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**Presented at
the 109th Convention
2000 September 22-25
Los Angeles, California, USA**



AES

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AN AUDIO ENGINEERING SOCIETY PREPRINT

Design of High-Quality Studio Loudspeakers using Digital Correction Techniques

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ABSTRACT

It is well-known that the inherent linear and non-linear distortion characteristics of conventional multi-way studio transducers can only partly be corrected by digital filter techniques, due to the fact that the acoustic output is distributed over the area of interest in a more or less non-uniform way. In this investigation, two different approaches to overcome this problem are presented and compared. First, it is shown that local equalization of a distributed-mode panel loudspeaker around the center of excitation, using the early part of the impulse response, yields a very good result over a wide area. The second approach employs a wide-band waveguide design, where the acoustic output of the compression driver is equalized, and nonlinear distortion is corrected by a cost-efficient implementation of a Volterra filter.

1 Introduction to the Problem

The control of desired space frequency-dependent radiation characteristics is considered an important design criterion for high-quality studio monitors. In 1986 [1], *Toole* has presented comprehensive measured data sets of studio loudspeakers that were available at that time, which included not only accurate on-axis amplitude responses under free field conditions, but with respect to different preconditions, spatially averaged off-axis responses as well. The intention was to obtain frequency curves that reflect more accurately the perception of a listener in a normal room, by taking possible colorations from reflected energy into consideration. It was reported that these so-defined characteristics correlated well with subjective ratings, which had been evaluated by controlled listening tests.

Despite of the fact that results of this rigorous analysis were available to loudspeaker designers, no essential progress in defining new synthesis rules for loudspeakers was reported in the literature and reflected in products since then, with some rare exceptions (e.g. [2]). Refinements of traditional designs succeeded, rather than revolutionary new approaches. However, conventional multiway speakers, which have been designed in accordance with *Toole's* research results, using more or less rectangular-shaped cabinets, have already reached a very high quality level, even though they still contain low-order passive crossovers [3, 4].

Attempts have been undertaken in improving accuracy and performance with the aid of digital pre-filters and digital crossover circuits, which can theoretically be done perfectly by using sufficiently high filter degrees. In practice, however, frequency response correction and spatial control are limited to a certain amount of accuracy, because the acoustical output is spatially distributed, while exact correction is possible at a single point only. Equalisation on axis does not guarantee proper polar response behaviour, due to cabinet diffraction effects and due to the fact that the acoustical centers of the drivers are not coincident [5]. This is the main reason why the few so far existing digital approaches are not necessarily superior to passive ones.

In this paper, the principle according to Figure 1 will be pursued. An electro-acoustical transducer is selected, which is able to generate sufficiently large acoustic pressure levels over a wide frequency range at a small aperture, that can be interpreted as a concentrated element with respect to the shortest wavelength. Since the acoustical output is now no longer distributed in space, its inherent linear and non-linear distortions characteristics can be modelled resp. compensated by digital pre-processing. The separate task remains to design and optimize acoustical systems (e.g. waveguides), such that the specified radiation characteristics and power output demands are met while keeping distortion low. Two completely different solutions will be presented.

The first comprises a cascade of a digital prefilter, a high-quality low-distortion compression driver, and an elliptical waveguide. The system exhibits controlled radiation characteristics with smooth frequency responses over a wide frequency range, and can thus be taken as a wide-band transducer in a two-way system. In addition, inverse nonlinear filters are used in order to compensate for remaining dominant second order distortion characteristics of the horn driver, based on Volterra models.

The second approach is based on Distributed Mode Loudspeakers (DML), a radically new concept that has already been proven to be able to fulfill even very high quality demands [19-22]. A so-called exciter is attached to a surface, initiating a high number of randomly distributed bending modes. The resulting radiation characteristics are similar to those of an ideal point source for the first wave front, followed by a later part that exhibits a diffuse behaviour. It is shown in the paper that the spatially averaged frequency response of the whole system can be controlled by inverting the windowed impulse response which has been measured at the point of excitation, using a digital FIR filter of fairly low degree. In such a way, remaining material resonances can be eliminated. Furthermore, the fact can be exploited that the main cause of nonlinear distortion arises from the driver itself, rather than from the radiating surface. Therefore, appropriate inverse nonlinear prefilters can be applied, similar as in our first approach.

2 Design of Acoustic Waveguides with Digital Equalization

The inverse filters of this section are designed by directly inverting the discrete, complex-valued spectra of given impulse responses, which have been measured with a maximum-length sequence (MLS) excitation and time-truncated, in order to suppress remaining unwanted room reflections. Target functions were appropriately chosen low-pass filters. The impulse response of the equalization filters are computed with the inverse FFT. Of course, causality has to be ensured by providing sufficient predelay. This direct method offers higher flexibility than LMS-based methods [6], because the inversion can easily be restricted to the band of interest, leaving the remaining band unchanged. Amplitude and phase distortions are simultaneously corrected. No significant cost reduction was found in most of the cases, if the minimum phase response was corrected only, as suggested in [6].

The success of the equalization depends on the quality of the system, a horn driver connected to a waveguide. Figures 2-4 document that fact. A first axi-symmetric conical prototype horn (opening angle $\pm 60^\circ$) was used, with a throat of 1.5". Since we intend to design a wide-bandwidth horn, without the need for an additional tweeter, the driver has to cover the whole

range until 20kHz. The first two drivers do not fulfill this condition. In the first case, non-minimum phase components and a notch at 15kHz lead to a non-decaying impulse response with excessive pre-ringing (Figure 2), the second one exhibits a deep notch at 12kHz, which cannot be equalized at reasonable cost (Figure 3). The third driver, however, shows excellent behaviour up to the desired frequency (Figure 4).

The term „acoustic waveguide“ was first introduced by *Geddes* in 1989. In his publication „Acoustic Waveguide Theory“ [7] he introduced a new axisymmetric horn type, which, due to symmetry, can be described by one parameter. Compared to the so far existing „Constant Directivity (CD)“ horns, good loading of the driver resp. efficiency, and directivity control, was achieved simultaneously, while avoiding discontinuities as they occur in two-section CD-horns or horns with a slit at the transition to the outer part (diffraction horns) that may lead to reflections and therefore frequency response irregularities. *Geddes* focused on formulating design equations that optimized the coupling from the horn throat, where plane waves are generated by the compression driver, to an essentially conical outer part, which guides the propagation of spherical waves, yielding frequency-independent directivity characteristics in theory. He also recommended the use of elliptical waveguides that exhibit an better controlled polar pattern along diagonal axes, compared to horns with a rectangular mouth. The theory was restricted to waveguides of infinite length (and thus infinite size). Truncation at the horn mouth to a desired finite length leads to reflections of the waves back to the horn, which causes frequency- and angle dependent amplitude response irregularities, in other words, deviations from the desired constant-directivity.

Johansen has studied these effects in more detail [8]. In his paper, he presented a numerical method to calculate the directional properties of un baffled, finite-length horn loudspeakers. He found through simulations that dividing the horn into at least three conical sections with increasing opening angle leads to a better approximation of constant directivity in a wide frequency range. In particular, the beaming in the frequency band above the cut-off frequency, and below the CD-region („low wave number narrowing effect“) will be reduced.

In practice, the curve at the horn mouth should be smoothed, as *Di Cola et al* showed in their recent paper [9]. They explained the mouth diffraction effects in a more intuitive way, and demonstrated the improved properties of waveguides with a „flared“ mouth experimentally, by building prototypes.

In the following section, two digitally equalized elliptical waveguides, which have been designed in accordance with the above results, will be discussed. They can both be applied in a two-way system with a crossover frequency of about 1 kHz. Throat diameter is 1 inch. They are composed of an inner part, according to *Geddes*, and an exponentially shaped outer section. The transition between the two sections has been chosen different. The first one is slightly longer, and will be referred to as the „long version“ (Figure 5), the second one as the „short version“ (Figure 6).

Figures 7 and 8 show measurements along the long ellipse axis of both types, respectively. The on-axis frequency responses have been digitally equalized (the 0° response is therefore a straight line). We can summarize the following observations:

- There are three overlapping regions with the following properties: Above about 3kHz a fairly good constant-directivity behavior up to the frequency limit of 18kHz, except of a slight attenuation around 6K-10K in Fig. 8, and 4K-6K in Fig. 9. Between 1K and 4K a more pronounced beaming effect, which corresponds to the „long wave number narrowing“ as described in [8]. The effect is less obvious in the short version (Fig. 7). The region where the

horn loses control over its directional properties below 1-1.5K (below the „Break Wave Number“ [8]).

- The wave guide exhibits a very well controlled behavior outside its region of coverage ($\pm 45^\circ$ according to the opening angle of the cone). The frequency responses are attenuated accordingly, but remain flat within about ± 3 dB up to 10 kHz.

In order to obtain a better approximation of constant directivity in the desired range, starting from 1 kHz, the equalization was now performed with respect to the 15° axis. The results are shown in Figure 9 and 10, respectively. The short version now shows a very satisfying result, especially in the listening zone $\pm 30^\circ$. The frequency response variations of the long version (Fig. 10) around 2 kHz are clearly more pronounced.

Finally, Figures 11 and 12 show the corresponding short axis responses (same equalization applied as before, with respect to 15° -long axis). The coverage angle is here $\pm 30^\circ$. A flat response within ± 2 dB can be achieved with the short horn (Figure 11), whereas the long one shows again a slightly worse behavior.

As suggested in [3], averaged responses were computed using the above free-field measurements. Since the waveguides attenuate the signals outside of their coverage areas considerably, the computed curves give a sufficiently accurate estimation of the horizontal mean and the total power responses above the horn's cut-off frequencies, even though the measurements were done only in areas limited by $\pm 60^\circ$. The results, depicted in the Figures 13 and 14, respectively, indicate a very well controlled behavior of both examples. We may even conclude, that those averaged responses should always be accompanied by individual measurements at distinct angles, in order to give a clear picture. In particular, the differences in quality between the two proposed waveguides are less obvious („averaged out“) than in the figures before. However, it can also be concluded, as suggested in [3], that no pronounced resonances are existing, because these should remain visible after averaging. If there were resonances, they had already been removed by the digital prefilter. This confirms the principle that was explained in section 1 of the paper, according to Figure 1.

Finally, Figure 15 shows measured results of a prototype of a large elliptical horn, that was successfully used in a two-way system with a crossover frequency of 450 Hz. Due to the coherent reproduction above that corner frequency, listeners reported improved stereo (phantom-) imaging, less colouration, and better resolution of musical details, compared to a conventional high-quality studio monitor.

Future improvements will include a refinement of the horn curves, using numerical simulation methods, comparable to those presented in [11]. Furthermore, the responses at a dummy head which will be used as a test microphone, will be considered and compared with other types of loudspeakers, such as multiway systems or DML panels. This should give more insight into the requirements concerning the properties of the soundfield that leads to subjective high ratings, especially under nearfield conditions in well-damped rooms (see also [10]).

3 Distortion Correction using Volterra Filters

Aim of this paragraph is to present the current state of the experiments that have been carried out in order to compensate the remaining nonlinear, second-order distortion of a horn driver by applying a nonlinear digital pre-filter. Main focus was to reduce the harmonic and intermodulation distortions that occur at normal listening levels such that they fall below the level of perception. As shown in Figure 16, second order harmonic distortion above 1 kHz is at

least by one order of magnitude higher than higher order distortion components. Here an 1.5“ driver was connected to a conical horn with rapid flare rate, that does not add significant additional errors. The amplitude was chosen such that the total harmonic distortion reached about one percent. We therefore do not try to operate the filter at very high levels, where the device becomes significantly nonlinear. Rather, the investigation should clarify, to what extent such a low distortion has an influence on the perceived sound quality at normal listening levels.

Figure 17 explains how a second order Volterra filter, denoted by the operator \mathbf{H}_2 , works in principle. It represents the second kernel of a time-truncated Volterra series expansion of a weakly nonlinear system. Rather than scaled taps of a delay line, like in a linear FIR filter, scaled products of delayed samples are accumulated. We assume in the model that the system is limited to a memory length N . A matrix of $N \times N$ coefficients is then required to describe the system. The determination of the $N \times N$ model parameters can be carried out by a so-called „Volterra identification“ procedure. A very efficient procedure, in terms of measurement time and required memory, was published by *Reed and Hawksford* in [12]. The authors have shown that a properly designed multilevel test sequence which has been constructed from a binary maximum length sequence of order N by basically shifting and summing, is able to fully excite the nonlinear system under test.

The method has been programmed in MATLAB, as it is described in [12-14]. The test sequences were generated off-line and replayed by using a DSP platform. The resulting signal was captured by a low-distortion microphone and fed back to the DSP after A/D conversion. The final analysis was again performed offline.

Figures 18 and 19 show the resulting second order kernel of the mentioned 1.5“ driver. When measuring at a certain distance from the horn, the resulting signal-to-noise ratio turned out to be insufficient, even if a mean over 10 consecutive measurements was applied. We therefore chose a measurement very close to the horn mouth (Figure 19), in accordance with our underlying paradigm (Figure 1). The complete Volterra model is depicted in Figure 20, comprising a linear filter with the impulse response \mathbf{h}_1 , and the second order kernel \mathbf{H}_2 , connected with an appropriate delay in series, in order to compensate for the different memory lengths of the linear and the nonlinear parts, respectively (compare [13]). Typical signals are shown in Figure 21. The energy of the output of the nonlinear part \mathbf{y}_n , referred to the energy of the output signal \mathbf{y}_a , is **-34 dB**, which can be interpreted as a distortion measure, with respect to the specific test signal used here. The difference of the sequences, obtained by the system under test, and the model, respectively, is a measure for the accuracy of the model. The corresponding signal energy is **-48 dB** (Figure 21), which indicates the maximal achievable distortion reduction of about **14 dB**.

The structure of the required inverse filter is drawn in Figure 22. The inverse of the linear part is required twice. Appropriate time windows are applied, in order to reduce the necessary filter degree. Further explanations can be found in [14]. After windowing, the remaining degrees for the linear filter is about 170, and the nonlinear part 625 (25*25). This could still be implemented in one DSP, regarding the fact that due to symmetry only half of the coefficients of \mathbf{H}_2 need to be computed. Otherwise, efficient frequency-domain algorithms could be applied, as presented in [15] and [16].

4 Application of Distributed Mode Loudspeakers

Distributed Mode Loudspeakers (DML) consist of one or several electrodynamic transducers, called “exciters”, that are attached to a large, thin, stiff and light panel made of some kind of foam material or even thin glass. Locally controlled bending waves are excited in the material, which become more and more irregular and diffuse at an increasing distance to the center of excitation. Sound radiation can be compared with a nearly ideal point source, offering an omnidirectional polar response, combined with a diffuse part that adds sound energy a few milliseconds time-delayed. This energy contributes significantly to the perceived loudness and efficiency, without disturbing localization. It is even possible to feed the exciters that are connected to the same panel with different input signals, such that the early response is equivalent to an acoustical superposition of different, possibly delayed point sources. In such a way, any desired directivity characteristic may be synthesized, by making use of Wave Field Synthesis principles [21]. A comprehensive introduction to DML, together with a reference list, is given in [17] by *H. Azima*. Measurement aspects specifically related to DML’s are explained in [19]. In [18] distortion mechanisms of DML’s are discussed and compared with conventional loudspeakers. It is concluded, that the main cause of nonlinear distortion arises from the driver itself, rather than from the radiating surface. With optimized driver design, the distortion can be typically by 10 dB lower, compared with equivalent, conventional piston speakers. Accordingly, the same non-linear correction schemes may be applied, as presented in the preceding section 3.

Some first experiments have been conducted with respect to linear pre-filtering of a large DML panel (70cm x 100cm), which allows to examine the frequency responses around the point of excitation, irrespective of boundary effects. The results are depicted in the Figures 24 – 26. First, the impulse response was measured at the point of excitation in the nearfield, at a distance of about 30cm. The obtained signal was gated with a short rectangular window (2msec), and the result inverted using the same procedure as in section 2 of this paper. A correction filter of degree 150 was sufficient, in order to obtain amplitude and phase compensation simultaneously (Figure 24). Figure 25 shows nearfield measurements of the uncorrected panel at two positions which are laterally displaced from the reference point, using full length impulse responses. In Figure 26 the same measurements are shown, but with the prefilter connected in cascade. The one-third-octave smoothed versions indicate clearly, that the equalization was successful, even at different positions than the reference point. We can therefore conclude that, rather than performing the equalization based upon a far-field magnitude response, as suggested in [6], a local (complex) equalization around the point where the exciter is attached leads already to convincing results.

The benefit of the designed filter has been confirmed by informal listening tests. Strongly reduced colourations in the mid frequency band, combined with a much better resolution of musical details, has been reported, which leads to the final statement that this type of transducer may become a very strong alternative, even for high-quality monitoring applications.

5 Conclusion and Outlook

Two different approaches have been presented, that could become building blocks of future generation studio and home loudspeakers. Each of them exploits the capabilities of digital

systems to eliminate linear and non-linear distortion characteristics of electro-acoustical transducers. In the first approach, an acoustical waveguide exhibiting particularly low distortion was used in conjunction with a digitally corrected horn driver. The second one dealt with a Distributed Mode Loudspeaker panel that was equalized locally. In future applications, a number of exciters that have been equalized individually, could be attached to a panel in a regular arrangement, in order to realize a loudspeaker with controlled and adjustable directivity characteristics, by applying Wave Field Synthesis principles [21].

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FIGURES

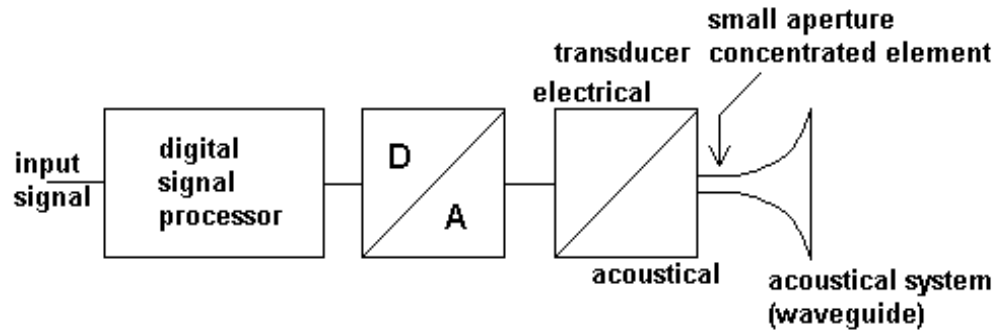


Figure 1: Providing an undistorted acoustical input signal by digital pre-processing

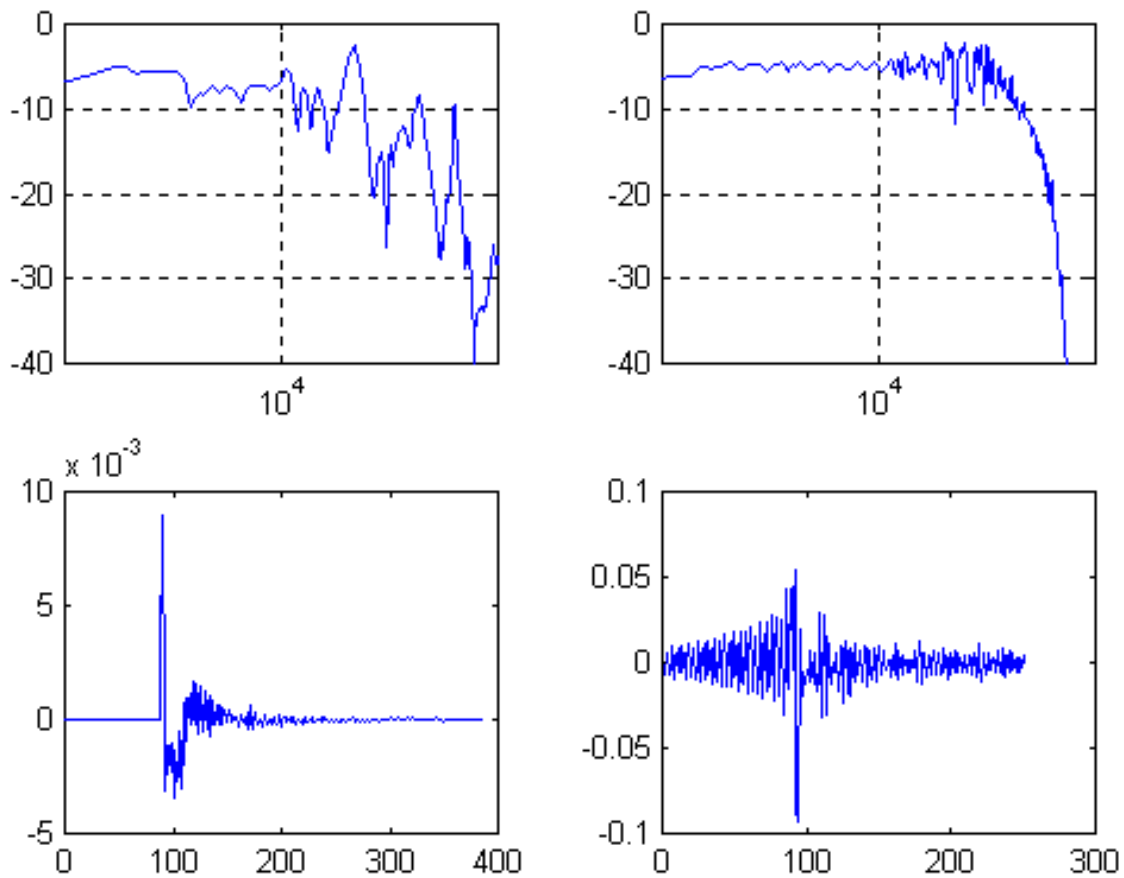


Figure 2: Transducer with apparent non-minimum phase signal components. Top left: measured frequency response 5..10 kHz; Bottom left: corresponding impulse response; Top right: Equalized frequency response; Bottom right: Impulse response of inverse filter

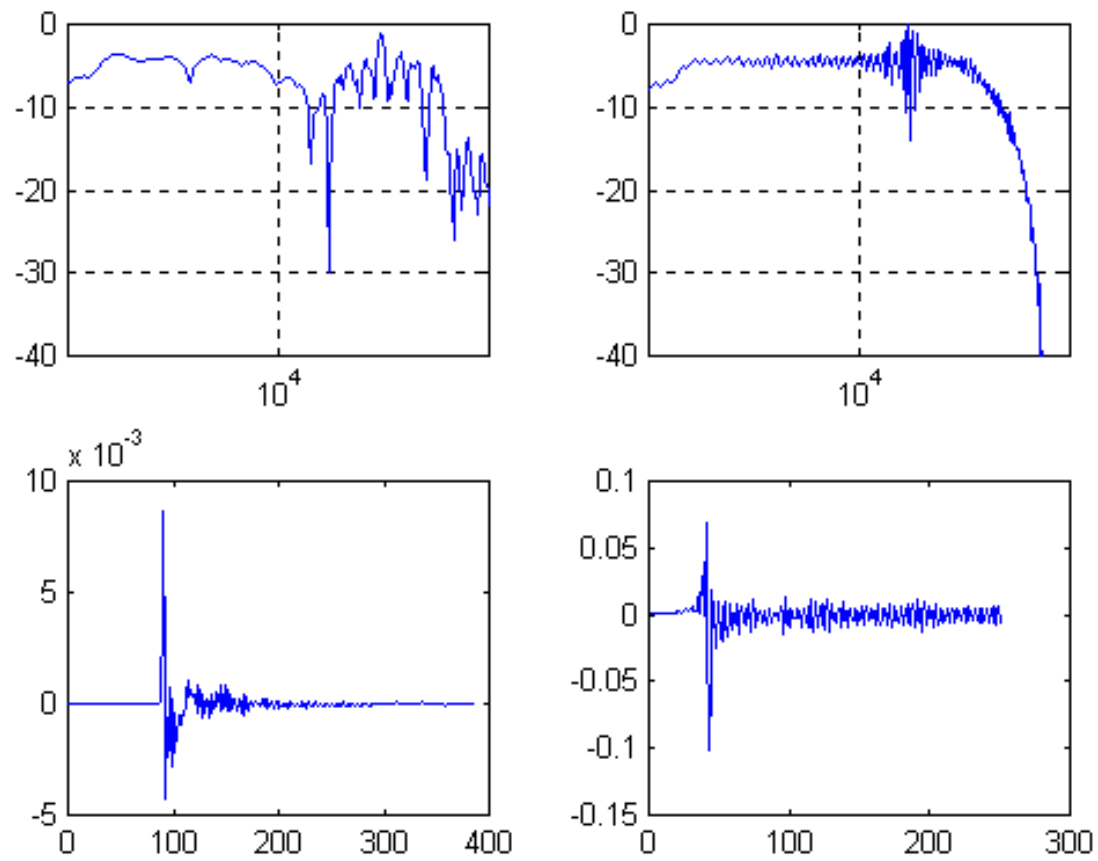


Figure 3: Predominant minimum phase but notch at 12kHz. Top left: measured frequency response 5..10 kHz; Bottom left: corresponding impulse response; Top right: Equalized frequency response; Bottom right: Impulse response of inverse filter

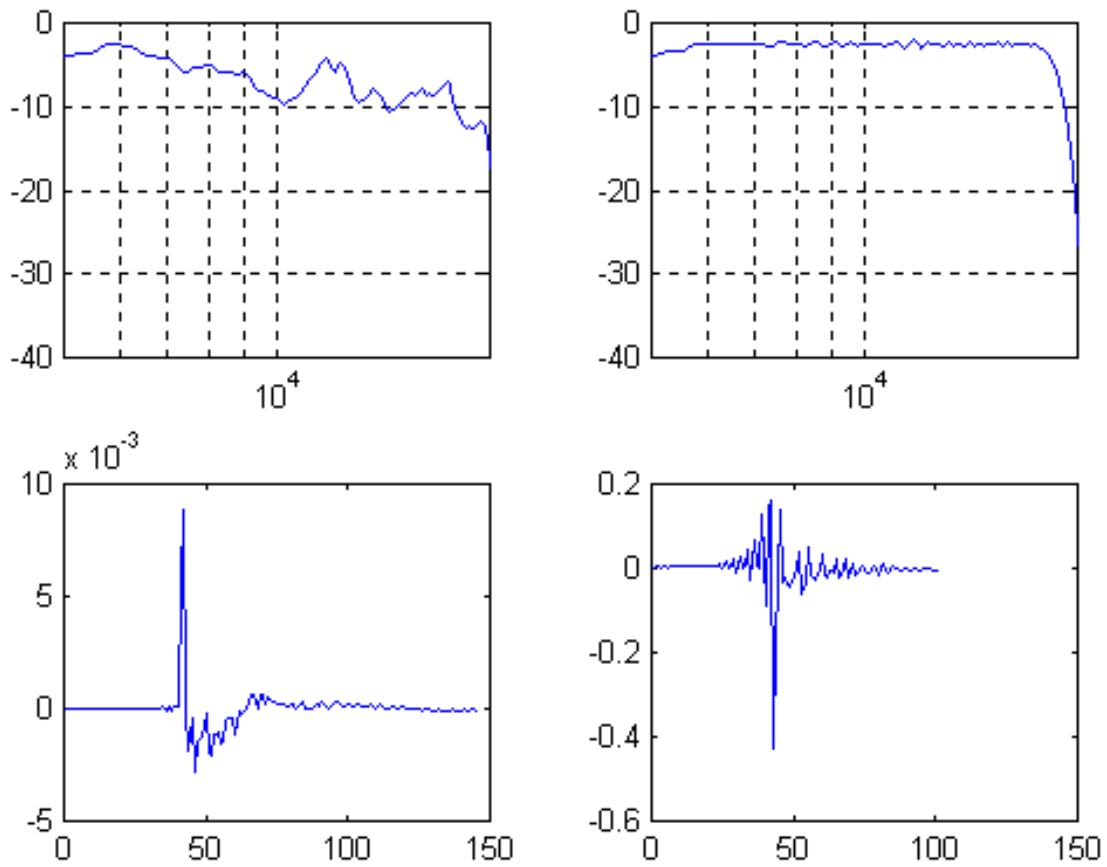


Figure 4: Wide-bandwidth transducer. EQ of degree 100 sufficient. Top left: measured frequency response 5..10 kHz; Bottom left: corresponding impulse response; Top right: Equalized frequency response; Bottom right: Impulse response of inverse filter

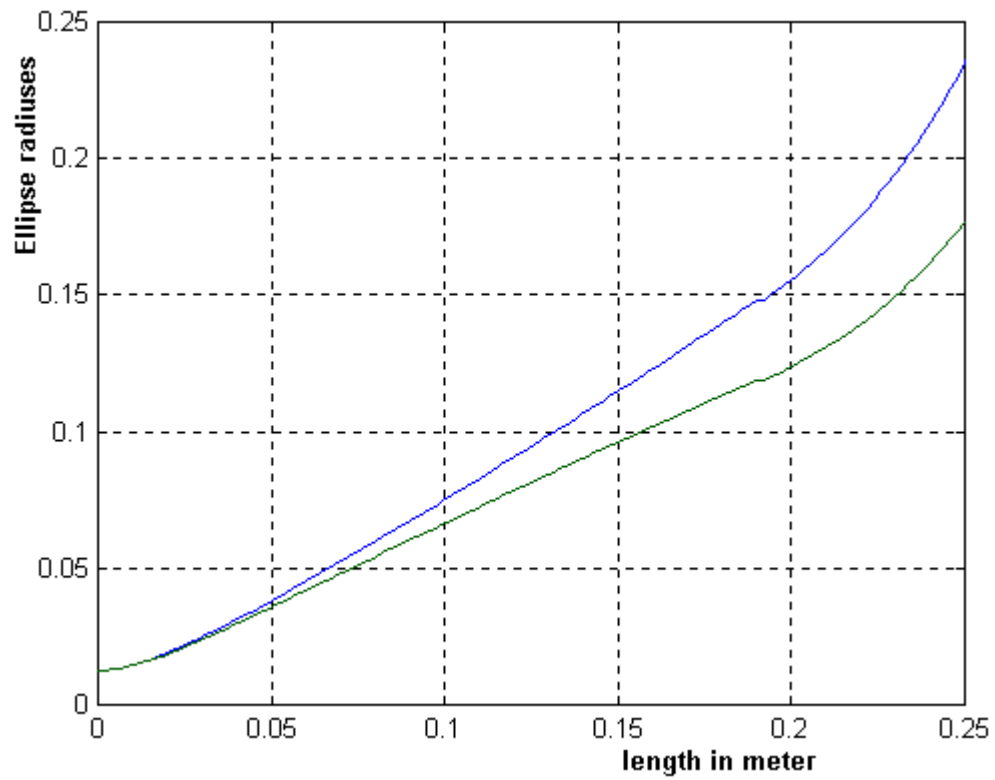


Figure 5: Elliptical horn 35cm x 46cm „long version“

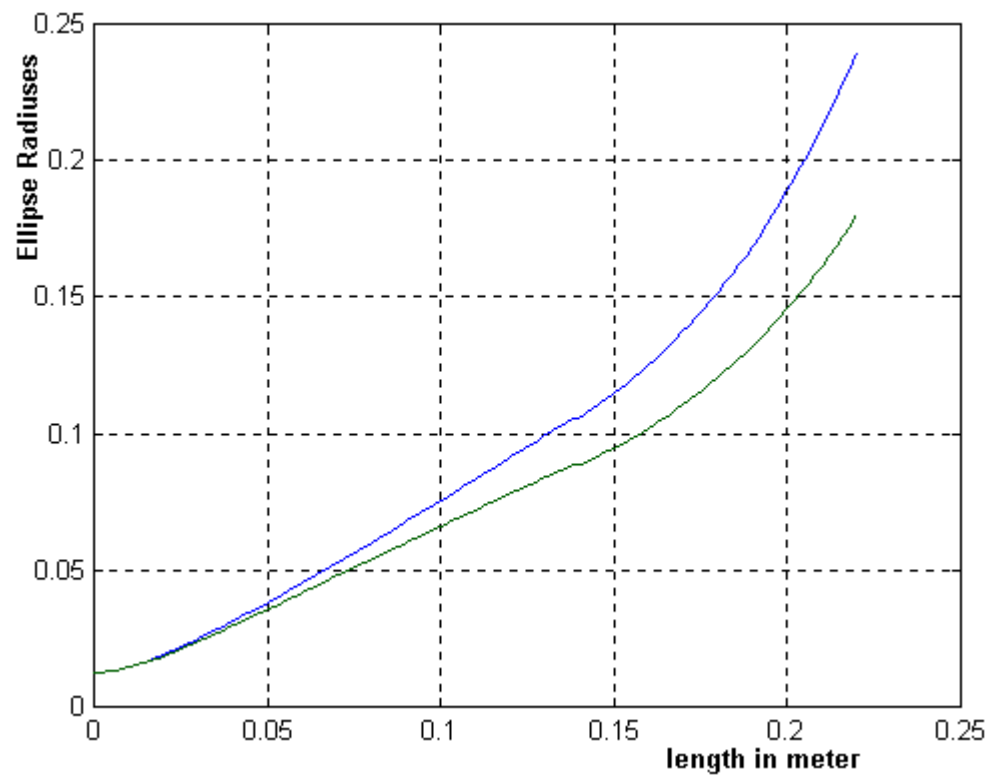


Figure 6: Elliptical horn 35cm x 48cm „short version“

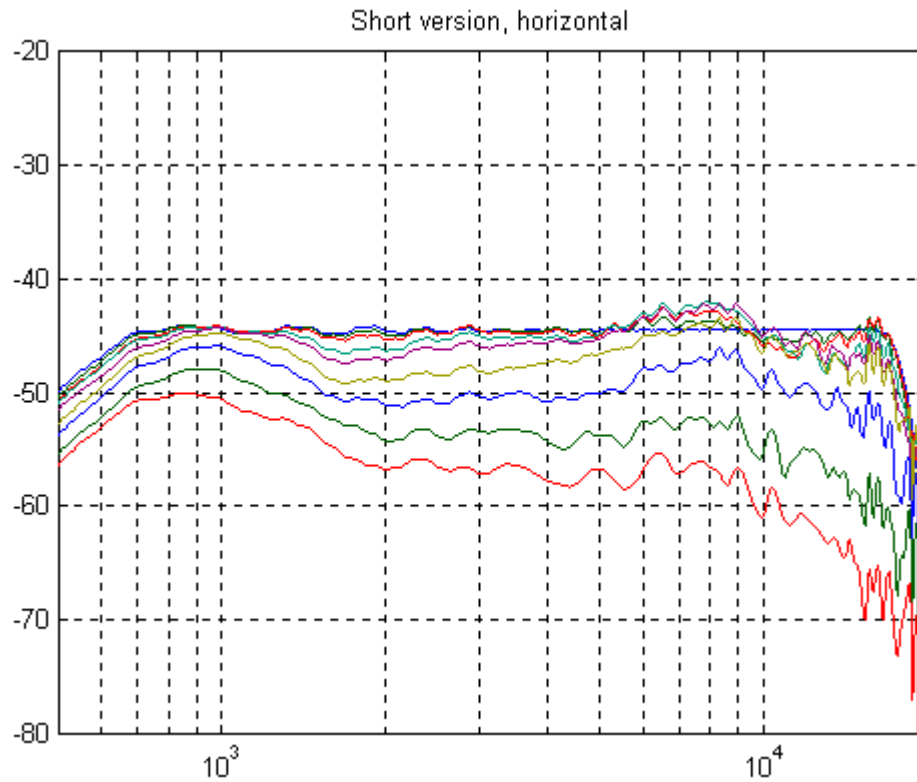


Figure 7: Measured frequency responses of short horn (figure 6), at 0°, 5°, 10°, 15°, 20°, 30°, 40°, 50°, 60° (top to bottom); long axis of ellipse; equalized at 0°

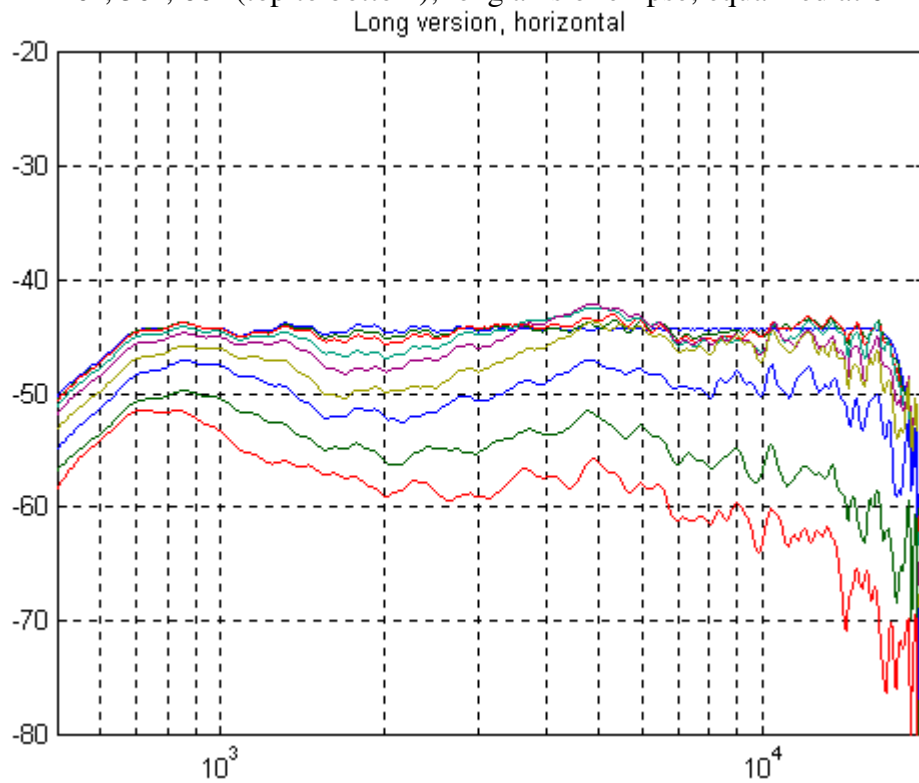


Figure 8: Measured frequency responses of long horn (figure 5), at 0°, 5°, 10°, 15°, 20°, 30°, 40°, 50°, 60° (top to bottom); long axis of ellipse; equalized at 0°

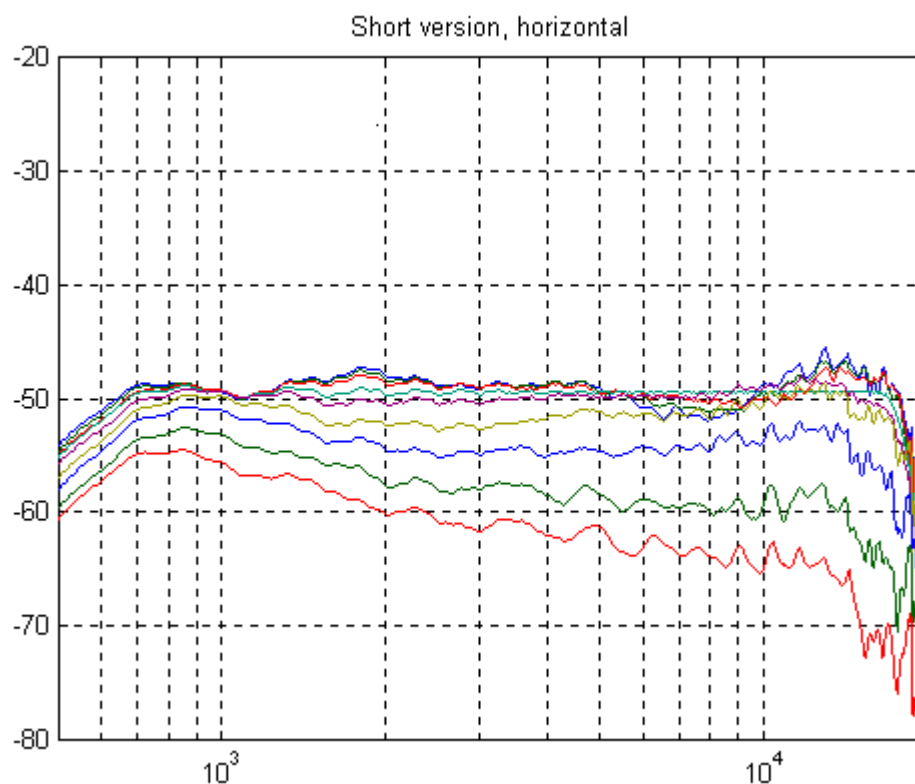


Figure 9: Measured frequency responses of short horn (figure 6), at 0°, 5°, 10°, 15°, 20°, 30°, 40°, 50°, 60° (top to bottom); long axis of ellipse; equalized at 15°

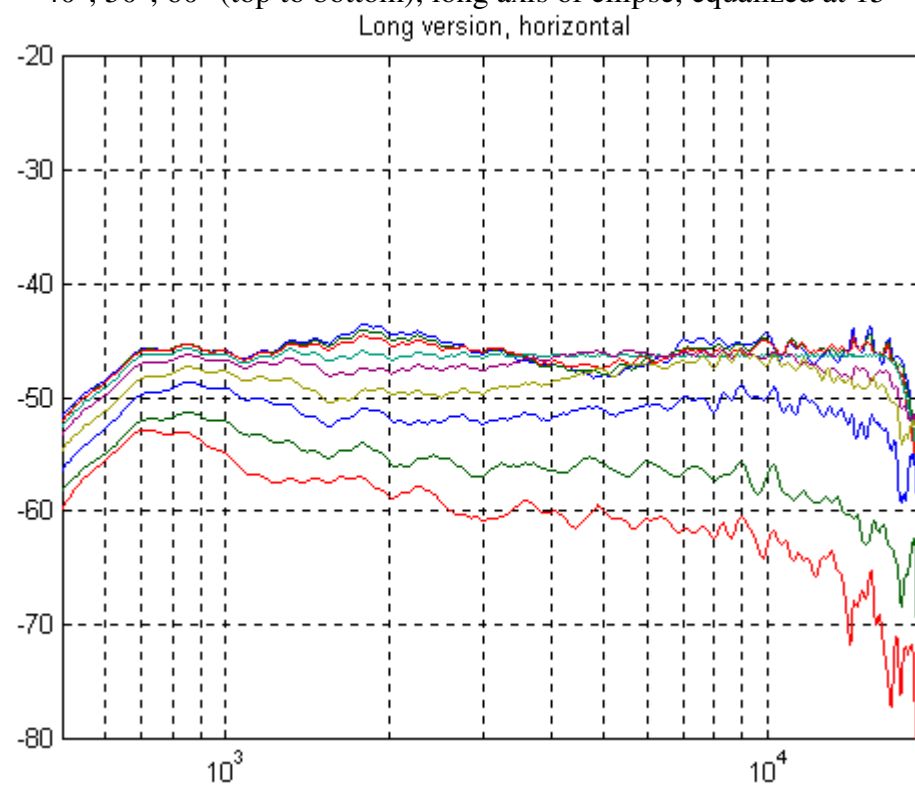


Figure 10: Measured frequency responses of long horn (figure 5), at 0°, 5°, 10°, 15°, 20°, 30°, 40°, 50°, 60° (top to bottom); long axis of ellipse; equalized at 15°

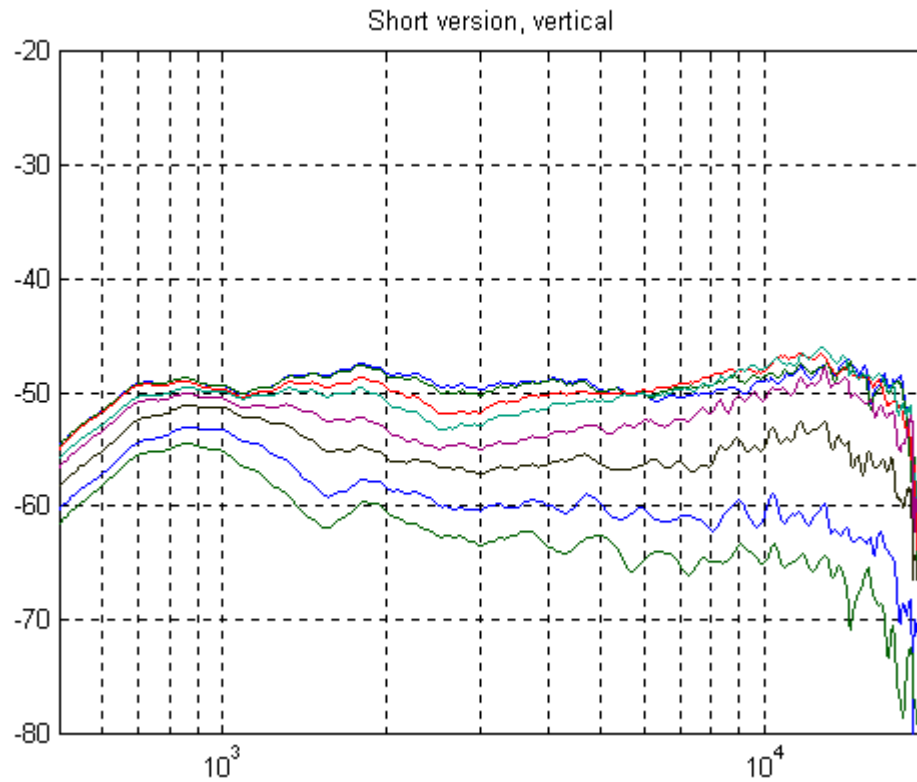


Figure 11: Measured frequency responses of short horn (figure 6), at 0°, 5°, 10°, 15°, 20°, 30°, 40°, 50°, 60° (top to bottom); short axis of ellipse; equalized at 15°

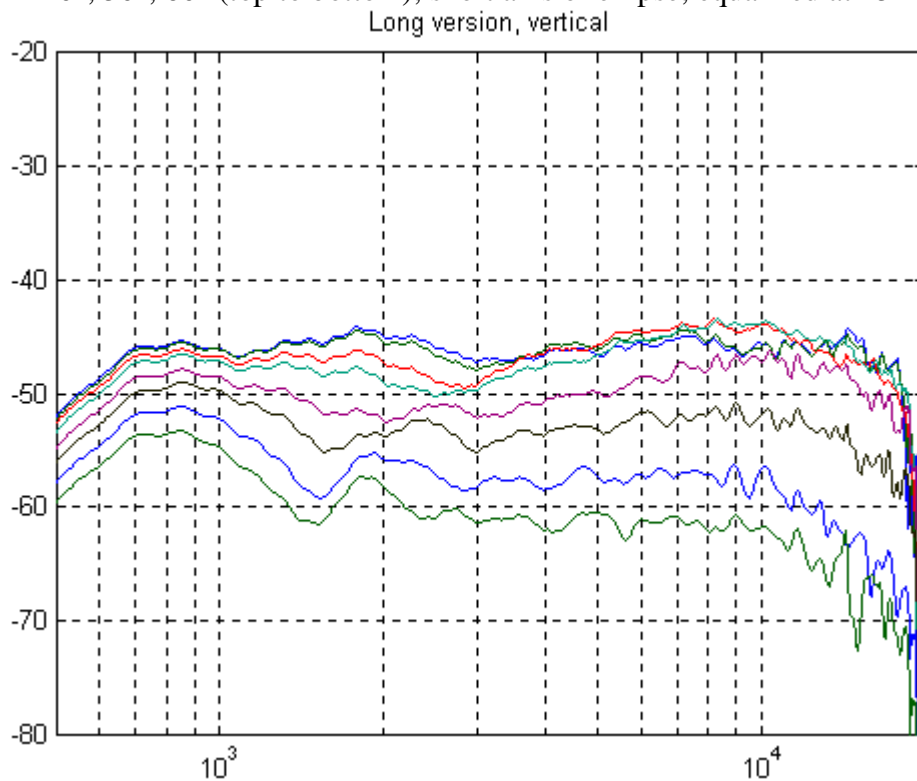


Figure 12: Measured frequency responses of long horn (figure 5), at 0°, 5°, 10°, 15°, 20°, 30°, 40°, 50°, 60° (top to bottom); short axis of ellipse; equalized at 15°

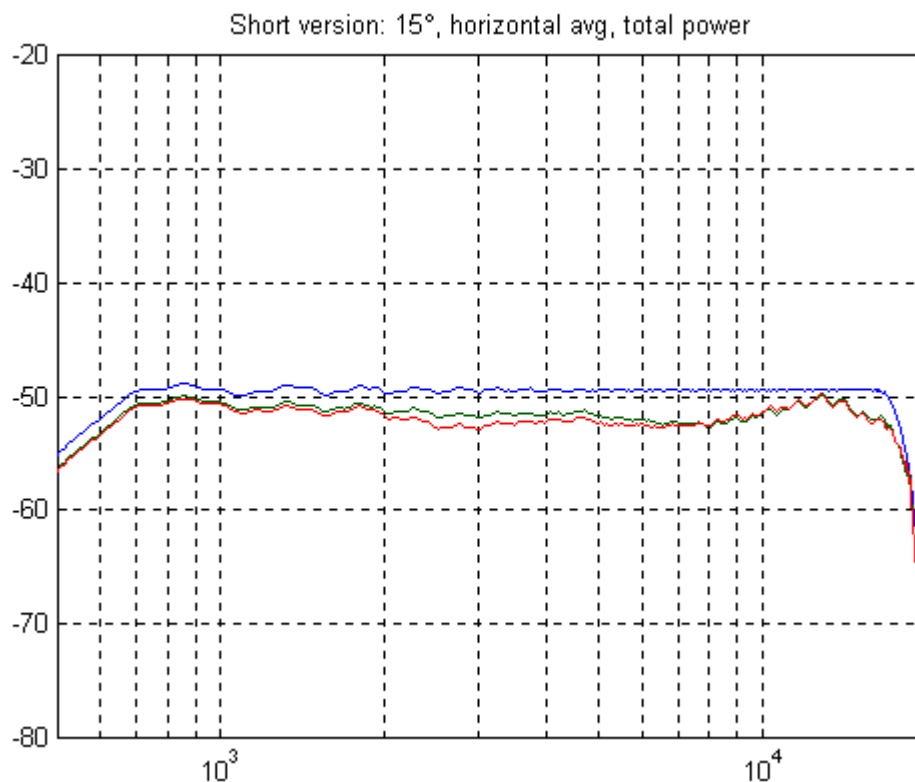


Figure 13: Short horn (figure 6), Top to bottom: Equalized response at 15°, Average over horizontal responses (long axis), Total power response

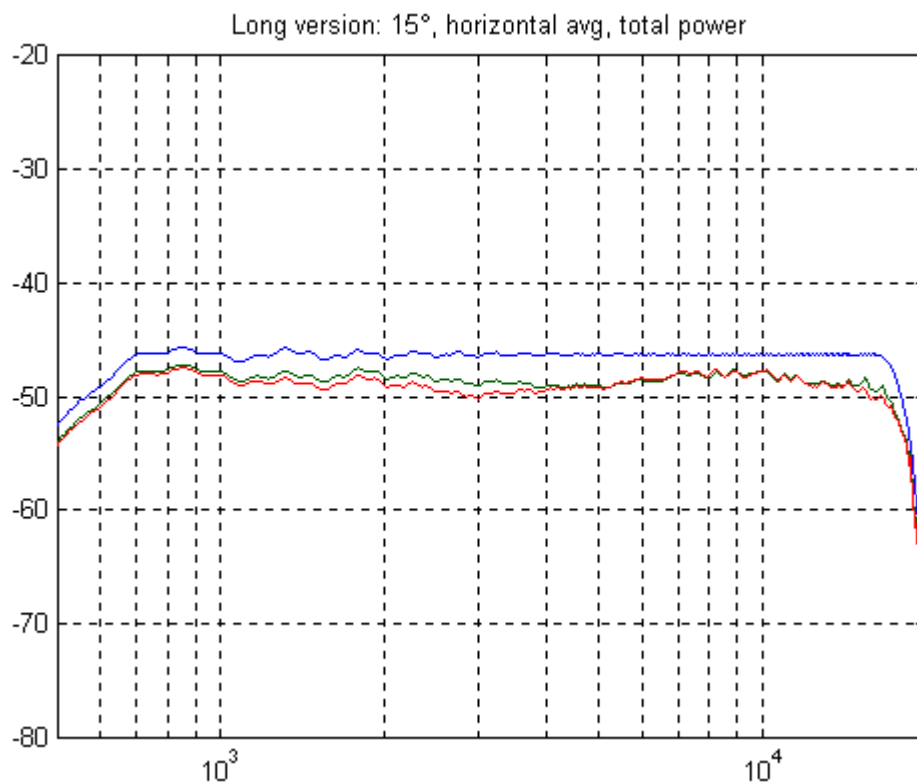


Figure 14: Long horn (figure 5), Top to bottom: Equalized response at 15°, Average over horizontal responses (long axis), Total power response

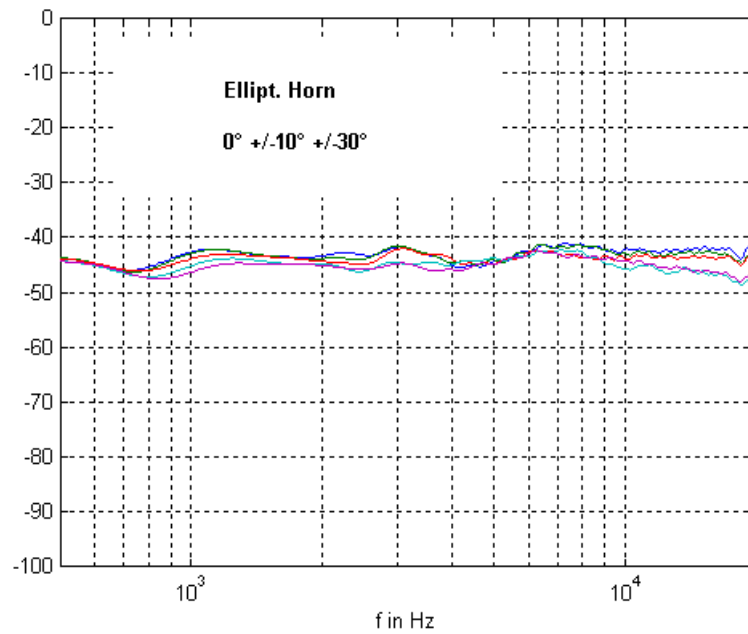


Figure 15: Measured frequency responses of an equalized large elliptical horn (60cm x 90cm), at 2 m distance, throat 1.5“

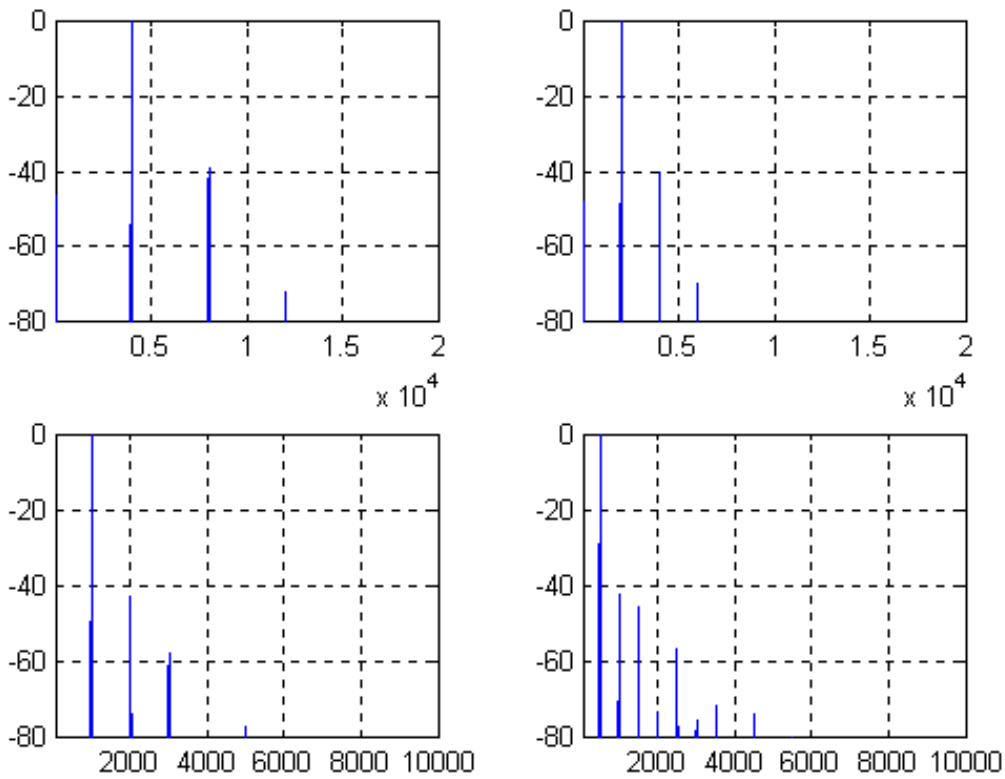


Figure 16: THD measurements of 1.5” horn driver. Top left: 4K, right: 2K. Bottom left 1K, right 0.5K input signal

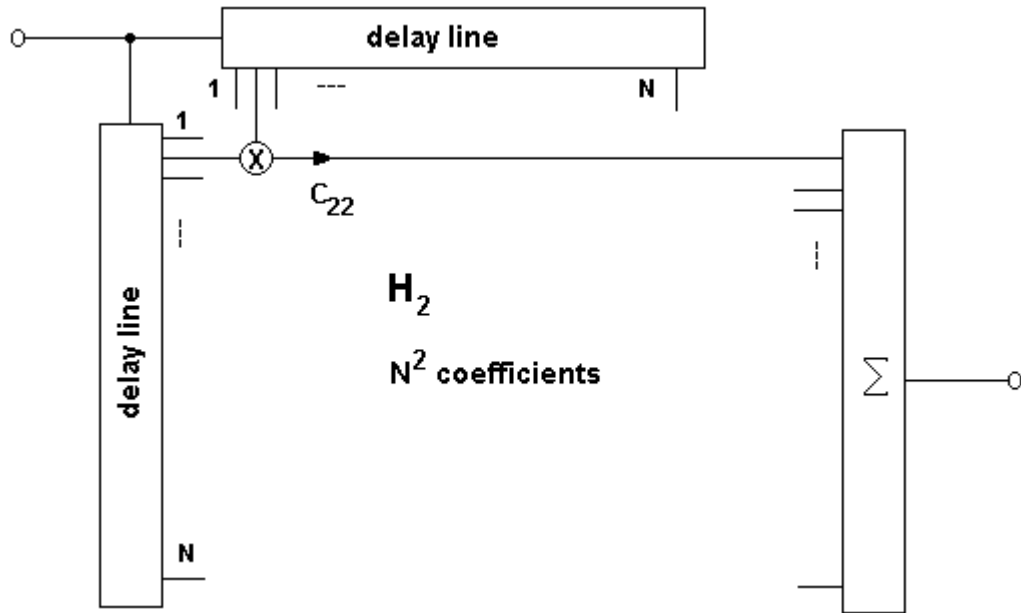


Figure 17: Filter, implementing second order Volterra kernel

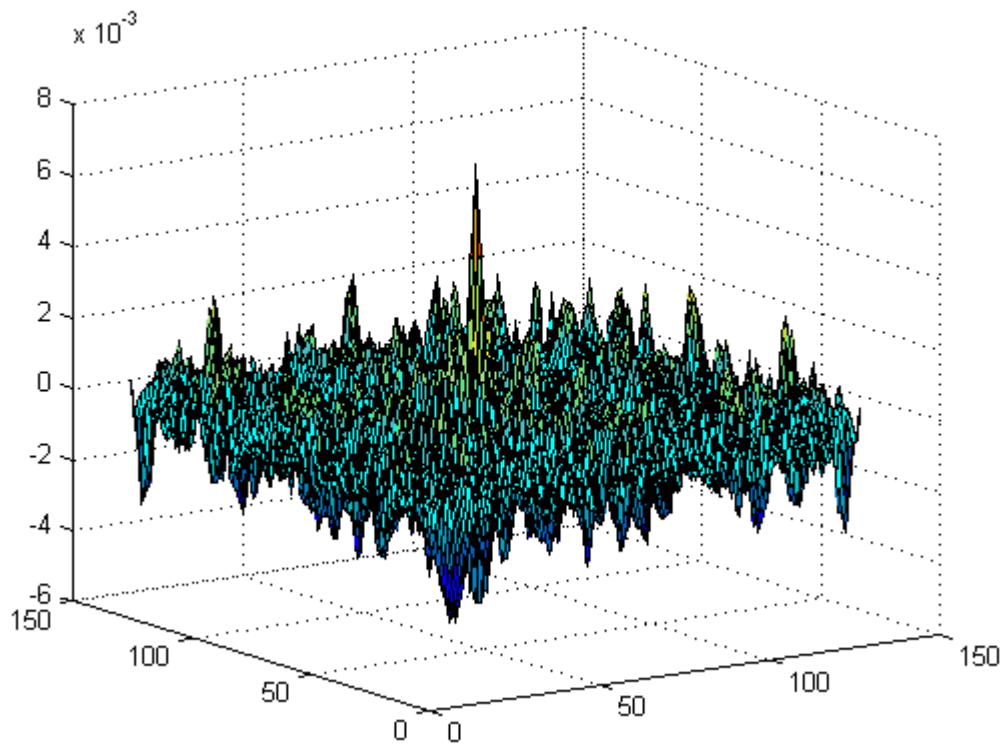


Figure 18: Measured second order Volterra kernel 0.5 m from horn mouth

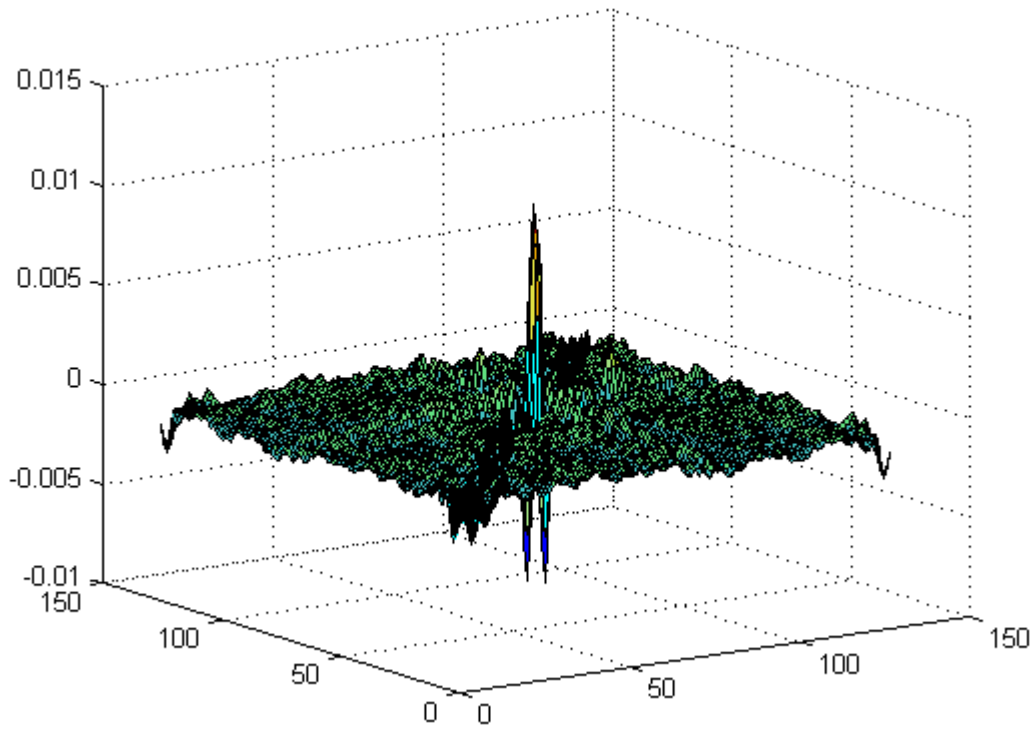


Figure 19: Measured second order Volterra kernel, close to driver

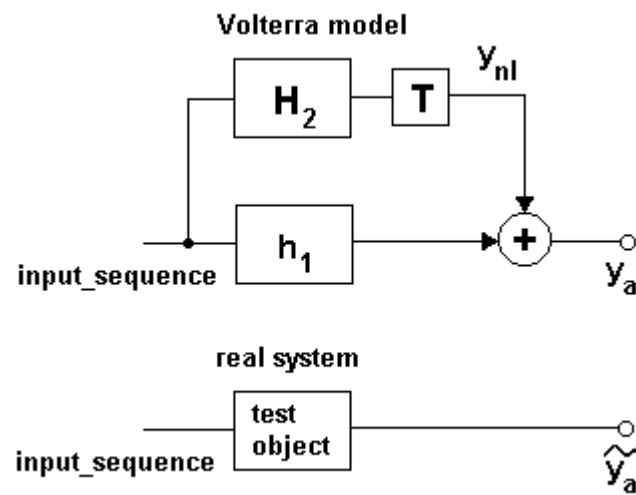


Figure 20: Output sequences of model y_a , test object \tilde{y}_a , and Volterra filter y_{nl}

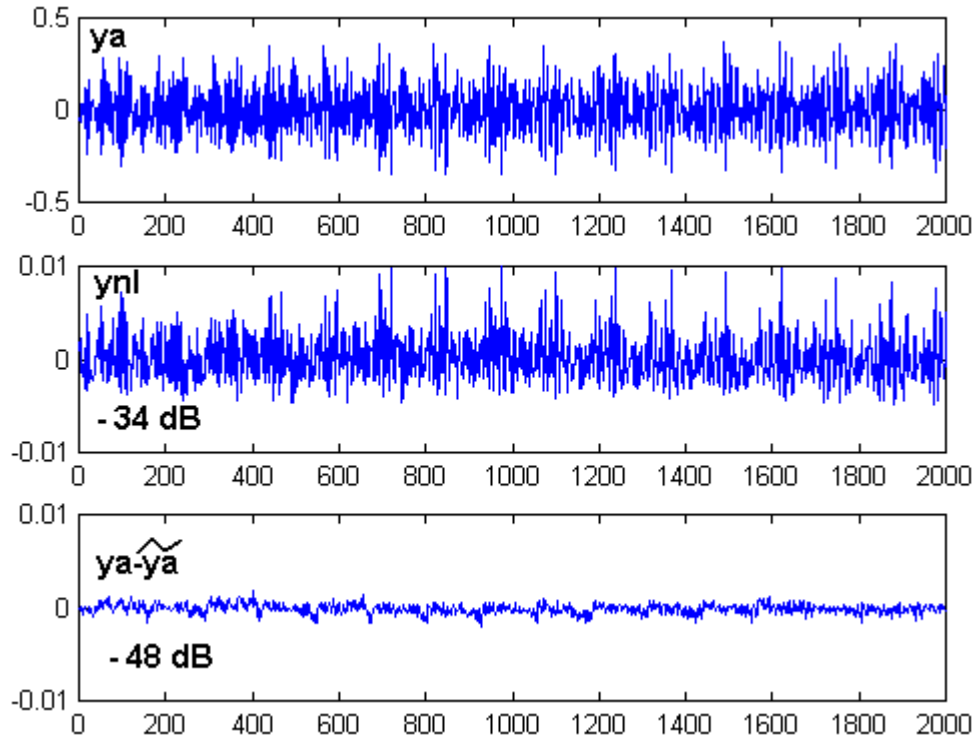


Figure 21: Top to bottom: Output sequence y_a , nonlinear part y_{nl} with total energy -34 dB, and difference between modelled and measured output sequences, residual energy -48 dB

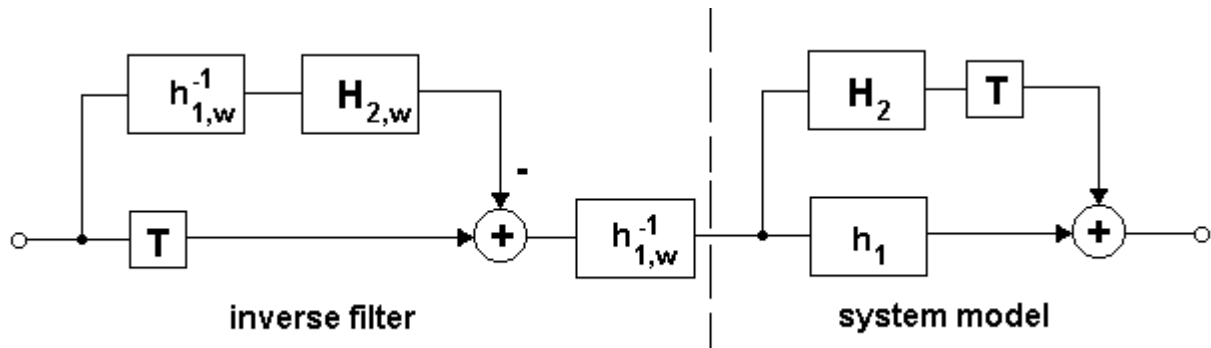


Figure 22: Cascade of inverse filter and Volterra model

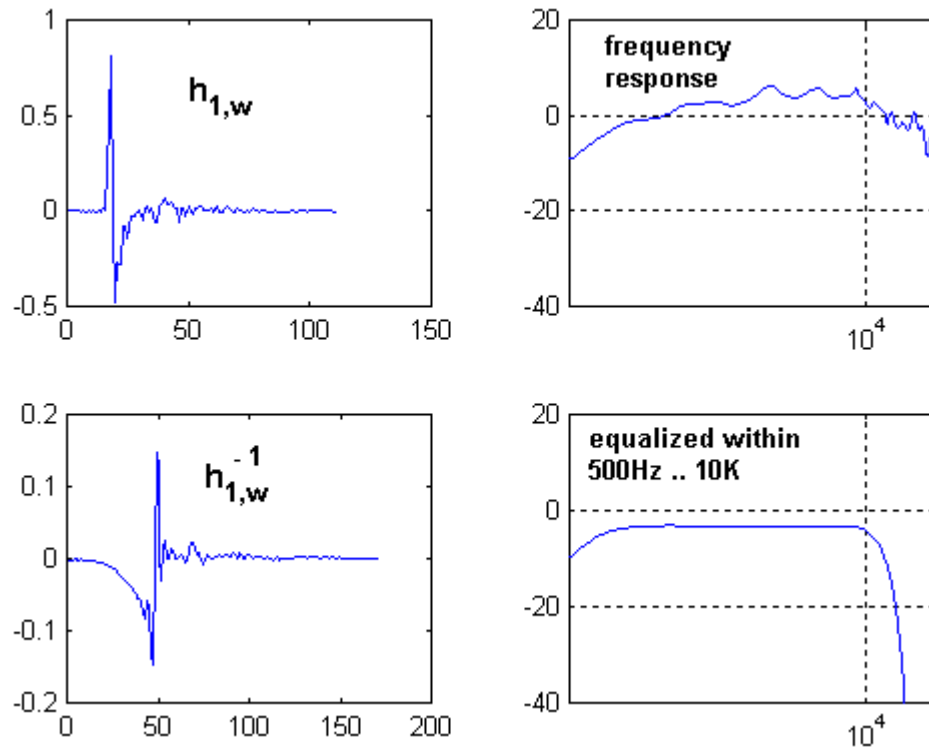


Figure 23: Top: Measured first Volterra kernel (windowed impulse response of linear part).
Bottom: Inverse system, and resulting bandlimited equalized response

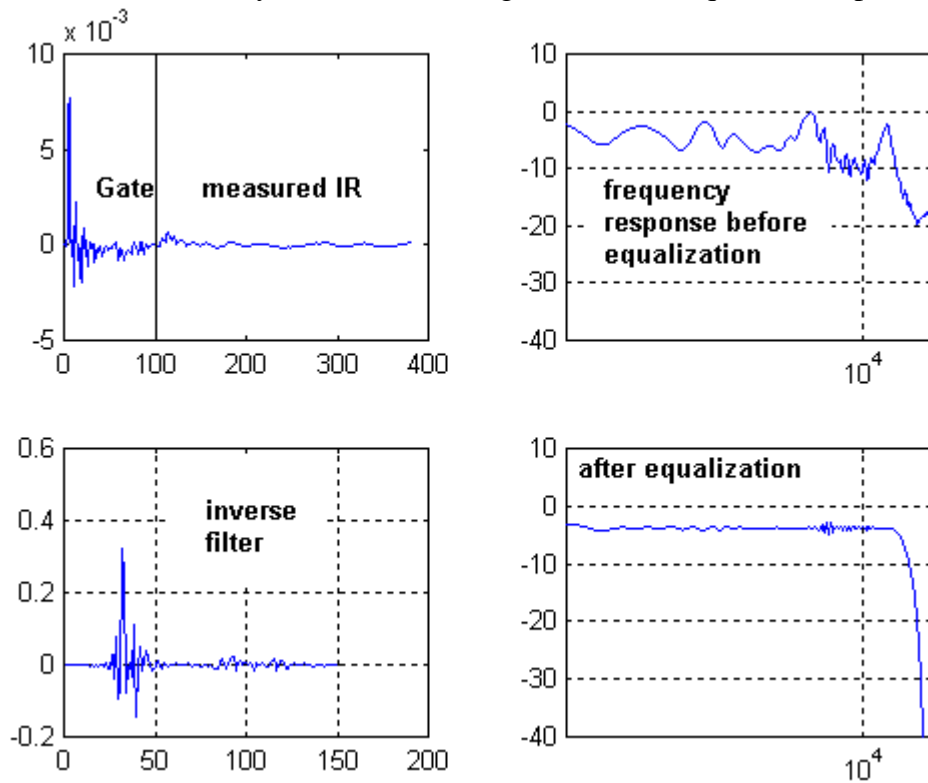


Figure 24: Equalization of DML Panel using near field, gated impulse response

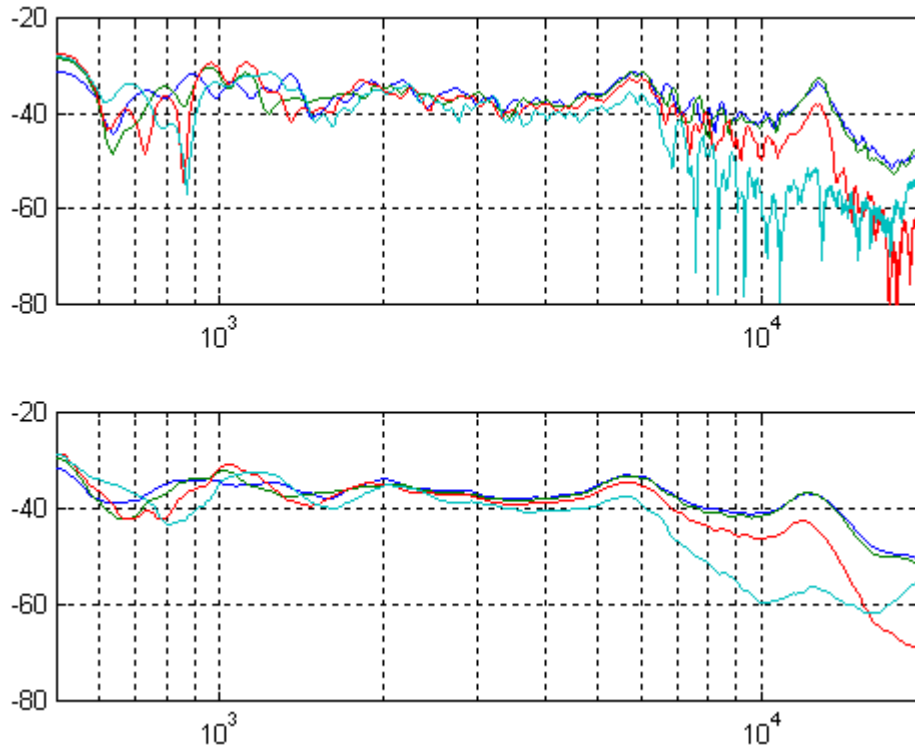


Figure 25: Measured nearfield frequency responses of DML panel loudspeaker (size 0.7m x 1m), at 30cm distance, at position of exciter, resp. 20cm and 40cm lateral. Top: high-resolution MLS measurement, Bottom: One-third-octave smoothed

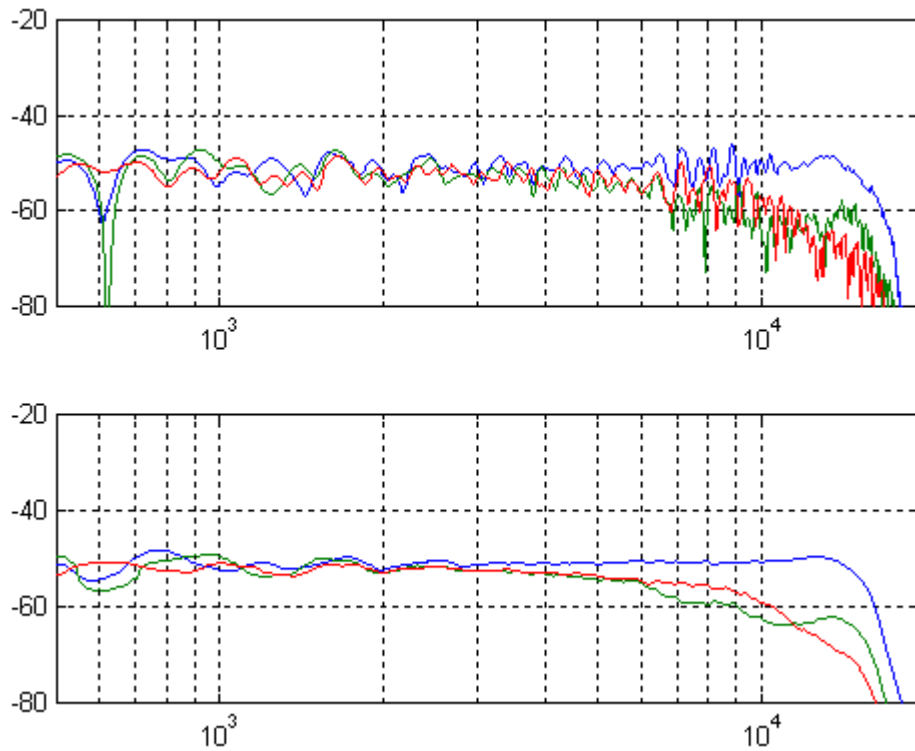


Figure 26: As above, with EQ filter applied, positions as in Figure 25