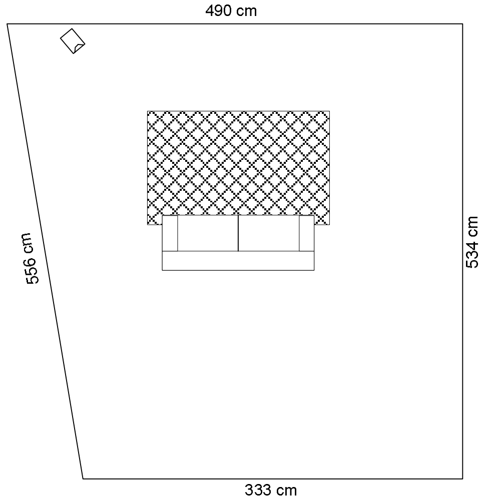

Figure 8 : Measurement setup in a listening room with a ceiling height of 260 cm. The listening position was in the center of the sofa.

An alternative loudspeaker system in a different room was also investigated through similar measurements. The dimensions of the alternative room was: (length x width x height) = 7.60m x 4.40m x 2.70m. The loudspeaker setup in the alternative room was a 2+2 system in which two corner placed woofers "Lyngdorf Audio W210" were playing up to 400 Hz and the rest of the frequency range was covered by more freely placed main speakers "Lyngdorf Audio MH-1". The main speakers were placed 1,10 m away from both side wall and end wall. An effective difference of distance from the listening position to the corner woofer and the main speaker was compensated through applying a delay of 5.2 msec. to the signal going to the main speakers.

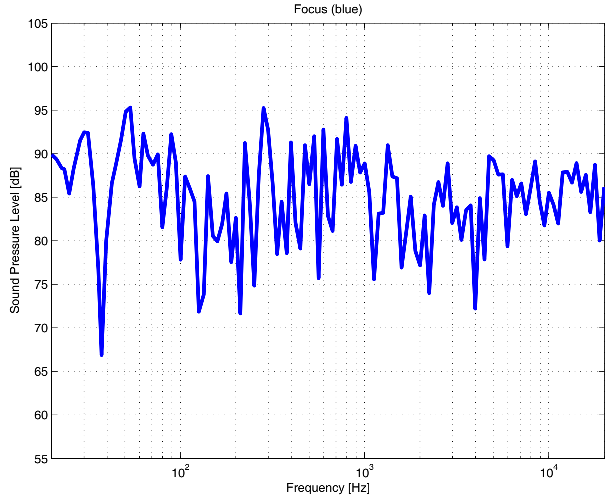

Figure 9 : Sound pressure in the listening position (focused curve).

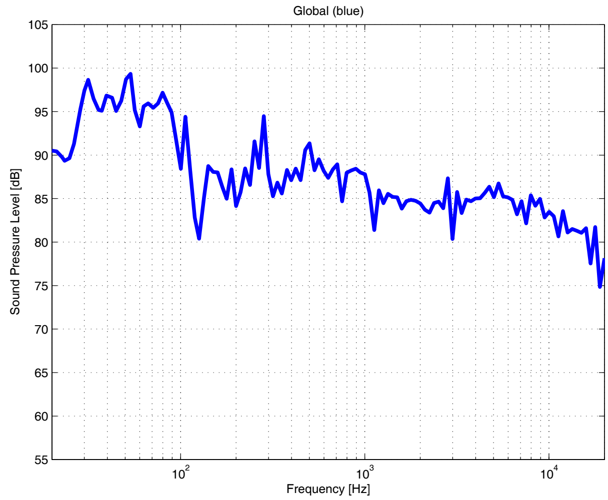

Figure 10 : Power averaged sound pressure across the 9 positions in the listening position (global curve).

2.1 Measured sound pressure

Figure 9 and 10 show the global and focused measured curves in the room shown in figure 8. The global curve is calculated from all 9 measurements using equation 1.

2.2 Filter characteristic

Figure 11 and 12 show the generated filters, which are inserted into the signal path before the power amplifier. The filter is calculated according to the procedure described in section 1.

Figure 11 shows the filter, which was generated to suit the listening room shown in figure 8. The generated filter for the alternative listening room with the 2+2 system installed is shown in figure 12.

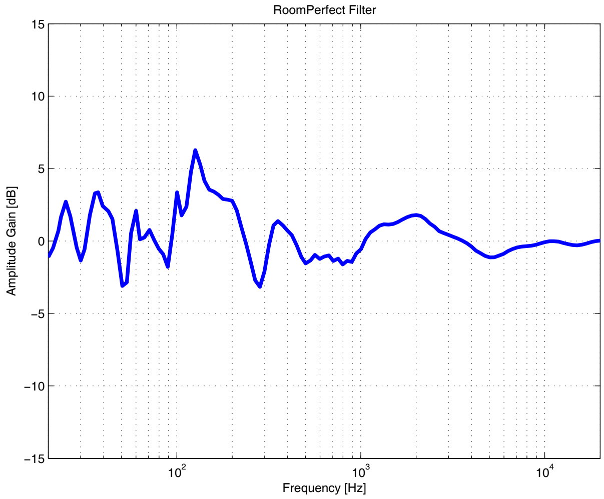

Figure 11 : Filter characteristic of the generated correction filter for the listening room shown in figure 8.

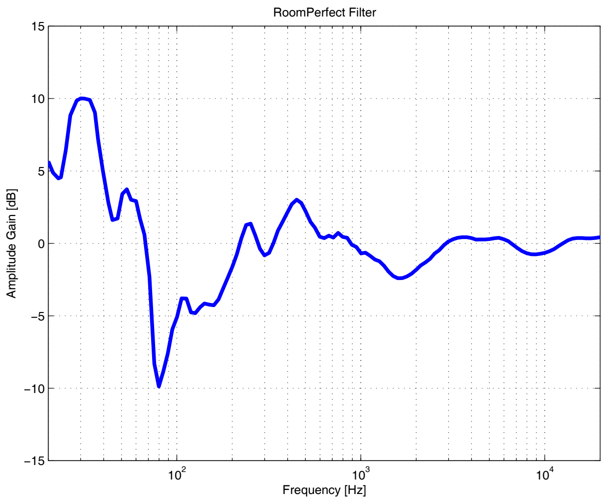

Figure 12 : Filter characteristic of the generated correction filter for the alternative listening room with a 2+2 loudspeaker setup installed.

2.3 Impact of the filter

Figures 13 and 14 show the sound pressure measured at the listening position before and after applying the filter shown in figure 11. Both figures also show the calculated target curve for focus.

Figures 15 and 16 then show in a similar way the measured sound pressure in the alternative room before and after applying the filter shown in figure 12. These figures also show the calculated target curves for focus.

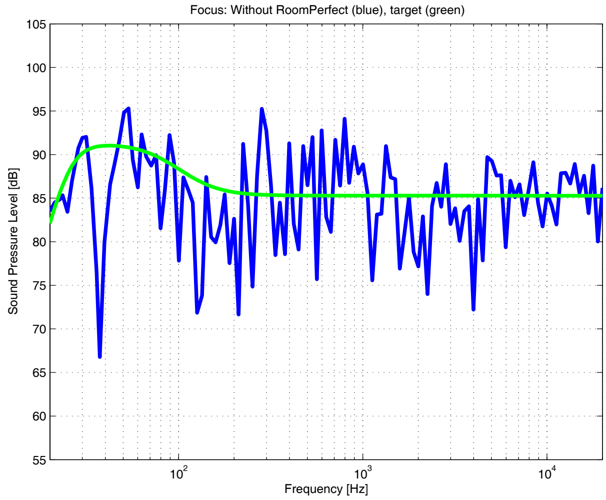

Figure 13 : Sound pressure measured at the listening position without applying the filter (black/blue). Gray/green smooth curve is the automatically calculated target curve.

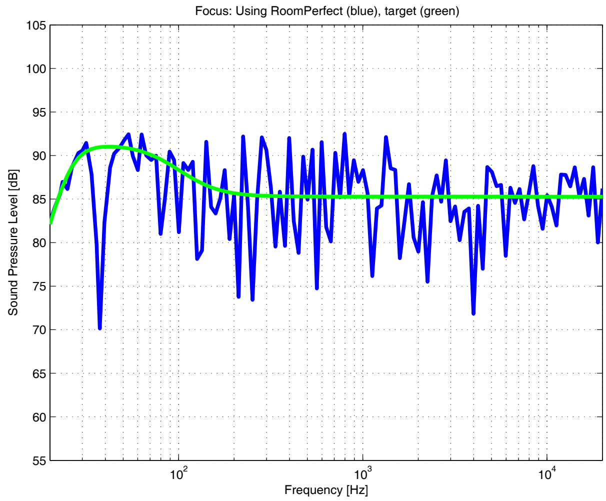

Figure 14 : Sound pressure measured at the listening position after applying the filter (black/blue). Gray/green smooth curve is the automatically calculated target curve.

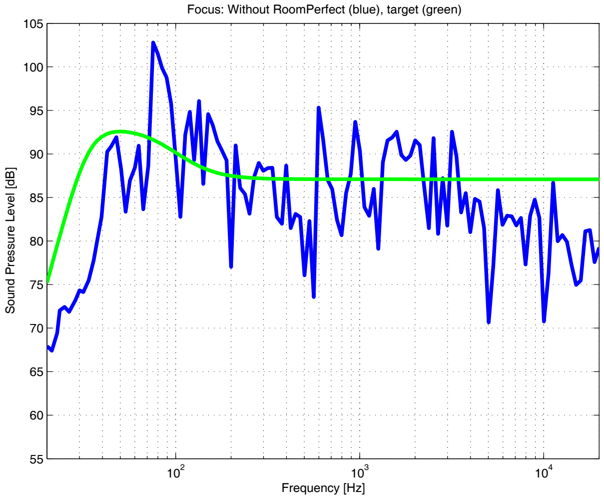

Figure 15 : Sound pressure measured at the listening position without applying the filter (black/blue). Gray/green smooth curve is the automatically calculated target curve.

Informal listening evaluations have shown that the developed system does provide a huge improvement not only at the measured listening position but also across the listening room. It was found that adding more room positions improved the perceived sound quality evaluated in the listening position. In addition more room positions also provided an improvement when moving away from the measured listening postion.
An important finding was, the lack of the artifacts normally found in single point room correction systems, like uncontrolled high gains.

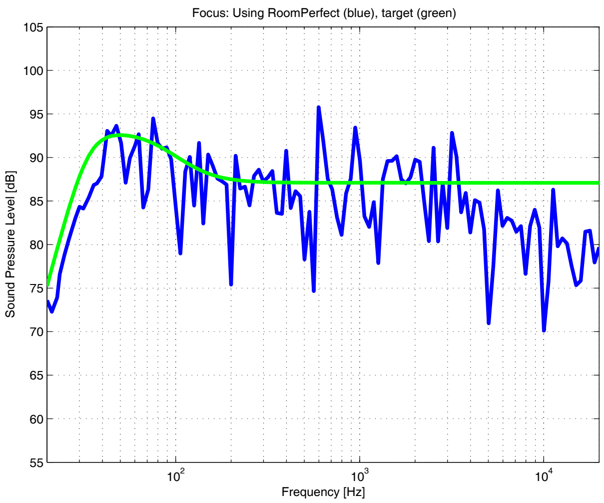

Figure 16 : Sound pressure measured at the listening position after applying the filter (black/blue). Gray/green smooth curve is the automatically calculated target curve.

Another important finding was that the developed system was able to preserve the basic character of the used loudspeaker. Listeners commented that they could still recognize the sound of the loudspeaker – only the impact of listening room was removed. This gave a much more natural sound experience compared to the traditional approach, where DSP is applied also to 3D phenomena like directivity index.

3 DISCUSSION

According to the presented method it is possible to provide a room adaptation of a loudspeaker which will provide a listener with a listening experience where severe coloration due to loudspeaker-room interaction has been significantly reduced and still without introducing coloration artifacts in locations outside the listening position.
Earlier reported single point room correction systems fail to perform an optimal correction for the influence of the listening room simply because these systems have too little information available. These systems are based on measuring the sound pressure in 1 position, which is a 1 dimensional function of frequency. This approach is a projection of a 3D object, i.e. the sound field, onto a 1 dimensional scale, i.e. the sound pressure in 1 point!
A projection from 3D onto a 1D scale can at some frequencies bee successful and meaningful; however at other frequencies the 3D object can be oriented in such a way, that the 1D projection doesn't make sense. One example could be a strong room mode, which is strongly excited due to the position of the loudspeaker. At the center frequency of the room mode the sound field is far too strong compared to other frequencies, i.e. it has far higher energy and should be attenuated.
However if the single point room correction is measured directly at the notch of this room mode, then the measured sound pressure would be very low, which in turn calls for a very high gain at this frequency, which would only make the room mode even stronger.
The developed system overcomes this problem by using the added information gained from the random room positions. In the described example the system would simply introduce an upper gain limit of 0 dB at the center frequency of the room mode, which would effectively prevent the system from adding more energy to the strong room mode even when the listening position is directly at the notch of the room mode.
On the other hand it is important to understand that the developed system is not just a linear weighted average between measurements performed at the listening position and measurements performed across the entire listening room like other reported multiple point room correction systems. If this was the case, then adding more and more room positions would degrade the performance at the measured listening position. The correction filter in the developed system is always based

on the one measurement at the listening position, but guided by the information about the complete sound field (room positions) to overcome the 3D to 1D projection artefacts.

4 CONCLUSIONS

With the developed system, it is possible to fully automatically equalize a loudspeaker to a listening position but still taking into account the general properties of the room. Even though the equalizing filter is based on a measured sound pressure at a specific listening position, the introduction of frequency dependent upper and lower gain limits based on an inverse of a transfer function representing a power averaged sound pressure in the room, it is possible to shape the equalizing filter according to the general acoustic properties of the room since these properties are inherent in the global transfer function. The developed system overcomes the artefacts normally found in single point room correction system by using the additional information about the 3D sound field contained in the measured room positions scattered randomly across the entire listening room. This effectively handles the 3D to 1D projection problems found when acquiring information about a 3D sound field by measuring 1D sound pressure in 1 position like earlier reported single point room correction systems.
The additional information is also used to automatically calculate a target function, which suits the loudspeaker actually used, which opens up the possibility of a fully automatic room correction system, where no user interaction is needed. This system also provides a more natural timbre to room correction systems by recognizing the fact that part of the influence of a room is perceived to be natural and should not be removed by a room correction system.

REFERENCES

[1] Søren Bech, "Perception of Timbre of Reproduced Sound in small rooms: The Influence of the room and the loudspeaker position", *J. Audio Eng. Soc.*, 42, 1994, 999 – 1007.

[2] Søren Bech, "Timbral aspects of reproduced sound in small rooms, I", J. Acoust. Soc. Am., 1995, 1717 – 1726.

[3] Søren Bech, "Timbral aspects of reproduced sound in small rooms, II", J. Acoust. Soc. Am., 1996, 3539 – 3549.

[4] K. O. Ballagh, "Optimum Loudspeaker Placement Near Reflecting Planes", *J. Audio Eng. Soc.*, vol. 31, number 12, 1983.

[5] Sean E. Olive *et al.*, "The Effects of Loudspeaker Placement on Listeners' Preference Ratings", presented at the AES 93rd convention, 1992 October. Preprint number 3352.

[6] Nick Zacharov *et al.*, "Multichannel Level Alignment, Part II: The Influence of Signals and Loudspeaker Placement", presented at the AES 105th convention, San Francisco, 1998, September 26–29. Preprint number 4816.

[7] R. V. Waterhouse, " Output of a Sound Source in a Reverberation Chamber and Other Reflecting Environments ", J. Acoustical Society of America, vol. 30, No. 1, pp. 4-13, January 1958.

[8] Roy Allison, "The Influence of Room Boundaries on Loudspeaker Power Output", *J. Audio Eng. Soc.*, vol. 22, number 5, 1974.

[9] Glyn Adams, "Time Dependence of Loudspeaker Power Output in small Rooms", *J. Audio Eng. Soc.*, vol. 37, number 4, 1989.

[10] Thomas Salava, "Acoustic Load and Transfer Functions in Rooms at Low Frequencies", *J. Audio Eng. Soc.*, vol. 36, number 10, 1988.

[11] Peter G. Craven and Michael A. Gerzon, "Practical Adaptive Room and Loudspeaker Equaliser for Hi-Fi Use", presented at the AES 92nd convention, Vienna, 1992, March 24–27.

[12] S. J. Elliot and P. A. Nelson, "Multiple-Point Equalization in a Room Using Adaptive Digital Filters", *J. Audio Eng. Soc.*, vol. 37, number 11, 1989.

[13] Ronald P. Genereux, "Adaptive Loudspeaker Systems: Correcting for the Acoustic Environment", presented at the AES 8th International Conference, Washington D.C., May 1990.

[14] Louis D. Fielder, "Analysis of Traditional and Reverberation-Reducing Methods of Room Equalization", *J. Audio Eng. Soc.*, vol. 51, number 1/2, 2003.

[15] Stephen T. Neely and Jont B. Allen, "Invertibility of a room response", J. Acoustical Society of America, vol. 66, pp. 165-169, July 1979.

[16] Per Rubak and Lars G. Johansen, "Design and Evaluation of Digital Filters Applied to

Loudspeaker/Room Equalization", presented at the AES 108th convention, Paris, 2000, February 19–22. Preprint number 5172.

[17] Per Rubak and Lars G. Johansen, "Listening Test Results from a new Digital Loudspeaker/Room Correction Systems", presented at the AES 110th convention, Amsterdam, 2001, May 12–15. Preprint number 5323.

[18] Glyn Adams, "Adaptive Control of Loudspeaker Frequency Response at Low Frequencies" presented at the AES 73rd convention, Eindhoven, 1983, March 15–18. Preprint number 1983.

[19] M. Miyoshi and Y. Kaneda, "Inverse Filtering of Room Acoustics", IEEE volume ASSP-36, number 2, 1988.

[20] Stan Curtis, "Room correction B&W's black Box", Hi-Fi news & record Review dec. 1991.

[21] Floyd E. Toole, "Loudspeakers and Rooms for Sound Reproduction – A Scientific Review", *J. Audio Eng. Soc.*, vol. 54, number 6, 2006.

[22] Todd Welti and A. Devantier, "Low-Frequency Optimization Using Multiple Subwoofers", *J. Audio Eng. Soc.*, vol. 54, number 5, 2006.

[23] John N. Mourjopoulos and Panagiotis D. Hatziantoniou, "Real-Time Room Equalization Based on Complex Smoothing: Robustness Results", presented at the AES 116th convention, Berlin, 2004, May 8–11. Preprint number 6070.

[24] Matti Karjalainen *et al*, "Low-Frequency Modal Equalization of Loudspeaker-Room Responses", presented at the AES 111th convention, New York, 2001, Novemver 30 – December 3. Preprint number 5480.

[25] Scott G. Norcross *et al*, "Evaluation of Inverse Filtering Techniques for Room/Speaker Equalization", presented at the AES 113th convention, Los Angeles, 2002, October 5–8. Preprint number 5662.

[26] Sofus Birkedal Nielsen and Adrian Celestinos, "Optimizing Placement and Equalization of Multiple Low Frequency Loudspeakers in Rooms", presented at the AES 119th convention, New York, 2005, October 7–10. Preprint number 6545.

[27] Rhonda Wilson, "The Loudspeaker-Room Interface-Controlling Excitation of Room Modes", presented at the AES 23rd International Conference: "Signal Processing in Audio Recording and Reproduction", Copenhagen, Denmark, 2003 May 23-25.

[28] Jan Abildgaard Pedersen, "ADAPTIVE BASS CONTROL – the ABC Room adaptation system", presented at the AES 23rd International Conference: "Signal Processing in Audio Recording and Reproduction", Copenhagen, Denmark, 2003 May 23-25, page 53-65 in "The Proceedings of the AES 23rd International Conference".

[29] Jan Abildgaard Pedersen, "Adjusting a Loudspeaker to its Acoustic Environment – the ABC system", presented at the AES 115th Convention, 2003 October 10 - 13, New York, Convention Paper number 5880.

[30] Jan Abildgaard Pedersen, "Loudspeaker-Room Adaptation for a specific Listening Position using Information about the Complete Sound Field", presented at the AES 121st Convention, 2006 October 5 - 8, San Francisco, Convention Paper number 6908.

[31] Jan Abildgaard Pedersen *et al.*, "The Distribution of the Low Frequency Sound Field and its Relation to Room Equalization", presented at the AES 96th convention, Amsterdam, 1994, February 26 - March 01. Preprint number 3852.

[32] Jan Abildgaard Pedersen, "Natural Timbre in Room Correction Systems", presented at the AES 122nd Convention, 2007 May 5 - 8, Vienna, Convention Paper.

[33] Fares EL-AZM and Jan Abildgaard Pedersen, "Natural Timbre in Room Correction Systems (Part II)", presented at the AES 32nd International Conference: "DSP for Loudspeakers", Copenhagen, Denmark, 2007 September 21-23.

[34] IEC rec. 268-13, "Sound system equipment, Part 13: Listening tests on loudspeakers", International Electrotechnical Commission, Geneva, Switzerland (1985).