Quantification of Subwoofer Requirements,
Part I: Generation of Stimuli and Listening System Equalization

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Quantification of subwoofer requirements, part I: Generation of stimuli and listening system equalization

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Abstract

Lower cut-off frequency & slope, amplitude ripple and group delay ripple have been examined. Shape and magnitude of both amplitude and group delay ripple were calculated from room simulations. Listening tests are to be performed using both loudspeakers in an anechoic chamber and headphones, which were equalized in an attempt to obtain equal results.

0 Introduction

The aim of this work is to study the psychoacoustically determined performance requirements for a subwoofer loudspeaker. The work is part of EUREKA Project 1653, which is referred to as "MEDUSA" (Multichannel Enhancement of Domestic User Stereo Applications). The "MEDUSA" project is a 3.5 years joint research project with the following partners: British Broadcasting Corporation, Institute of Sound Recording at University of Surrey, Nokia Research Centre, Genelec Oy and Bang & Olufsen A/S.

This is part I in a series of papers, and this paper describes how DSP is used for generation of stimuli and equalization of the listening system. Part II in this series of papers: "The influence of lower system cut-off frequency and slope, and pass-band amplitude and group delay ripple" describes the actual listening experiments, i.e. experimental strategy, procedure and results [1].

The increasing use of digital surround sound systems, like 5.1 systems, leads to an increased attention of low frequency reproduction. Considering 5.1 systems, especially the ".1" channel and modern programme material, loudspeaker manufacturers has a desire to quantifying the requirements of subwoofers. It has been assumed that Digital Signal Processing (DSP) is employed for all filtering and control tasks in the subwoofers considered. This was decided to avoid limitations on the applicability of the conclusions.

Essential requirements of subwoofers are the lower cut-off frequency and slope, because these are linked to maximum displacement, system tuning and power requirements. In this context it is chosen to focus on two additional parameters: amplitude ripple and group delay ripple, both in the pass band. Amplitude ripple sets the requirement to how much the amplitude response is allowed to deviate from the ideal response, which could be a flat response. Deviations could arise from variations in the parameters of the loudspeaker drive units, influence of the room etc. Requirements regarding maximum group delay ripple imposes a limit on how rapid the phase is allowed to change.

The experiments are divided into 2 separate groups of experiments: lower cut-off frequency and slope in the first group and amplitude and group delay ripples in the second group. All the parameters in each group must be able to be varied independently, i.e. orthogonal variation of the parameters.

1 Generation of Stimuli

Stimuli for the tests are produced by different filtering of 7 selected programme pieces. First the lower cut-off frequency and slope are changed and then both amplitude and group delay ripples in the pass band are added.

1.1 Programme material

Seven pieces of programme material is selected from a wide set of considered material. Two pieces are used during training of subjects, while 4 pieces are used in the actual experiment. The last piece is used for calibration purposes only. The spectral distribution, especially below 200 Hz, are of special interest when selecting the 7 pieces of programme material. Figure 1 includes a table, which specifies the selected 7 pieces.

Total loudness at the listening position is used to calibrate the playback level of the experiments. The calculation of total loudness requires a stationary signal, which is chosen to be “Pink Noise”, as seen in figure 1. The reproduction level of “Pink Noise” is adjusted from the loudness measurements, and the levels of the other 6 pieces are individually adjusted relatively to “Pink Noise” based on calculations of the total power contents of the 7 pieces. Equation 1 gives the relative gain of programme piece “A” relative to “Pink Noise” and the total power content is given for each of the 7 pieces in figure 1. Since the power levels given in figure 1 are already expressed in dB, the relative gain of any of the programme pieces can be found simply by subtracting the power of the actual programme piece from the power of “Pink Noise”.

$$relative\_gain\_of\_A = \sqrt{\frac{\Sigma\,|Pink\_Noise(n)|^2}{\Sigma\,|A(n)|^2}} = \sqrt{\frac{Power(Pink\_Noise)}{Power(A)}} \quad (1)$$

Power spectra of all the programme pieces has been calculated using the full 1 minute of each programme piece. Figure 2 gives the power spectrum of the 2 training pieces, i.e. the first 2 pieces in figure 1 and figure 3 gives the power spectrum of the 4 pieces that is used in the actual experiment, i.e. the next 4 pieces in figure 1. The original programme pieces are turned into ".WAV" files (Microsoft Wave Files): 2 channels, 16 bits and 44100 Hz sampling frequency. All processing on the programme pieces are performed in this format, i.e. the programme pieces stays ".WAV" files right until conversion to analog signals.

1.2 Variable lower cut-off frequency and slope

The lower cut-off frequency is varied from 20 Hz to 50 Hz by prefiltering the programme material using a digital high-pass filter with variable cut-off frequency. The range from 20 Hz and 50 Hz are considered to be a typical range of lower cut-off frequencies for a subwoofer. An intermediate value of 35 Hz is chosen as a third cuf-off frequency. Additional information about the experimental plan is found in part II in this series of papers [1]. Figure 4 illustrates the variation of lower cut-off frequency.

Not only the cut-off frequency is of importance when considering the lower roll off of a subwoofer, also the order, i.e. slope, is important. A closed box system is of 2nd order and a vented box is of 4th order. Considering these very commonly used systems, it is chosen to change the order of the high-pass filter between 2nd, 4th and 6th order. Figure 4 shows this variation of the slope from 2nd order up to 6th order.

1.3 Amplitude and group delay ripple

Ripples, both in amplitude response and in the group delay, are found in practical usage of subwoofers. Ripples can arise for a number of reasons, e.g. tolerances of components, limited possibilities of equalization, poor tuning of the system, diffraction, reflection and room modes. Knowledge about how large ripples are allowed to be would be useful, when designing and installing a subwoofer system. Ripples of different magnitude can be added to the programme material before playback by introducing a digital filter comprising a non-flat amplitude response or a non-linear phase response. Non-flat amplitude response in the pass band of the subwoofer yields amplitude ripple and non-linear phase response yields a non-constant group delay, i.e. group delay ripple.

Shape and magnitude of both amplitude and group delay ripple are calculated from room simulations of an IEC 268-13 sized room. The room simulations are based on the principle of mirror images and carried out using a developed program: "ConSim". Documentation of the development and usage of "ConSim" was published in an earlier paper [2]. "ConSim" calculates the first 1000 msec. of the impulse response in a specified receiver position, given a specified source position. Both source and receiver are assumed to be omni directional. The impulse response is sampled at a rate of 4 kHz.

The simulated room is a rectangular room with dimensions 5.03m x 6.02m x 2.50m (width x length x height)=(X,Y,Z). The source position was chosen to be (X,Y,Z)=(0.30m, 0.40m, 0.35m), which is considered to be a realistic corner position, that will excite all room modes, which in turn will yield large ripples. The receiver position was chosen to be at a typical listening position near the centre of the room, but offset about 30 cm to avoid abnormal effects due to perfect symmetry: (X,Y,Z)=(2.20m, 2.70m, 1.10m).

All the above parameters of the room simulation are fixed in all simulations, but the reverberation time (T_{60}) is varied from anechoic environment to 0.8 sec. to control the magnitude of the ripples, i.e. a hard room yields large ripples compared to a well damped room. The reverberation time of the simulated rooms are estimated by time reversed integration of the squared impulse response. Equation 2 can be used to calculate a reverberation plot, *rp[n]* based on the impulse response *h[n],* which is of length *N*. Further information about time reversed integration can be found in [3].

$$rp[n] = \Sigma_{i=n}^{N} h[i]^2 \tag{2}$$

The reverberation time, T_{60}, can then be found by inspection of the reverberation plot, by determining the time, where the level is decreased 60 dB relative to the initial level. Note that the length of the simulated impulse response should be longer than the true reverberation time. If that is not the case, then the reverberation plot will decrease below 60 dB simply because there are no further samples in the impulse response.

Four levels of ripples are calculated by simulating 4 rooms with different reverberation time (T_{60}): anechoic, 0.2 sec, 0.4 sec and 0.8 sec. The reverberation time is adjusted by changing the reflection coefficients of the walls, floor and ceiling. Figure 5 includes a table of the reflection coefficient for the 6 surfaces in each of the 4 rooms. Amplitude and phase response of each room are normalised according to the anechoic room, i.e. the anechoic case yields zero amplitude ripple and zero group delay ripple. The group delay is calculated from the unwrapped phase response by differentiation. Figure 6 and 7 shows all 4 levels of amplitude ripple and group delay ripple. Please note, that ripples are only implemented from these target curves up to a certain upper frequency, i.e. to emulate the upper frequency limit of a subwoofer. Two upper frequency limits are implemented to investigate the influence of the range in which ripples are implemented in: 80 Hz and 120 Hz.

1.4 Implementation

Variation of the lower cut-off frequency and slope are implemented using digital high-pass filters with variable cut-off frequency of 2nd, 4th and 6th order according to the experimental plan [1].

The amplitude and delay deviating filters, or ripple filters, were designed based on the impulse responses produced by simulated room response calculations, found in section 1.3. Only deviations in the bass range are included, so deviations in amplitude and delay are only included up to a certain frequency, and no deviations exist above this frequency.

The experimental design [1] has three variables for the ripples: the limiting frequency below which the deviations exist (two values), the amount of amplitude deviation (four values), and the amount delay deviation (four values). The amounts of deviation were implemented by varying the reverberation time of the room simulation.

Two filter designs for each value of the reverberation time were then produced – a linear phase filter containing the the amplitude deviations but no delay deviations, and an all-pass filter containing just the delay deviations but no amplitude deviations. The delay filter was made causal by adding a constant delay.

The experimental plan [1] uses the amplitude deviation data from one reverberation time simulation with the delay deviation data from any of the four reverberation times. This was implemented by convolving an amplitude filter with a delay filter, producing one filter for each of the amplitude/delay deviation combinations. A filter has a sampling rate of 4kHz. It is 2.04s in length (8996 samples) when it contains both amplitude and phase deviating properties.

The audio sample files (.WAV files) containing the desired deviations are produced using a multi-rate convolution process. The audio data is crossover filtered at one of the desired two frequencies, the lowpass section is convolved with the deviation filters described above, and the final signal is produced by summing the two signals with necessary gain and delay alignment. For delay ripple filtering, the delay alignment can not be exact over the whole crossover frequency band, and this will produce small changes in the amplitude and delay properties at the crossover point. However, these changes are small, and do not effect the results. The convolution is done in IEEE floating point arithmetic, and the source and destination files are 16-bit fixed point, sampling frequency 44.1kHz. The lowpass section was processed at sampling frequency 4410Hz (decimation by 10). Distortion and noise levels were studied at each step of the process and proper dithering was applied at quantizations to maintain linearity througout the processing chain and to minimize noise.

2 Experimental Setups

The experiments are to be carried out using two different setups: loudspeakers in an anechoic chamber and using headphones in a “Quiet room”. One aim is to investigate if headphones could substitute the usage of an expensive large anechoic chamber when quantifying the requirements of subwoofers.

2.1 Anechoic Chamber

The processed audio samples were played back using a pair of “Reference Loudspeakers” especially build for this purpose. These “Reference Loudspeakers” extends down to 20 Hz (+/- 2.5 dB) and keeps the harmonic distortion below 2% (90 dB SPL at 1 m) in the range from 20 Hz to 50 Hz, while harmonic distortion is below 0.5% (90 dB SPL at 1 m) in the range from 50 Hz to 250 Hz. Group delay is kept below 10 ms from 50 Hz to 20 kHz, and below 20 ms from 20 Hz to 20 kHz. It is an active three way closed box system. Crossover frequencies are 160 Hz and 1.6 kHz (4th order). The separate power amplifiers for the three bands are 400 W, 160 W and 120 W.

The setup comprises two “Reference Loudspeakers” placed in a 3 m isosceles triangle in a large anechoic chamber, which can bee seen on the plan of the setup in figure 8. The subjects are to be seated in the chair. Both loudspeakers and the chair are placed on grids that are held in place by poles extending from the bottom of the anechoic chamber. The anechoic chamber is located at the Technical University of Denmark, Department of Acoustic Technology. The free space in the anechoic chamber is about 1000 m^3. The three dimensions are of similar magnitude, but not equal, and the lower limiting frequency is about 50 Hz. A detailed description can be found in [4].

A computer based system for automated run and control of the listening tests are utilized. The system is known as "GuineaPig", and the system was presented at AES-16 [5]. Additional information about "GuineaPig" can be found in [6]. "GuineaPig" was run at a Silicon Graphics workstation: Octane. This workstation includes 8 channels of 24 bit (ADAT) input and output. Graphical presentation to the subjects and feedback from the subject will take place using a Laptop PC connected to the workstation.

Control of analog level and conversion from digital to analog are performed in a Digital Mixing Console: YAMAHA O3D. This mixer comprises 20 bits linear 8 times oversampling D/A converters. Figure 9 shows an overview of the signal path starting at the original programme pieces located on the original Compact Disks (CD).

2.2 Headphones

As an alternative to using an anechoic chamber, headphones are utilized in an attempt to obtain equal results. A number of different headphones were tested and an electrostatic type was chosen: SENNHEISER HE 60, Electrostatic. The amplifier for this headphone is SENNHEISER HEV 70.

Figure 10 includes measurements of the distortion of the headphone at a max. 100 dB SPL level. The headphones were placed on a Brüel & Kjær "HATS" (Head and Torso Simulator: B&K 4128), and measurements were taken using a Brüel & Kjær Audio Analyzer: B&K 2012. The "HATS" was mounted with "Hard Ears" as an option from Brüel & Kjær. Measurements took place in the anechoic chamber at Bang & Olufsen. The levels of both 2nd and 3th harmonic are generally more than 75 dB (0.02%) below the fundamental, except for a number of frequencies, where the level is 60 dB (0.1%) below the fundamental. The levels of both 2nd and 3th harmonics are raising to about 50 dB (0.3%) below the fundamental at 20 Hz.

During listening experiments, subjects will be seated in a "Quiet room" with the headphones mounted. The "Quiet room" is located in the basement of the department for Research & Development at Bang & Olufsen. The background noise level has been found to be max. 25 dB(A) SPL ("Slow" averaging).

In an attempt to obtain equal results, it was decided to include a non-linear simulation of the "Reference Loudspeaker", such that distortion due to non-linearities of the "Reference Loudspeaker" is added to the signals before playback using the headphones. Section 3.2 describes how the program "LoDist" is used to perform this simulation. In order to make the sound pressure at the ear drums as similar as possible in the two cases: listening to real loudspeakers in the anechoic chamber and listening via headphones, advanced equalization (EQ) has to be performed on the signals before playback via headphones. Section 3.3 describes this equalization in detail. Figure 11 shows the signal path in use when headphones are used in the "Quiet room".

3 Listening System Equalization

The setup in the anechoic chamber is measured and calibration is performed using "Pink Noise" and a Loudness meter. In order to compare the two setups, both the non-linearities of the loudspeaker and the difference in linear transfer function of the headphones compared to listening in the anechoic chamber are taken into account.

3.1 Anechoic Chamber

The middle curve in figure 12 shows a measurement of the left "Reference Loudspeaker" placed in the anechoic chamber according to figure 8. The measurement was taken using a free field microphone (B&K 4133), measurement amplifier (B&K 2690: NEXUS) and an Audio Analyzer (B&K 2012). The free field microphone was placed at the listening position pointing directly towards the left reference loudspeaker. The first mode of the anechoic chamber clearly shows in figure 12 at approximately 35 Hz, which is a frequency so low that the damping is insufficient to be anechoic. This mode in not desirable, but it was decided to leave the system like this and take the response in figure 12 as part of the circumstances for the experiment.

During the listening experiment 3 different playback levels, which are spaced by 10 dB between each level, are to be use. The highest level is found by applying full scale "Pink Noise", and adjusting the level to 2 dB below "clipping" indication on the rear of the loudspeaker. Adjusted to this level, a sound pressure level of 68.8 dB (LIN) was measured using the free field microphone at the listening position, when "Pink Noise" was applied to the left "Reference Loudspeaker". In an IEC 268-13 standard listening room this adjustment corresponds to approximately 80 dB (LIN) sound pressure level.

Still using the free field microphone, the loudness level was measured to 22.15 Sones via a loudness meter: B&K 2144. This loudness meter is based on ISO 532B: Zwicker Loudness. When a 1 kHz, full scale on a CD, was applied using this adjustment, 78.4 dB (LIN) was measured via the free field microphone. The two lower playback levels are simply 10 dB and 20 dB below this adjusted level.

3.2 Nonlinear simulation of subwoofer

A non-linear simulation of the "Reference Loudspeaker" are utilized when playback is to be performed using headphones. This ensures that distortion due to non-linearities of the "Reference Loudspeaker" is added to the signals before playback using the headphones. This operation is performed using the non-linear loudspeaker driver simulator program "LoDist".

"LoDist" operates on the basis of position, i.e. displacement, dependant parameters: Bl(x), Cms(x) etc. All relevant parameters are determined as a function of absolute displacement from the neutral position. Measurements were taken on a woofer driver similar to the ones used in the "Reference Loudspeakers". Based on these parameters, "LoDist" solves numerically the differential equations that describes the loudspeaker driver and it environment, e.g. cabinet. More information about measuring position dependant parameters, operation and principle of "LoDist" can be found in [7].

When running the simulation, care was taken to ensure, that the electrical voltage applied to the simulated driver, was similar to the voltage applied to the physical driver in the "Reference Loudspeaker". This was achieved by applying appropriate scaling, when relating the digital signal fed to "LoDist" and the corresponding analog voltage.

3.3 Headphones

Listening tests are to be performed using both loudspeakers in an anechoic chamber, and headphones. In order to make the sound pressure at the ear drums as similar as possible in the two cases, advanced equalization (EQ) was performed on the signals before playback via headphones. Head Related Transfer Functions (HRTF) were measured in the anechoic chamber using the setup, which was described in section 2.1. Head Related Transfer Functions are described in [8].

The "HRTF's" were measured using the left "Reference Loudspeaker" and a "HATS" (Head and Torso Simulator: B&K 4128) placed in the listening position in the anechoic chamber facing a point just half way between the speakers. Figure 12 includes two HRTF measurements at the top of the figure: Left Speaker to Left Ear (the upper most curve) and Left Speaker to Right Ear. The shadow effect of the head is obvious above 400 Hz, where Left Speaker to Left Ear is much higher than Left Speaker to Right Ear. Below 150 Hz both HRTF's coincides with the middle curve, i.e. the free field microphone measurement. Perfect symmetry of the acoustic setup is assumed when forming Right Speaker to Right Ear and Right Speaker to Left Ear, but differences in sensitivities of the two ears of the "HATS" are incorporated.

The selected headphones were placed on the "HATS", and the transfer functions to each ear were measured. These amplitude responses are shown at the bottom of figure 12. As described in section 3.2, "LoDist" simulates the non-linear behavior of a loudspeaker driver, but to do so accurately it has to have the right linear response as well, i.e. the linear response of the "Reference Loudspeaker" is also part of the "LoDist" simulations. But the linear response of the "Reference Loudspeakers" is also included in the measured "HRTF": transfer function from input to the speakers to output of the "HATS". For this reason, an inverse filter of the linear transfer function of "LoDist" should be included in the equalization (EQ), in order to cancel one of the two linear transfer functions of the "Reference Loudspeakers" found in the signal chain.

A ".WAV" file containing a digital Dirac Impulse was created and fed through "LoDist" at a level, that didn't give raise to displacements of an order, where non-linearities comes into play, e.g. a small fraction of 1 mm. Figure 13 shows the determined amplitude response along with a bold curve, which has limited attenuation towards lower frequencies. This limitation below 15 Hz is used to avoid dynamic/numerical problems when inverting the transfer function.

A digital filter (EQ), to be inserted in the signal chain before the headphones, see figure 11, was designed from the measured "HRTF's", transfer function of headphones on "HATS" and the linear transfer function of "LoDist". When considering the selected programme pieces, the equalizer (EQ) should take the signals which are meant to go to the "Reference Loudspeakers", and prepare them to be fed to the headphones in such a way that the sound pressure at the ear drums of the "HATS" are equal to the situation where the "HATS" is placed in the anechoic chamber in front of the "Reference Loudspeakers".

In the situation of calibration using "Pink Noise", the aim of the equalizer (EQ) is somewhat different. In this situation, the equalizer (EQ) should take the "Pink Noise", which is meant for playback via loudspeakers, and measurement via a free field microphone and prepare it for playback via headphones mounted on the "HATS", in such a way that the output of the "HATS" is similar to the output of an free field microphone sitting in the anechoic chamber in front of the loudspeakers. The reason for this desire is that the loudness meter operates on input from a free field microphone when calculating the loudness level. Figure 14 summarizes all these considerations regarding calculation of the EQ filter in the two cases: programme pieces (using "HRTF" measurements) and "Pink Noise" for calibration (using the free field microphone measurement).

When listening to two speakers in the anechoic chamber, each of the two ears can hear sound coming from both speakers, i.e. cross talk from both speakers to both ears. Figure 15 shows a schematic of how this is achieved using 4 individual filters for each equalizer (EQ) in the situation: programme pieces, i.e. the equalizer (EQ) for the programme pieces consists of the four filters: H_{RR}, H_{RL}, H_{LR} and H_{LL}. H_{RR} is calculated from the ratio of the "HRTF" from Right Speaker to Right Ear and the product of the transfer function of the Right Headphone and the linear transfer function of "LoDist". H_{RL} is calculated using the HRTF from Right Speaker to Left Ear, Left Headphone and the inverse transfer function of "LoDist". H_{LR} and H_{LL} are calculated accordingly. The equalizer (EQ) for the "Pink Noise" doesn't have cross talk because the free field microphone is used. Due to the assumption of perfect acoustic symmetry, the numerator in figure 14 is exactly the same considering both right and left speaker. The only difference is between the two headphones, i.e. in the denominator in figure 14. This leaves only two filters for this equalizer (EQ).

Figure 16 shows all 4 filters, that forms the equalizer (EQ) for the programme pieces, H_{RR} and H_{LL} are the two upper curves between 400 Hz and 8000 Hz. Figure 17 shows the two filters forming the equalizer (EQ) for the "Pink Noise", which is used for calibration. All measurements are taken using 1/24 octave logarithmic frequency resolution, and the filter design uses a FFT size of 65536. This calls for interpolation, which is performed in the complex plane after removing the pure time delays of all the relevant transfer functions. After interpolation, all pure time delays are reintroduced to ensure the correct time response. The final impulse response of the equalizing filters are truncated after 10000 samples. Filtering is then performed directly on the ".WAV" files using a convolution program developed for this purpose, based on floating point arithmetic.

The fixed point nature of ".WAV" files, i.e. 16 bits, is taken into account by down scaling the equalizing filters appropriately and reintroducing the removed gains when converting to analog signals. The level calibration using "Pink Noise" and the appropriate equalizing filters lead to an adjustment of the gain in the Digital Mixing Console, so that the resulting Loudness level was 21.8 Sones and the sound pressure level was 69.0 dB (LIN) for one channel playing.

4 Conclusion

Seven programme pieces have been selected for use as stimuli in listening experiments: 2 for training sessions, 4 for the actual listening experiment and 1 for level calibration of the setups. "Pink Noise" was used to calibrate the loudness level of the two setups: "Reference Loudspeakers" in anechoic chamber and headphones in a "Quiet room". Three different reproduction levels were used, each 10 dB apart. The loudest level was calibrated to 22 Sones (69 dB SPL) for one channel playing in the anechoic chamber.

Three different lover cut-off frequency was implemented: 20 Hz, 35 Hz and 50 Hz, combined with 3 different orders/slopes: 2nd, 4th and 6th order. Based on room simulations 4 different levels of amplitude ripples and group delay ripples in the pass band were implemented. Amplitude ripples vary between 0 dB and +/- 20 dB, and group delay vary between 0 ms and +/- 200 ms. These variations are found by varying the reverberation time, T_{60}, from anechoic to 0.8 sec. in room simulations of an IEC 268-13 sized room.

The setup in the anechoic chamber comprizes 2 "Reference Loudspeakers" and a listening position arranged in a 3 m isosceles triangle. The anechoic chamber has 1000 m^3 of free space. Listening tests are to carried out using an automated system: "GuineaPig" installed on a Silicon Graphics Octane Workstation, which comprizes 8 channels of 24 bit input and output. The second setup is placed in a "quiet room" and comprizes a pair of electrostatic headphones.

In an attempt to obtain equal results in the 2 setups, advanced equalization of the signals and simulation of the non-linear behavior of the woofer drivers were performed before feeding the signals to the headphones. The equalization was based on measurements of Head Related Transfer Functions in the anechoic chamber, transfer function of the "Reference Loudspeakers" using a free field microphone, transfer function of headphones mounted on a Head and Torso Simulator and the linear transfer function of "LoDist", which is the simulation program for the non-linear simulation of loudspeaker drivers. All signal processing was performed directly on Microsoft Wave files (.WAV).

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6 References

[1] Søren Bech: "Quantification of subwoofer requirements, part II: The influence of lower system cut-off frequency and slope, and pass-band amplitude and group delay ripple", presented at the 108th AES Convention, Paris 2000 February 19-22.

[2] Jan Abildgaard Pedersen, Kjeld Hermansen and Per Rubak: "The Distribution of the Low Frequency Sound Field and its Relation to Room Equalization", presented at the 96th AES Convention, Amsterdam 1994 February 26 - March 1, preprint #3852.

[3] F. Jacobsen & J.H. Rindel: "Time Reversed Decay Measurements", Journal of Sound and Vibration, Vol. 117, 1987, page 187-190 (L).

[4] F. Ingerslev, O.J. Pedersen, P.K. Møller & J. Kristensen: "New Rooms for Acoustic Measurements at the Danish Technical University", ACUSTICA, Vol. 19, 1967/68, page 185-199.

[5] Nick Zacharov, Jyri Huopaniemi and Matti Hämäläinen: "Round Robin Subjective Evaluation of Virtual Home Theater Sound Systems at the AES 16th International Conference", presented at the AES 16th International Conference: "Spatial Sound Reproduction", Rovaniemi 1999 April 10-12, page 544-556 in "The Proceedings of the AES 16th International Conference"

[6] Jussi Hynninen and Nick Zacharov: "GuineaPig - A Generic Subjective Test System for Multichannel Audio", presented at the 106th AES Convention, Munich 1999 May 8-11, preprint #4871.

[7] E. Sandermann Olsen: "Nonlinear Modeling of Low-Frequency Loudspeakers-A Practical Implementation", presented at the 102th AES Convention, Munich 1997 March 22-25, preprint #4469.

[8] Kiyofumi Inanaga, Yuji Yamada and Hiroshi Koizumi: "Headphone System with Out-of-Head Localization Applying Dynamic HRTF (Head-Related Transfer Function)", presented at the 98th AES Convention, Paris 1995 February 25-28, preprint #4011.

Title	Main Artist	Record Company	Number	Track	Time	Power [dB]
Rush	Eric Clapton	Reprise	9 26794-2	2	0:00 1:00	-18.8
Fourplay	Fourplay	Warner Bros.	7599-26656-2	10	0:25 1:25	-12.9
Ray of Light	Madonna	Maverik/Warner Bros.	9362-46847-2	5	0:40 1:40	-8.9
Bladerunner	Vangelis	East West	4509-96574-2	1	1:00 2:00	-16.2
The Hunter	Jennifer Warnes	BMG	261974	9	3:40 4:40	-13.9
Amused To Death	Roger Waters	Columbia	468761 2	3	0:05 1:05	-25.3
Pink Noise	-	-	-	-	0:00 1:00	-9.7

Figure 1: Specification of the 7 selected pieces of programme material. The first 2 are used during training and following 4 are used during the actual experiment. The last piece is 1 min. of "Pink Noise".

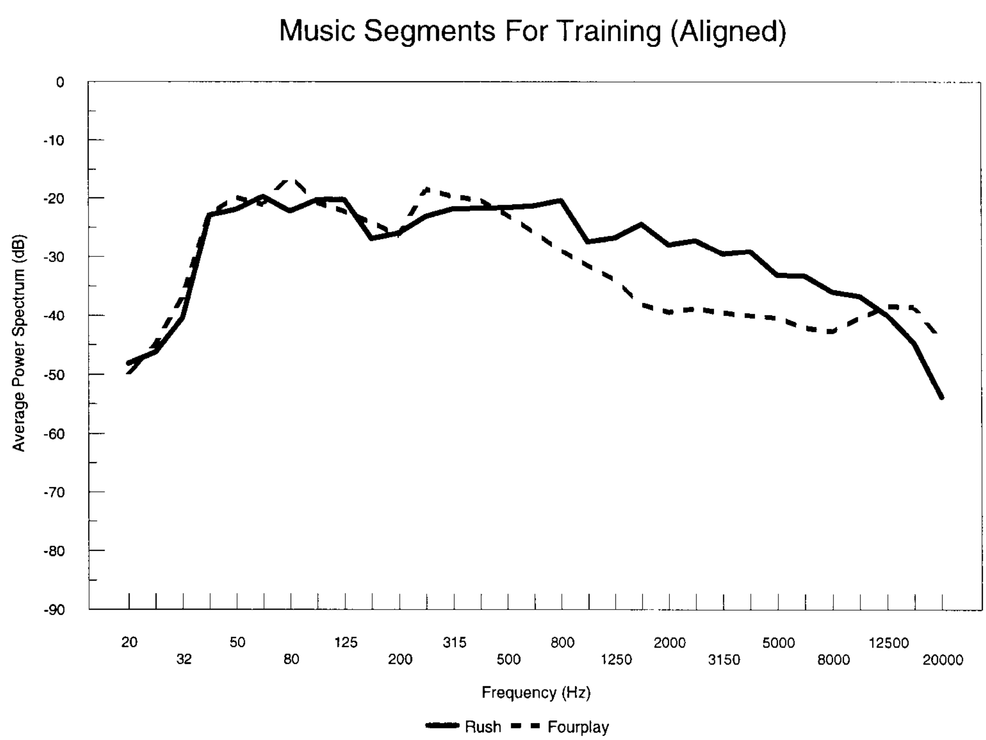

Figure 2: Power spectrum of the 2 first programme pieces (for training).

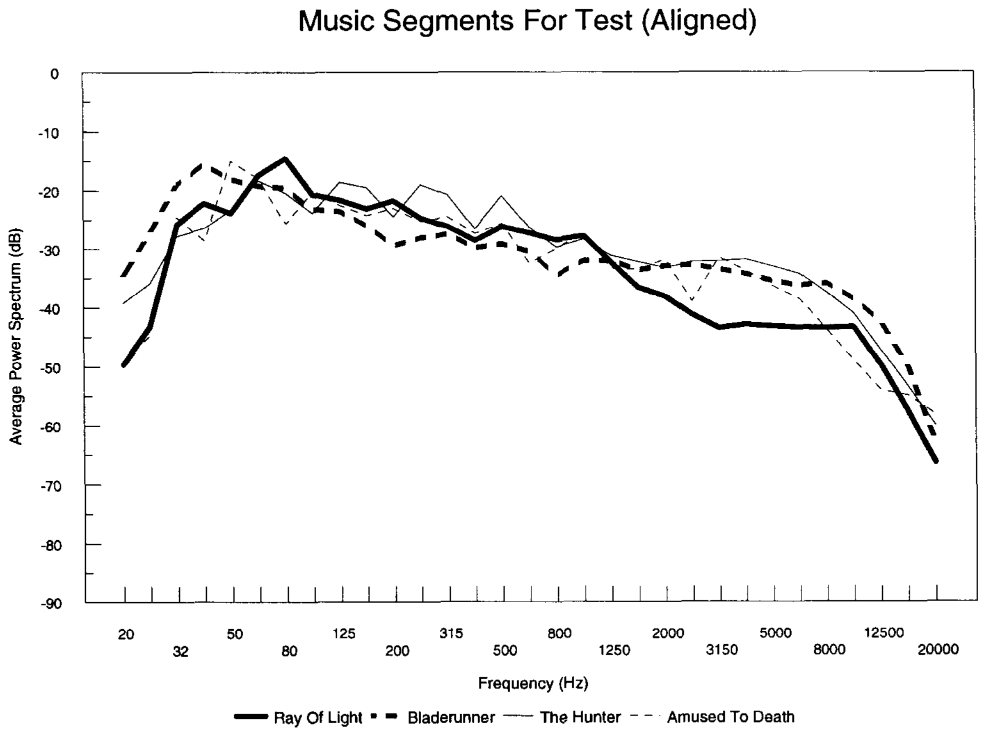

Figure 3: Power spectrum of the 4 programme pieces, that are used during the actual experiment.

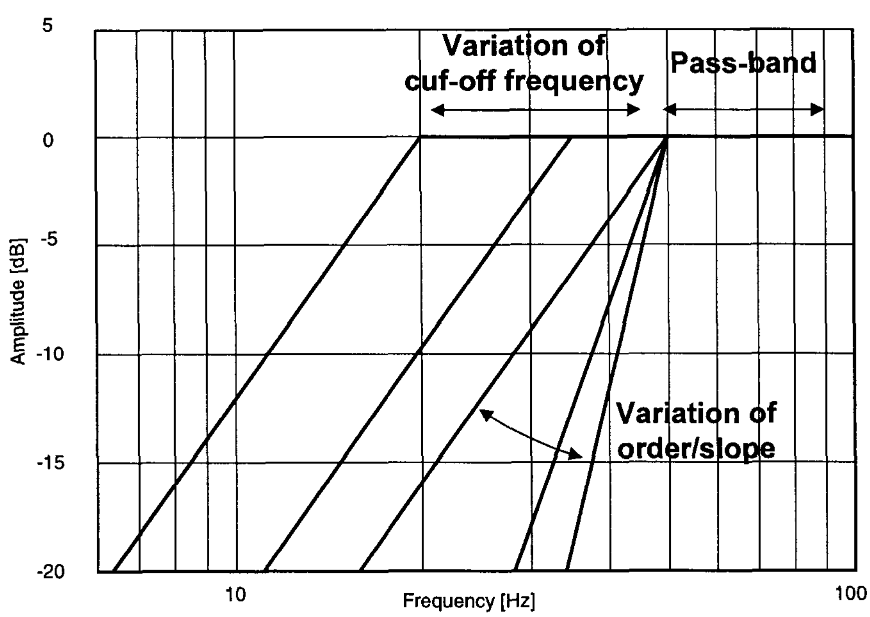

Figure 4: Variation of cut-off frequency and slope.

T_{60}	Walls	Floor	Ceiling
Anechoic	0.000	0.000	0.000
0.2 sec.	0.650	0.440	0.290
0.4 sec.	0.805	0.540	0.360
0.8 sec.	0.895	0.600	0.400

Figure 5: Reflection coefficient of walls, floor and ceiling in the 4 cases of reverberation time, T_{60}.

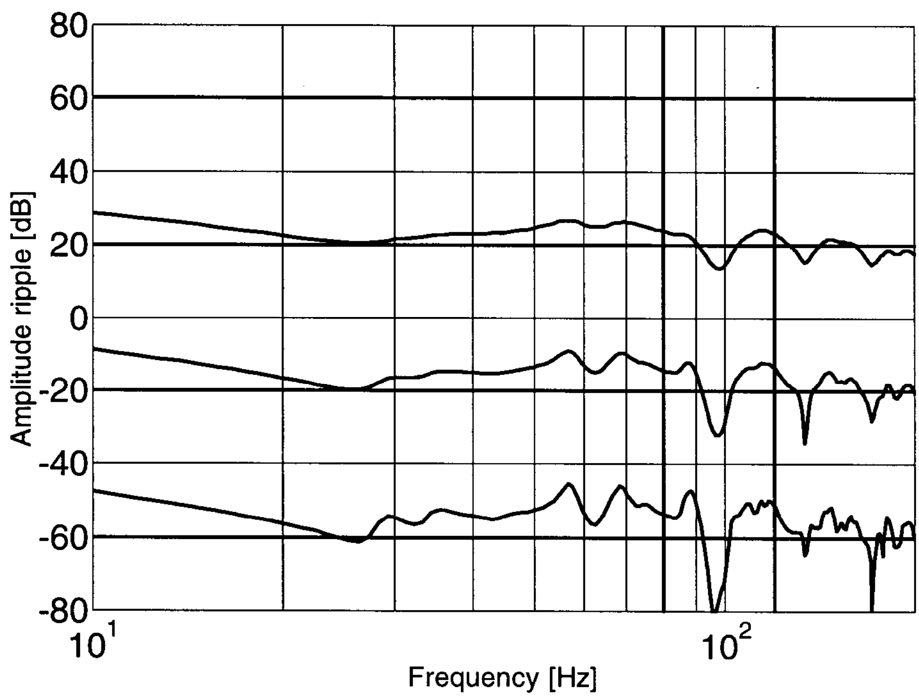

Figure 6: Amplitude ripple in 4 cases. From the top: Anechoic, T60=0.2 sec, T60=0.4 sec., T60=0.8 sec. Both 80 Hz and 120 Hz are marked as upper limits for implementing the ripples. The 0 dB level has been shifted, for a better view, to -60 dB, -20 dB, 20 dB and 60 dB respectively as indicated by bold horizontal lines.

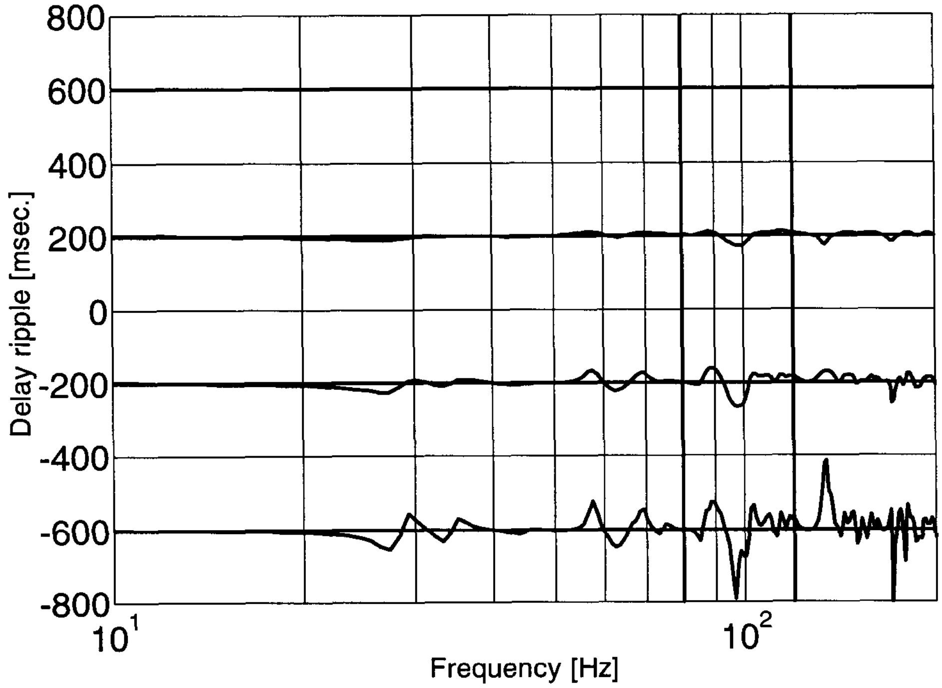

Figure 7: Delay ripple in 4 cases. From above: Anechoic, T60=0.2 sec, T60=0.4 sec., T60=0.8 sec. Both 80 Hz and 120 Hz are marked as upper limits for implementing the ripples. The 0 msec. level has been shifted, for a better view, to -600 msec., -200 msec., 200 msec. and 600 msec. respectively as indicated by bold horizontal lines.

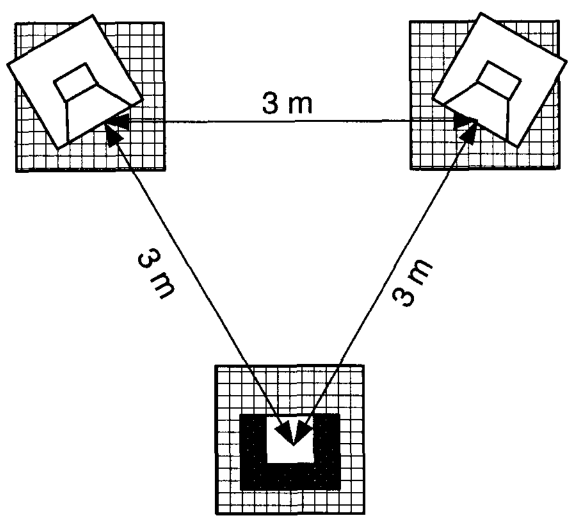

Figure 8: Setup in the anechoic chamber, comprizing 2 "Reference Loudspeakers" and a chair for listening.

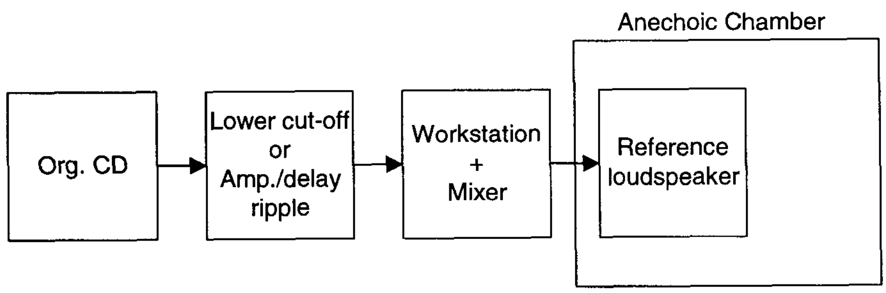

Figure 9: Signal path from original CD to playback in the anechoic chamber using the "Reference Loudspeakers".

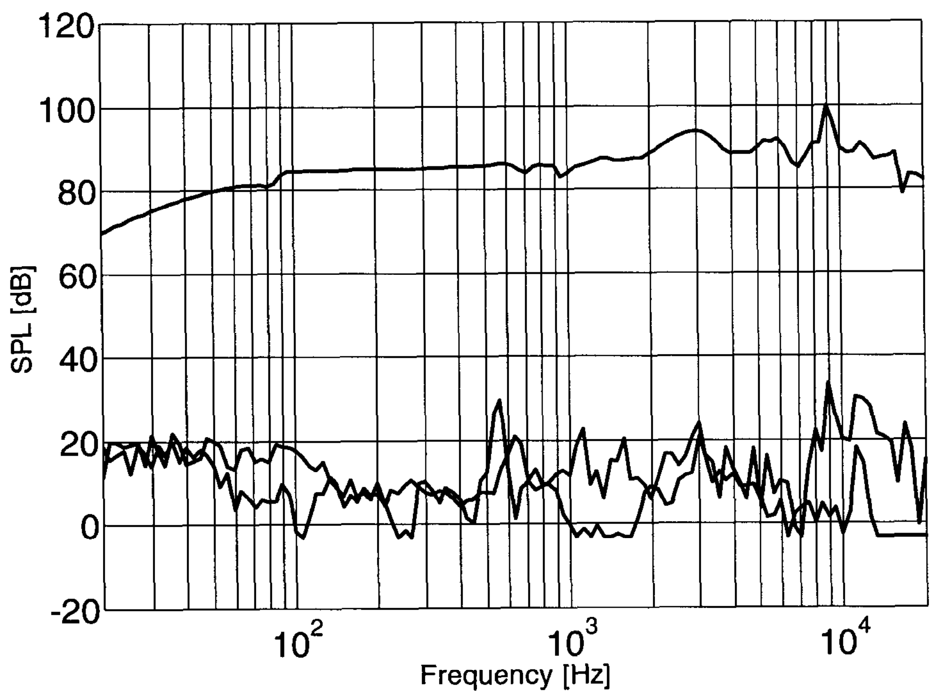

Figure 10: Fundamental, 2. And 3. Harmonic measured using HATS and SENNHEISER HE 60. Maximum SPL at 9 kHz was 100 dB.

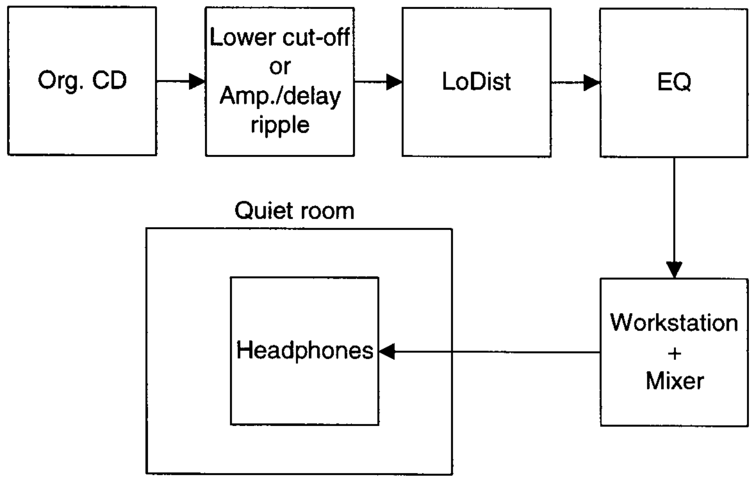

Figure 11: Signal path from original CD to playback in the quiet room using headphones.

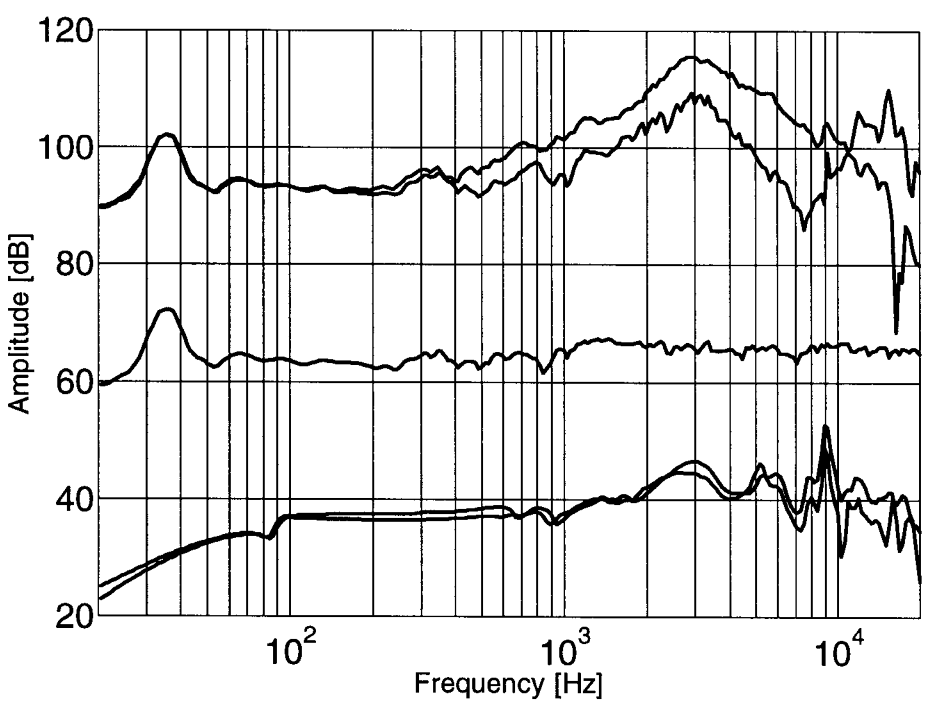

Figure 12: Transfer functions (from the top): left loudspeaker to left ear, left loudspeaker to right ear, left loudspeaker to free field microphone, left headphone and right headphone mounted on HATS. The curves have been shifted apart to increase the global view.

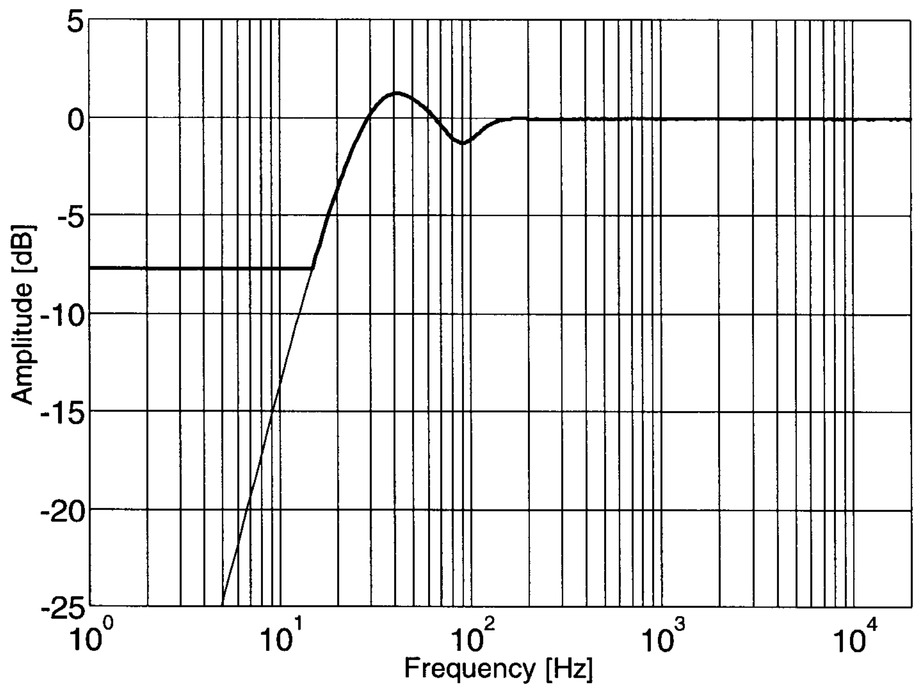

Figure 13: Determined transfer function of LoDist, which is limited below 15 Hz to avoid high gains when implementing the inverse filter.

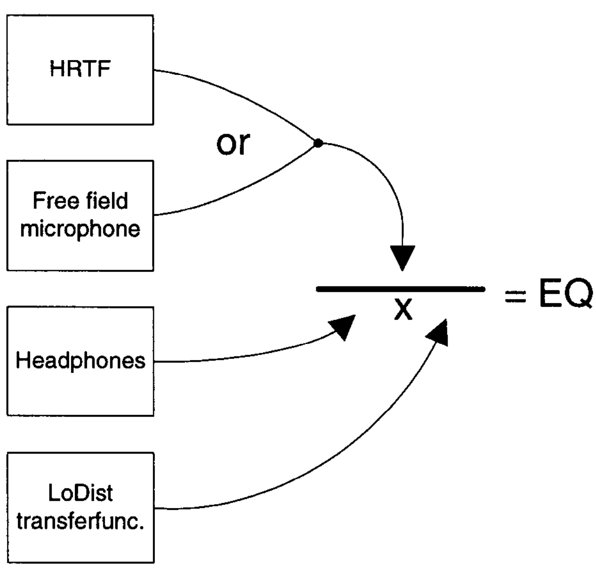

Figure 14: Calculation of the Equalization filter for either the programme material using the Head Related Transfer Functions (HRTF) measurements or the "Pink Noise" using the free field microphone measurements.

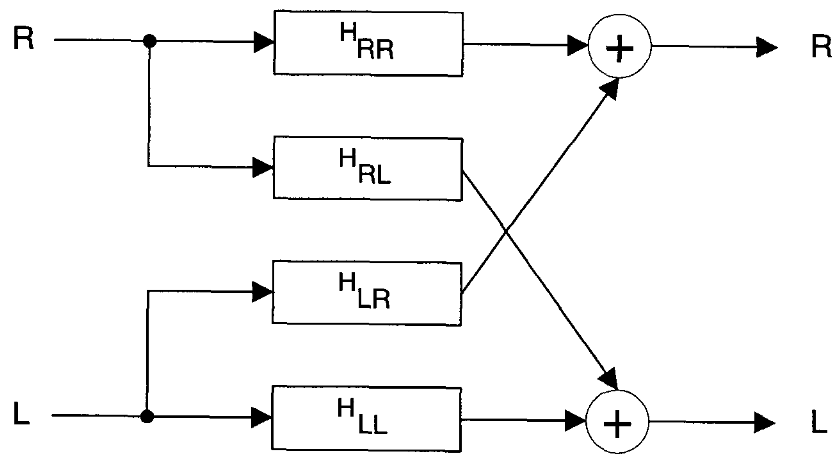

Figure 15: Four different filters, that takes the two signals intended for the loudspeakers into the two signals going to the headphones.

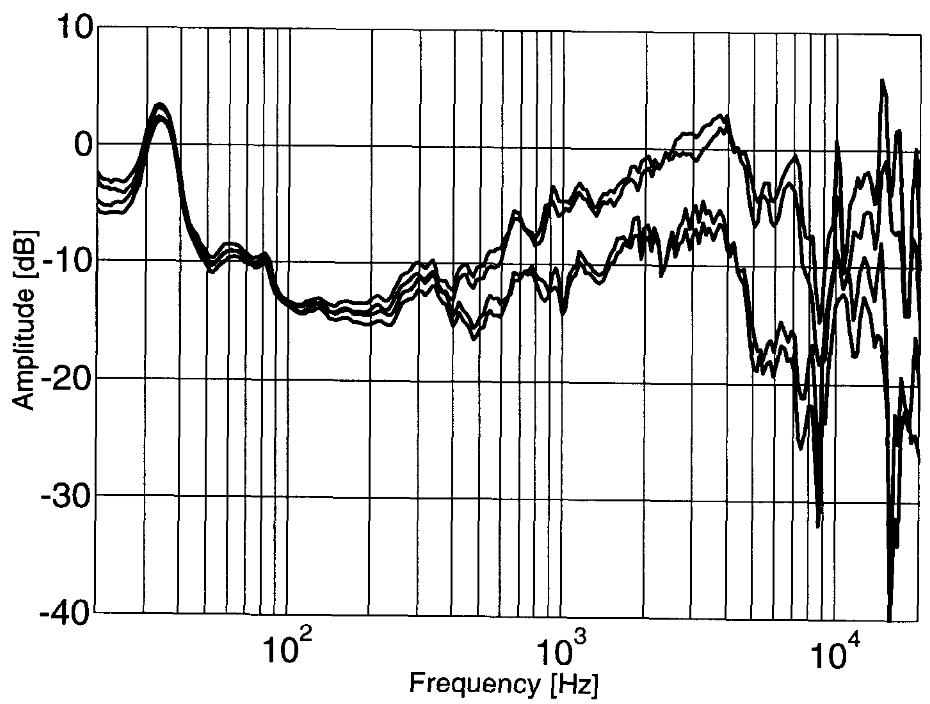

Figure 16: Amplitude response of the 4 filters, which takes the signal through the two loudspeakers to both ears of the HATS and equalized for both the headphones mounted on the HATS and the linear transfer function of "LoDist".

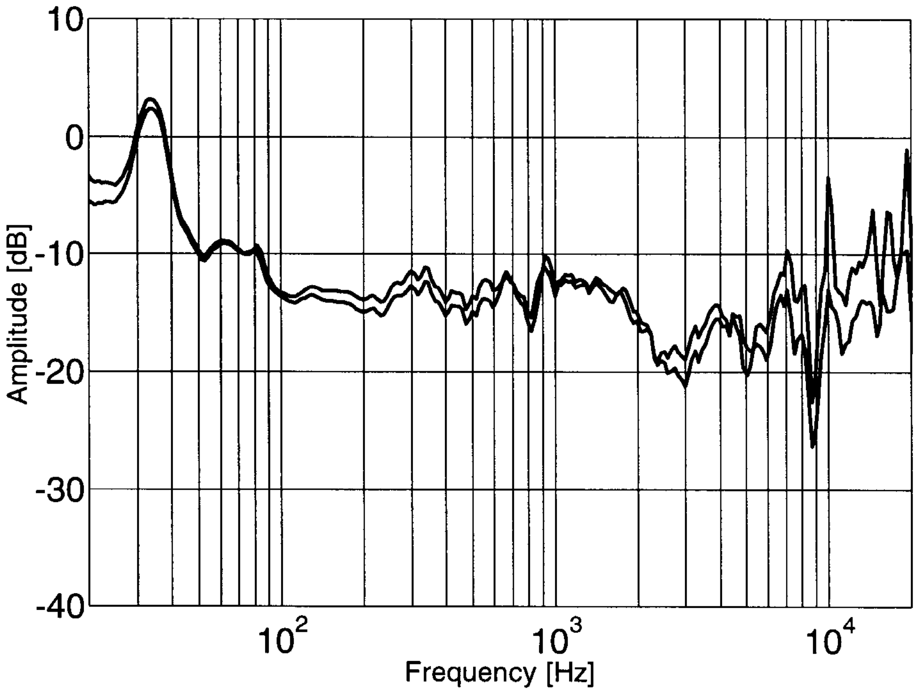

Figure 17: Amplitude response of the 2 filters, which takes the signal through the two loudspeakers to the free field microphone and equalized for both the headphones mounted on the HATS and the linear transfer function of "LoDist".