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Strategies for bass enhancement in Multiactuator Panels for Wave Field Synthesis

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ABSTRACT

In this paper, an extension of the useful bandwidth in the low frequency band is applied to compensate the inherent poor bass response of Multiactuator Panels. These are a special type of flat panel loudspeakers that are commonly used to reproduce spatial audio under the Wave Field Synthesis system. The proposed algorithm combines two strategies: first, a dynamic electrical equalization, applied to additional exciters, which are carefully positioned to excite the lower frequency modes of the panel; second, a psycho-acoustical approach, taking profit of the behavior of human hearing based on the missing fundamental principle. For comparison purposes, the shelving and peak equalizations are also applied to the MAP prototypes developed in this paper. Objective and subjective results show that the combined approach results in an effective extension in the low frequency end of MAP with very low levels of distortion, outperforming conventional equalization methods, with better low frequency behavior, and hence better audio quality perception.

1. Introduction

Distributed Mode Loudspeakers (DML) are flat panel loudspeakers, the transduction technology of which is based on the distributed mode operation [1,2]. A DML consists of a thin panel that is set into vibration by means of a special electroacoustic transducer called an exciter or actuator [3]. The exciter is normally a moving coil device, which is carefully positioned and designed to excite the natural resonant modal structure of the panel optimally. One of the main benefits of DML is that the polar response is quasi-omnidirectional over the audio frequency band and is substantially independent of its size [4]. DMLs can easily be integrated into a living room because of its low visual profile [5,6], being unnoticed as part of walls or furniture. Due to these properties, DMLs have been proposed as loudspeaker arrays for multichannel audio systems, such as Wave Field Synthesis (WFS) [7–9] in which a set of exciters is attached to a single vibrating surface, each exciter being driven by an independent signal to create a given acoustic field. In this context, DMLs which are excited by several exciters are known as Multiactuator Panels or MAPs [10,11].

However, since the radiation of MAPs is based on the excitation of modal frequencies to create a seamless modal density, their typical frequency response is irregular, with large dips and peaks [10]. They also present a poor bass frequency response since the useful bandwidth at low frequencies is determined by the lowest excitation modes. To excite these modes, panels must have impractical dimensions. To compensate for the irregular frequency response of MAPs, the authors have already proposed an efficient equalization method in [12]. The drawback with the poor low frequency response is generally solved by placing a supporting subwoofer elsewhere in the room, which handles the low frequency end of all channels. Notwithstanding, providing a proper bass frequency response to the panel by no external means is still an open problem.

In this paper, an extension of the useful bandwidth in the low frequency band is applied to MAPs with a noticeable perceived low frequency improvement behavior by combining two strategies. On the one hand, a level- and frequency-controlled electronic boost (dynamic filter) is applied to properly positioned exciters for physically improving the response at low frequencies. On the other hand, in order to create an illusion of an improved low frequency extension, the psychoacoustic phenomena of the missing fundamental is applied [13]. Although the methodology is valid for DMLs as a general purpose low frequency extension, this paper is focused on the application to MAPs for WFS. Since exciters are located equidistantly on a line onto the panel to follow the requirements of WFS [7], there still exist free locations which would excite lower frequency modes than those of the WFS exciters. This advantage is used here by placing additional exciters at accurate positions to generate the bass frequency band that is missed by the WFS exciters. Since the proposed approach employs the already generated WFS signals, there is no need of processing more additional WFS channels to feed the extra LF exciters.

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The paper is organized as follows. An overview of MAPs with application to WFS is given in Section 2. Section 3 describes the fundamentals of the bass enhancement by means of the combined strategy that is applied to the panels. The two MAP prototypes in which the methodology is applied and the measurement setup is given in Section 4. According to these measurement procedures, in Section 5 the method is applied to both prototypes, the results of which are discussed in terms of sound quality and psychoacoustic performance. To study the impact of the method in the perceived bass quality, some subjective assessments are carried out. Finally, the paper is concluded in Section 6.

2. Multiactuator Panels for Wave Field Synthesis

The output signals of multichannel audio systems, such as WFS, are reproduced by a number of transducers distributed in a loudspeaker array. This section addresses MAPs as arrays for the WFS reproduction stage.

2.1. Wave Field Synthesis

WFS is a spatial reproduction technique that, by analogy to holography and based on the Huygens's principle, reproduces an acoustic field inside a volume from the stored signals recorded in a given surface. Huygens's principle states that the wave front radiated by a source behaves like a distribution of sources that are in the wave front, named secondary sources, together creating the next wave front.

WFS was first proposed with application to 3D sound by Berkhout [7]. The synthetic wave front is created by loudspeaker arrays that substitute the individual secondary sources. The ideal situation would be when an area, which is completely surrounded by loudspeakers, is fed with signals that create a volumetric velocity proportional to the particle velocity normal component of the original wave front. The application of planar loudspeaker arrays, as prescribed by the Huygens's principle, would involve a very high number of loudspeakers and reproduction channels. Therefore, practical WFS systems employs a 2.5D version [14]: linear loudspeaker arrays are used to synthesize the field of 3D sources in the ear plane of the listeners, as depicted in Fig. 1. The reduction in dimensionality neglects the correct synthesis of elevated wave components for which the human hearing mechanism is less sensitive. However, this restriction also holds for most of the currently applied surround systems.

The main advantage of these systems is that the acoustic scene has no sweet spot since it recreates the wavefront of the virtual sources. When listeners move inside the listening area, the spatial sound sensation changes also in a realistic way according to its relative position to the virtual source. In addition to virtual sources behind the loudspeaker array, it is also possible to synthesize sources inside the listening area, the latter known as focused sources. However, the recreation of a true natural wave field can only be fulfilled with certain restrictions. Huygens's principle needs to be discretized in practice, which means that an infinite continuous secondary source distribution is replaced by a number of finite arrays of equidistant discrete loudspeakers. The lack of continuity leads to a maximum usable frequency, known as spatial aliasing frequency, whereas the finiteness of the array causes some truncation effects. For example, a typical loudspeaker distance in the technical literature is 18–20 cm, which gives an aliasing frequency of about 1 kHz. A detailed description of these drawbacks within a listening room can be found in [15–17].

This setup is present in the loudspeaker arrays for WFS reproduction that have been designed during the past years. Basically, according to the transduction mechanism, two types of loudspeaker arrays have been developed. Firstly, arrays based on the dynamic transducer, which have extensively been described in the technical literature and have been generalized for an arbitrary loudspeaker directivity in [18]. Secondly, arrays comprising several exciters onto a MAP substituting individual loudspeakers. In the following sections, the distributed mode operation and the development to produce MAPs will be explained in detail.

2.2. Multiactuator Panels

MAPs are an extension of DMLs for multichannel audio applications, in particular to WFS. The DML is a loudspeaker based on the vibration of a flat, lightweight panel, driven at particular points by means of electromechanical transducers called exciters or actuators. In a DML panel, the acoustic output is the sum of a large number of uncorrelated modes. For a dense and even frequency mode distribution, the excitation points are chosen in such a way so as to excite as many modes as possible [1,2].

Encouraged by the positive results on sound localization, the applicability of single-exciter DMLs for WFS reproduction was tested for the first time in [5], reporting that individual panels reconstructed the wave field correctly. In [10], it was proposed to extend the DML technology to a panel with multiple exciters, each driven with a different signal, which was later named MAP. In a MAP, a set of exciters, evenly distributed on a line, are attached to a single vibrating surface. Each exciter, that must act as a secondary source of the WFS algorithm, receives an independent signal to create a multichannel system. To avoid back to front cancellation, which results in a smoother frequency response and better low frequency reproduction, and for practical support, the panel is mounted on a housing with absorbing material [19].

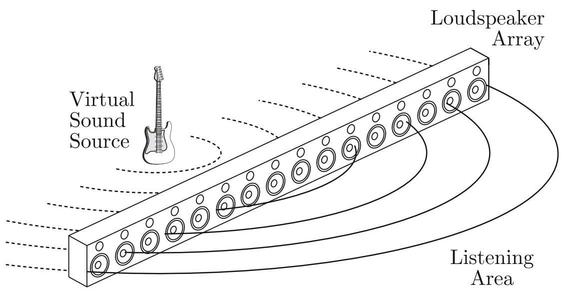

Fig. 1. In Wave Field Synthesis, a linear loudspeaker array synthesizes a virtual sound source inside the listening area.

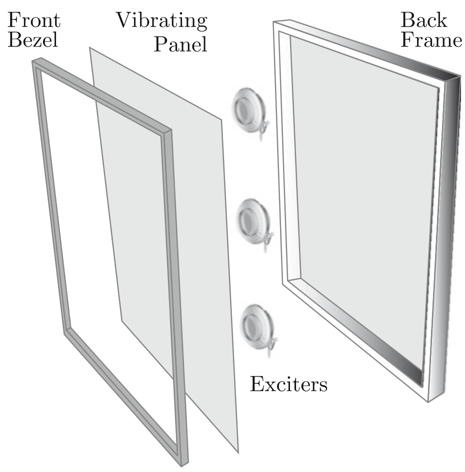

Fig. 2. Graphical representation of a three-exciter multiactuator panel. Electric wiring is omitted for simplicity.

A graphical representation of the comprising parts of a MAP is given in Fig. 2.

MAPs present several advantages for a WFS listening area. Due to their diffuse radiation and omnidirectionality, the individual components merge correctly into the desired wavefront for a wider area. Also, being planar transducers, the pressure level decay with distance is less pronounced in comparison with typical piston-like loudspeakers. However, the modal nature of the panel creates an irregularly frequency response, with peaks and dips, that need to be equalized by appropriate filtering [12,20].

However, the main drawback of MAPs is their poor low frequency response, which can be explain as follows:

1. Lack of low frequency modes to create significant radiation for practical panel dimensions.
   For a smooth frequency response, it is of crucial importance to have a high modal density in the entire audio frequency range. However, since the human perception of frequency is logarithmic, a constant modal density leads to a lack of modes per frequency band at low frequencies, which may be spaced in an uneven manner, affecting the required smoothness [21].
2. Worse low frequency performance than single-excited panels since excitation points are not optimal.
   In a single-exciter panel, the optimal positions are chosen to excite the maximum number of vibration modes. This results in an even distribution of modes, and a high modal density [21]. For WFS operation, such optimal points cannot be observed since they are defined by the secondary sources of the WFS algorithm. Away from the optimal points, there is no guarantee of a proper distribution of modes, being specially noticeable in the low frequency region.

As stated in the introduction, to diminish the above problem the use of these panels for WFS applications is generally supported by subwoofer dynamic loudspeakers that handle the low frequency end of the WFS program [22]. As an alternative, this paper addresses a combination of physical and psycho-acoustical techniques in order to extend the low frequency region of a MAP with no external aids. The physical approach consists in applying a dynamic electrical boost to certain WFS exciters or, when possible, to a separate dedicated low frequency exciter. The psychoacoustical approach takes advantage of the missing fundamental phenomenon, which will be explained in the next section.

3. Bass enhancement for MAPs

As described in the preceding section, the two strategies intended for bass enhancement are discussed in the following sections.

3.1. Physical bass enhancement

For single- and multi-exciter DML panels, the exciter position is selected so that a broad range of lower order modes in the plate can be promoted [23]. By controlling plate material properties, the modal density at low frequencies can be optimized. As in WFS the drive points are determined, there are free areas in the panel in which additional exciters can be applied to excite the lower order modes.

According to this, electrical equalization, analog or digital, could be used in the added exciters to extend and emphasize the lower frequencies in a controlled way. This equalization with Shelving filters or Peak Filters [24], typical in audio equalizers, could improve the bass response when used with a reasonable gain value and at low sound pressure levels. It is possible to extend, with the added exciter, the frequency response below the natural cut-off frequency f_c of the panel, which is given by

$$f_c \approx 2.5 f_0 = 2.5 \frac{\pi}{l_x l_y} \sqrt{\frac{D}{\mu}}, \tag{1}$$

for isotropic plates [25], where l_x and l_y are the width and height of the plate, D is the bending rigidity and μ is the areal density. The problem arises when the desired bass enhancement is much lower than f_c or the degree of demanded enhancement is considerable. Below f_c, the efficiency of the panel decreases fast [26]. The natural high-pass behavior of several DMLs with different exciter positions shows slopes larger than 24 dB per octave when the number of modes is very small at this low frequency band [12]. So, high gain Shelving or Peak filters will be needed to compensate for the poor bass response, with larger filter gains as the frequency of enhancement is lowered. This has two clear drawbacks: first, it increases the electrical power dissipated in the voice coil of the exciters and even the amplified voltage signal could get distorted; and second, the demanded mechanical elongation will be over its mechanical maximum elongation, and it will create audible distortion. The combination of these two drawbacks could even damage the exciters.

As described in Section 4, several tests have been carried out equalizing the low frequencies of different MAPs configurations, experimenting the commented drawbacks when high degree of enhancement was demanded at low listening levels. With higher listening levels, severe power handling and elongation problems appeared, achieving a high distortion and annoying sound reproduction. To avoid that, a frequency and level controlled dynamic behavior of this equalizer is preferred. This is called a dynamic equalizer, which achieves larger gains when the level of the signal is lower, and lower equalization gains when the level of the signal arises. This will maintain the electrical signal applied to the exciters within its allowed power and elongation ranges, which is actually quite small (typical ±0.2–0.5 mm for exciters with voice coils between 13 and 25 mm).

3.2. Psychoacoustic bass enhancement

The other possibility for improving the "perceived" bass reproduction is the use of the psychoacoustic effect of the *missing fundamental*. It has been lately commercialized and used with success mainly on the market for small and medium sized electro-dynamic loudspeakers. Its principles could also be used for MAPs in combination with the physical strategies of the previous section.

An excellent reference that collects history and implementation issues about bandwidth extension is [13], where a more in depth description of the principle could be found. Human brain is trained to perceive patterns, what is called pattern recognition. Normally, any kind of musical instrument or melodic sound with periodicity is composed by a series of frequencies harmonically related, being the perceived pitch, the fundamental frequency f_0, that normally is the partial with higher amplitude. This can be seen on Fig. 3, where the fundamental frequency is plotted in gray.

The frequency difference between each partial is f_0, so they are harmonically related. When f_0 is attenuated with the use of a high-pass filter, or even it is eliminated, the spectrum content is shown in black at Fig. 3. Now the largest partial is $2f_0$, but the decoded pitch will be again f_0 because the distance between the partials $2f_0$, $3f_0$, $4f_0$, etc, is still f_0. The decoded pitch is not $2f_0$, because the presence of the non harmonically related partials $3f_0$ and $5f_0$, forces the brain to detect the pattern with a frequency difference f_0 between the partials, so the decoded pitch is again f_0. This effect is also called the difference tone effect.

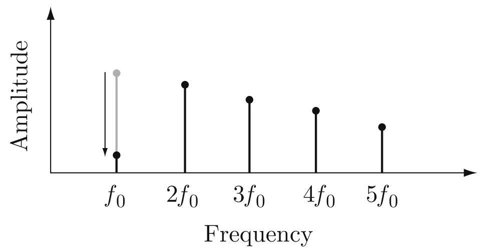

Fig. 3. Missing fundamental effect on a harmonically related spectrum.

In this paper, this principle is used to attenuate the lower frequencies that will not be reproduced by the MAPs, and substitute it with an excited content that will be almost one octave up and will be reproduced by the added exciters in MAPs without problems. This psycho-acoustic effect will facilitate the detection of the original pitch. At the same time, most of the low frequency content will be attenuated. This will decrease the power dissipated on the exciters and their demanded elongation. Generally, this is the frequency band, where most of the energy is concentrated in music.

One important issue is that the excitation of the harmonics or partials must be done through a non-linear treatment of lower frequencies. The only way to create the harmonics is by means of distorting, modulating, or generating and then adding the harmonics to the original signal once detected its pitch, being all these processes non-linear.

The first devices that implemented this effect in an electronic circuit, where all analog, starting with valves and transformers [27,28], and lately with operational amplifiers, active filters, diodes and transistors, with rectifiers and integrators, etc. [29,30]. The actual implementations have moved to the digital domain executed on low-cost Digital Signal Processors (DSPs) [31] or even software implementations to be included in host audio applications as plug-in modules. There are several solutions on the market that could be found on small Hi-Fi systems or multimedia loudspeakers, but all of them focused and optimized for electro-dynamic loudspeakers. A review of some of them could be found at [13].

In a general way, the common stereo block diagram implementation of the devices that improve the low frequency perception based on the missing fundamental effect is the one displayed at Fig. 4 [13].

The left and right channels, L_{in} and R_{in} respectively, are just filtered by the high-pass (HP) blocks with a cut-off frequency f_{ch}. High-pass filters of order between 2 and 4 are usually employed, being third order Butterworth B3 the most common one since the cross-over stage achieves an all-pass system. This attenuates the low frequency content below f_{ch}. The filtered signals are routed directly to an adder at the outputs. The selection of f_{ch} will be MAP dependent and should not be more than one octave lower than the natural cut-off f_c of the MAP in order to get a noticeable bass enhancement effect.

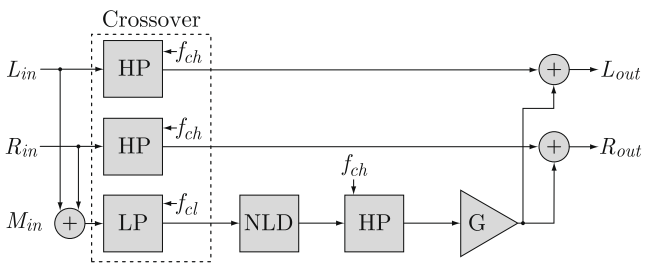

Fig. 4. Block diagram implementation to improve the low frequency perception based on the missing fundamental effect.

A mono input M_{in} is obtained and is low-pass filtered at f_{cl} with the low-pass (LP) block. Normally, the low frequency content of music material is mono because there is no effective localization at low frequencies. The values of f_{cl} and f_{ch} are usually the same to split the high and low frequency bands. As seen, a dual band processing scheme is implemented. The combination of the HP and LP filters form the classical cross-over setup. Then, the low-pass filtered mono signal gets into the Non-Linear-Device (NLD) block that is the one responsible for creating the harmonic related excitation. This is the key block, and the way it will create the harmonics will have a great impact on the quality and perception of the bass enhancement. As mentioned above, there are simple NLDs working as rectifiers or rectifiers with integrators and reset control [30]. An interesting implementation is [31], where a feedback loop creates the successive multiplications of the signal to create the harmonics very efficiently. Other more complex solutions include a pitch detection and posterior synchronous modulation to create the harmonics [32]. But basically, all of them do the same: to create harmonics from the low frequency content.

In this work, a 6th order polynomial has been employed in order to generate up to the 6th harmonic. Several psycho-acoustical tests [33,34] have demonstrated that the harmonics involved in the pitch detection reach up the 6th harmonic, what is called the dominance region, so there is no reason to go further increasing the computational cost on its implementations. The polynomial series expansion of 6th order to distort the signal is given by

$$y[n] = b_0 + b_1x[n] + b_2x^2[n] + \cdots + b_6x^6[n], \tag{2}$$

where the coefficients b_i of the polynomial have been adjusted performing a polynomial fit to different curves like arctan or exponential ones. After some subjective testing and comparisons with a bass enhancement commercial implementation based on the missing fundamental effect, a polynomial approximation to an exponential curve was used [35]:

$$y(x) = e^{\alpha(x-1)}, \tag{3}$$

where α is a gain factor that controls the amount of added distortion and the input signal x is assumed to be normalized.

Then, the excited low frequency band is high-pass filtered at f_{ch} with the next HP block. This filter will attenuate the original lower frequencies and maintain unaltered the harmonics with the effect of the missing fundamental, which will be reproduced by the MAP without problems. A final posterior gain (G) is applied to control the quantity of effect added to the output with the high-pass filtered signal from the left and right channels.

After doing some test with real MAP configurations, depending on the NLD implementation and its parameters, some kind of control with the enhanced signal was needed to limit the amount of electrical power and the demanded mechanical elongation to the exciters. For that purpose, a Dynamic Controller (Dyn) with an expander, compressor and a limiter was included after the NLD block to avoid excessive effect and the posterior distortions and system overloading. To consider the MAP multichannel input in WFS applications, instead of having only into account the left and right channels, the block diagram of Fig. 4 has been modified and is presented in Fig. 5. The n input channels are filtered by the HP blocks with the same f_{ch} and then are routed to an adder. As in the stereo block diagram, the summation of all input signals is scaled and low-pass filtered at f_{cl} with the LP block. Since this addition is performed before the LP filter, the phase lags between channels as a consequence of the WFS algorithm are insignificant and therefore, the sum is coherent, which poses no problem.

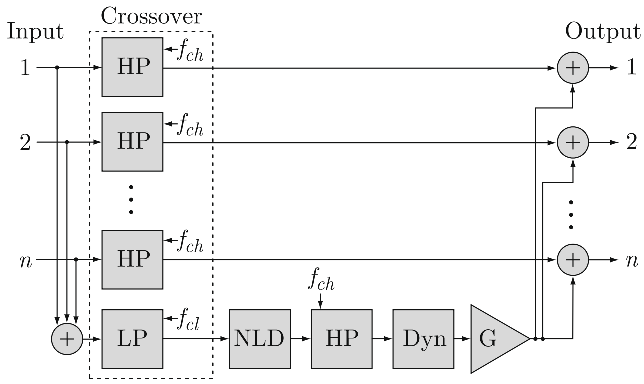

Fig. 5. Adaptation of the block diagram of Fig. 4 to multichannel application.

A dual band processing scheme with harmonic generation and dynamic control is implemented to create the psychoacoustic bass enhancement. If a variable gain is introduced in the creation of the mono signal, an electrical equalization of the frequencies between the LP and HP filters is obtained achieving a Peak filter in the frequency band between them. The difference with an equalization only approach is that now the lower frequencies are eliminated and replaced by their superior harmonics that the MAP will be able to reproduce, letting the brain decode the original low frequency pitch. The use of the posterior Dyn controller allows such a dynamic equalization. With this scheme, lower original bass frequency content and lower energy and elongation is introduced to the exciters, achieving at the same time a better perceived bass response.

In the case that only the enhanced bass response is desired to be reproduced by the added exciters, and work with it like a conventional low frequency channel, the high-pass filtered signals from left and right could be eliminated from the outputs.

3.3. Bass enhancement solution

The proposed solution scheme for bass enhancement in a WFS application using MAPs is shown in Fig. 6.

To test the bass enhancement solution, a digital signal processing implementation based on building blocks has been developed

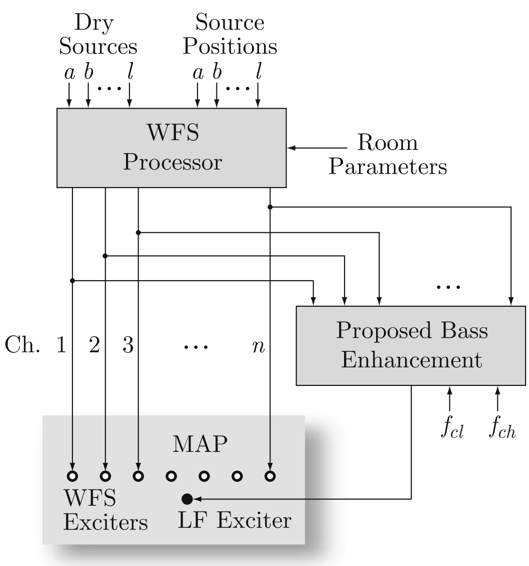

Fig. 6. Bass enhancement processing by means of combined physical and psychoacoustical approaches.

using SigmaDSP audio signal processors from Analog Devices Inc. [36], with real-time control on the most important parameters (f_{cl}, f_{ch}, NLD harmonic creation, Dyn controls, and G) in order to achieve the best sounding solution for a given MAP loudspeaker. A proprietary WFS Processor was used to generate the required multichannel signals. Fig. 7 shows the measured behavior of the crossover filters with f_{cl} and f_{ch} at 100 Hz, using third order Butterworth filters.

Fig. 8 shows the behavior of the NLD with a 50 Hz input signal after the HP filter. The NLD is configured to create a high degree of harmonics. It is noticeable that up to the sixth harmonic has been created, as commented before. In normal operation, the Total Harmonic Distortion (THD) levels could rise up to 50% when the input fundamental frequency is lower than f_{ch}, since the level of harmonics could be larger than that of the fundamental, as observed in Fig. 8.

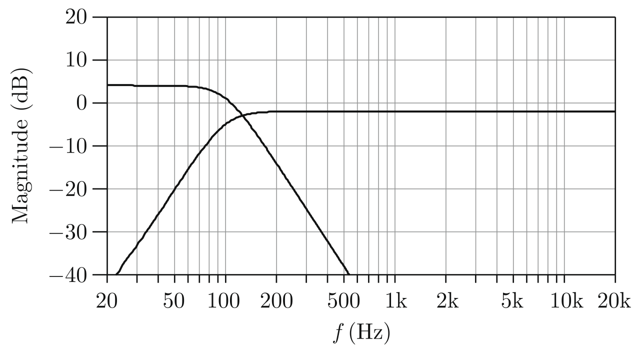

Fig. 7. Measured frequency response of the crossover filters with cut-off frequencies f_{cl} and f_{ch} of 100 Hz (HP and LP blocks of Fig. 5).

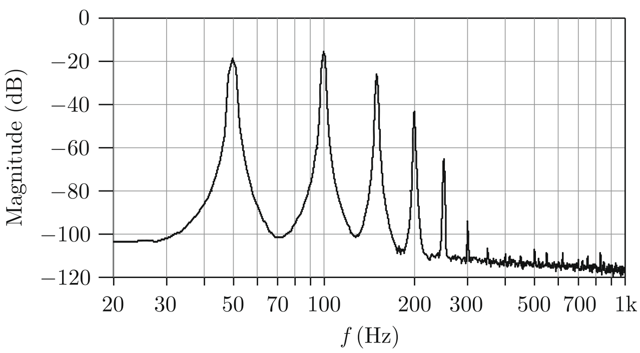

Fig. 8. Measured behavior of the Non-Linear-Device creating the harmonic related excitation with a 50 Hz input signal (NLD block of Fig. 5).

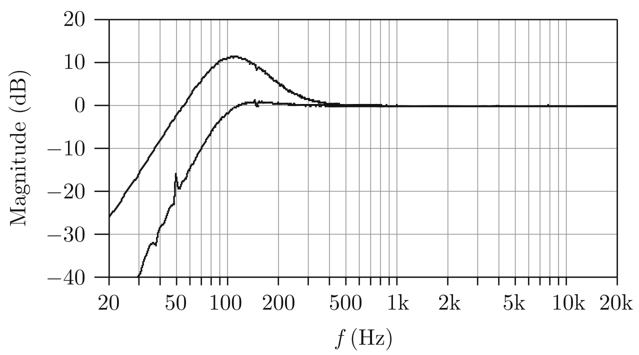

Fig. 9. Measured frequency response of the complete algorithm with two gain values of 0 and 12 dB (G block of Fig. 5).

Finally, Fig. 9 displays the measured frequency response of the complete algorithm with two G values. It is noticeable how the lower frequencies are eliminated and that the electrical boost is applied in the frequency range, where the panel is able to work properly. The Dyn controller allows to apply dynamically this electronic boosting based on the level of the input signal.

4. Experimental setup

The proposed bass enhancement process is validated for two MAP prototypes having different aspect ratios. The first prototype is a small MAP containing three exciters which show an aspect ratio of approximately 3:2. This is a usual aspect ratio in general applications, when the panel is integrated in walls or when it is also used as a projection screen. Fig. 10a illustrates the geometry of the introduced prototype, which will be denoted as MAP3 in the following. The second prototype is a MAP that is extended in the horizontal dimension and restricted in the vertical dimension. This high aspect ratio panel or HAR MAP in the following, contains five exciters in a geometry similar to that of the MAP3. In general, high aspect ratio MAPs are typically found in applications such as TV, where there is limited space at the side of TV screen to fit speakers. Geometric details of the prototype can be found in Fig. 10b.

The distribution of exciters in MAPs must follow the evenly spaced positions of the WFS secondary sources. For a MAP to substitute a loudspeaker array for WFS, each exciter must act as an individual source. According to [10], the main contribution of energy comes from the region around the exciter position, with a diameter of a few centimeters due to the high internal damping of the panel. As a result, the optimal excitation points cannot always be satisfied in WFS configuration.

The criterion used to determine such optimal points is related to the panel node lines: the excitation of a mode is not possible if the excitation point is located on a corresponding node line. Therefore, to excite as many low frequency modes as possible, the node lines of the low frequency modes must be determined. For a set of preferred aspect ratios, such as that of MAP3, the optimal excitation points are accurately specified in [37], corresponding to four places in the center area of the panel. None of the MAP3 exciters is located in these optimal points, so there are four candidate driving points to excite the lowest frequency modes that can be generated for this panel dimensions, the coordinates of which (x, y) in centimeters are $(23.1, 19.4)$, $(30, 19.4)$, $(24, 15)$ and $(30.86, 15)$. The last three positions are very close to the central WFS exciter, so the extra low frequency exciter will be located in the first optimal drive point, as depicted in dashed line in Fig. 10a. This exciter will only receive the processed low frequency signals, as discussed in Section 3.3.

For the HAR MAP, however, there are no precalculated optimal points since the particular aspect ratio is not considered in [37]. Notwithstanding, the criterion of placing the excitation points away of node lines still applies, so a simulation was made to determine the distribution of nodal lines over the panel. For that purpose, a modal network solver called “Pansys” [38], which is specially intended for this type of flat panel loudspeakers, was used with the following data input. The panel width, height and thickness were 90, 20, and 0.5 cm respectively. Its sandwich structure was made of an impregnated paper honeycomb core (cell size 4.8 mm) in a translucent polyester skin, whose material properties are: bending rigidities D_1 and D_2 of 12.88 and 7.46 Nm, respectively, and areal density μ of 0.602 kg/m^2. Finally, a free boundary with suspension was selected as the edge boundary condition. The suspension was a Low Density Polyethylene foam strip of rectangular section 0.5×1 cm running along the edge of the panel. For a detailed discussion on edge boundaries for MAPs, refer to [39].

The simulation results showed some areas, where to put the LF exciters optimally but these were too close to the regular WFS exciters. This avoids placing additional exciters for this particular case and use, instead, some of the existing WFS exciters. For that purpose, the simulation showed a suboptimal, off-center area, in which exciters 1/2 or 4/5 were located. Among the pairs, the exciter number 2 was chosen due to the slight proximity to an optimal point. However, as it will be confirmed in frequency measurements, the first exciter would also behave properly. This case study is interesting for exploring the method in aspect ratios apart from the standard, in which additional LF exciters can be placed.

As shown in Fig. 10, both prototypes share the same exciter spacing of 18 cm, which sets a maximum usable frequency without spatial aliasing of 1 kHz. Moreover, to facilitate the distribution of several units in landscape orientation, the distance between the first/last exciter to the edge is 9 cm, half the exciter separation.

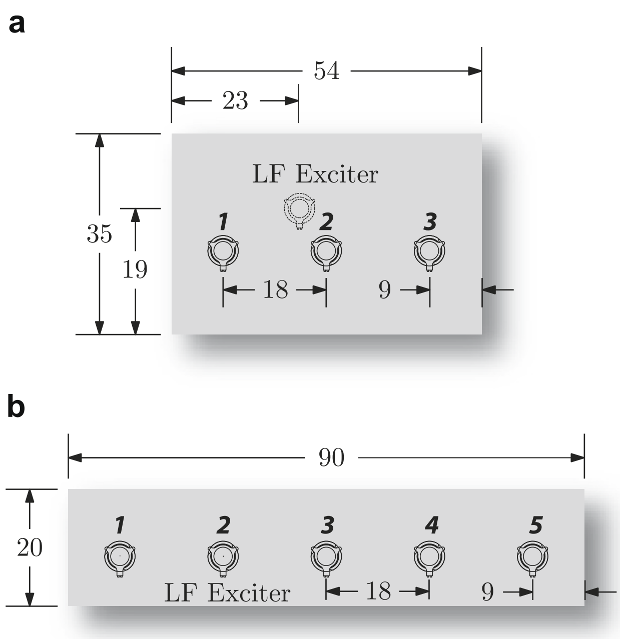

Fig. 10. Geometric details of MAP prototypes tested for bass enhancement. (a) MAP3 prototype. (b) HAR MAP prototype. Dimensions in centimeters. Exciter numbering is depicted in italic boldface.

5. Results and discussion

The MAP prototypes will receive the bass enhancement processing and a subjective assessment will also be carried out to ensure a reasonable limit in the processing. All measurements have been obtained using a 1/2” electrostatic microphone placed 1 m from the MAPs, and using Maximum Length Sequences (MLS) as excitation signals in order to improve the signal-to-noise ratio, and minimize the effect of unwanted reflections [40].

5.1. Bass enhancement for the MAP3 prototype

In order to evaluate the frequency response of the WFS exciters of the MAP3, individual on-axis measurements were carried out. Fig. 11 shows the frequency responses without any equalization for the three exciters. As shown in the figures, the behavior of the exciters positioned in symmetrical places at both ends of the panel is almost identical since the generated modal response is

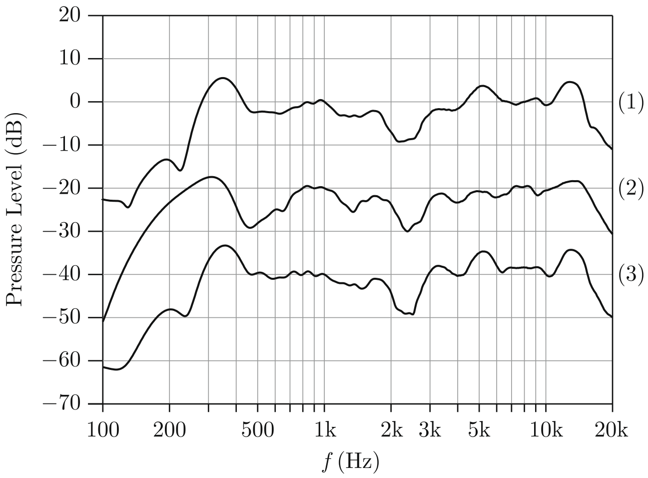

Fig. 11. On-axis frequency response of the MAP3 prototype without any additional processing. (1) Exciter 1. (2) Exciter 2. (3) Exciter 3. Representations are shifted 20 dB in order to maintain clarity.

similar. Then, for both exciters 1 and 3, responses reach 0 dB approximately at 300 Hz and continue increasing to a peak at approximately 340 Hz. With regard to the central exciter, the low frequency response is different from that of the edge exciters since the mode pattern is also different. These irregularities in the frequency response are inherent to the DMLs and are normally compensated for by means of some sort of equalization, like that proposed by the authors in [12] for MAPs.

An additional exciter is placed at the excitation point (23.1, 19.4) cm, so as to generate as many low frequency modes as possible. This exciter will receive the processed signal to enhance the bass frequencies. Fig. 12 shows in solid line the on-axis frequency response of the LF exciter, which has a certain resemblance to the response of the WFS central exciter, depicted in Fig. 11. The figure also depicts in dashed, dash-dotted and dotted lines the resultant responses when minimum, medium and maximum harmonic levels are applied, respectively. The maximum enhancement matches the maximum physical elongation of the LF exciter. Note that the effect on the perceived bass response due to the non-linear process is present, but not observed directly in the frequency response graphs.

In this particular panel loudspeaker, the values of f_{ch} and f_{cl} were 130 Hz and three degrees of harmonic addition and dynamic bass-boost were employed. The Dynamic Controller was configured as a compressor and a limiter to limit the amount of added distortion and low frequency level up to the maximum physical elongation of the exciters, with attack and release time constants of 10 and 100 ms respectively. These values were tuned by ear during previous listening tests.

5.2. Bass enhancement for the HAR MAP prototype

For observing the frequency response of every exciter on the panel, including the WFS exciter that will handle the bass enhanced signal, individual on-axis measurements were carried out. Results are presented in Fig. 13.

In a MAP with an aspect ratio such as that of the MAP3 prototype, the low frequency response is due to a mixture of modes both in the horizontal and vertical dimensions. As the aspect ratio increases, the modes in the longer dimension tend to dominate and there is no sufficient modal density to get a smooth response. This effect can be seen in the reduced bandwidth for the responses of all exciters in Fig. 13.

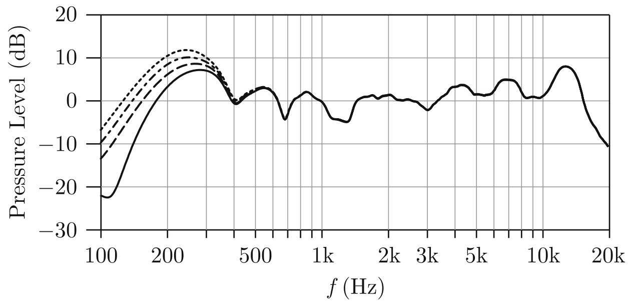

Fig. 12. On-axis frequency response of the additional LF exciter for different enhancement levels.

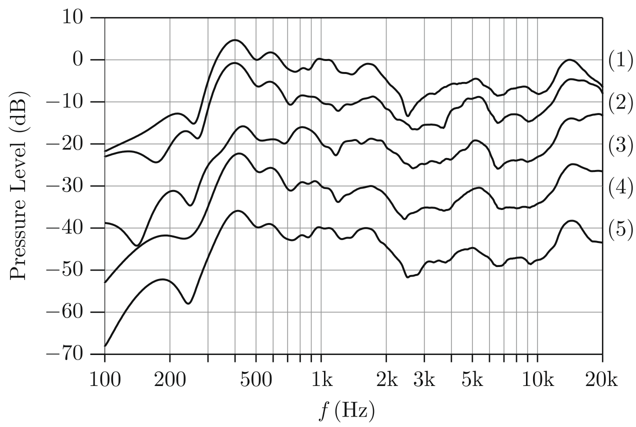

Fig. 13. On-axis frequency response of the HAR MAP prototype without any additional processing. (1) Exciter 1. (2) Exciter 2. (3) Exciter 3. (4) Exciter 4. (5) Exciter 5. Representations are shifted 10 dB in order to maintain clarity.

For the HAR MAP, the chosen WFS exciter to reproduce the low frequency signals was exciter 2, which is verified here in Fig. 13 by frequency response measurements. Due to the symmetrical position of exciter pairs 1–5 and 2–4, the measured frequency responses are similar, as expected.

In Fig. 14, the unprocessed response is shown in solid line together with the bass enhanced response in dashed line. The crossover filters of the algorithm were set to $f_{cl} = f_{ch} = 160$ Hz and, since the exciters are the same as in the MAP3, the Dynamic Controller was configured with the same set of parameters. After applying the processing, the bass enhancement is remarkable: from a cut-off frequency of about 300 Hz, an extension of approximately 100 Hz is achieved.

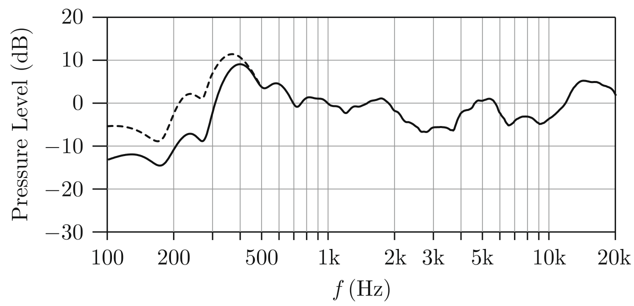

Fig. 14. On-axis frequency response of the exciter 2 for different enhancement levels. Solid line: unprocessed response. Dashed line: maximum harmonic addition.

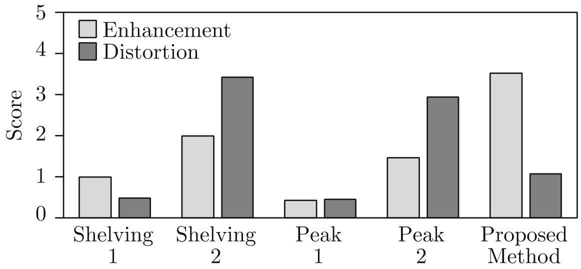

Fig. 15. Results of the comparative subjective test for three enhancement methods applied to the MAP3 prototype.

5.3. Comparative subjective assessment

In this section, a comparative subjective assessment is carried out on the MAP3 to study if the perceived quality of the bass enhancement for both shelving, peak and the proposed method introduces annoying distortions due to excessive elongation of the exciters' voice coils or the panel. For that purpose, 10 listeners with normal hearing formed a jury that participated in a test to observe which method gives the deepest and more natural bass sensation without distortion. In this test, listeners must give an score from one to five to the following methods: (a) Shelving Level 1, (b) Shelving Level 2, (c) Peak Level 1, (d) Peak Level 2, and (e) Proposed bass enhancement. For comparison purposes and as a reference of proper bass reproduction, the tests always started with a sample without bass enhancement, followed by the same sample, reproduced with a 8" bass reflex subwoofer, which is set as the maximum bass quality (score 5).

As shown in the results of Fig. 15, a shelving or peak equalization is not proper for the purpose of the bass enhancement since a lot of distortion is introduced without acquiring a desirable degree of bass enhancement. On the one hand, for shelving equalization with level 1, a low degree of distortion is perceived while the bass sensation is also very low. When applying higher level, the enhancement increases accordingly but with an unacceptable level of distortion. As an example, for a 200 Hz cut-off frequency, distortion starts to be annoying when a gain of 15 dB is applied. On the other hand, peak equalization with level 1 shows a reasonable degree of distortion but the enhancement is virtually null, whereas for level 2, distortion is quite high. Listeners reported that an increase in the enhancement level gives rise to a pipe sound. In general, to acquire the same degree of bass sensation, the peak equalization must receive higher level than with shelving, which causes a distorted signal. Finally, the proposed method depicts a very well controlled distortion level for a moderate to high degree of bass sensation, as described by the jury. If the enhancement level increases, distortion will rise dramatically, which sets a gain limit of 15 dB for this particular prototype. Although the maximum score of the subwoofer sensation is not achieved (score 5), there is a remarkable increase in the perceived bass sensation for a loudspeaker which has a natural poor low frequency response and hence, better audio quality is perceived.

6. Conclusion

MAPs are multichannel flat panel loudspeakers whose main drawback is the inherent poor low frequency response, which is normally solved by placing a supporting subwoofer. In this paper, an extension of the useful bandwidth in the low frequency band of MAPs has been proposed and validated by means of a combination of a dynamic filtering processing and taking advantage of the psychoacoustic phenomena of the missing fundamental. Hence, the perceived low frequency enhancement is the result of two processes. First, an electrical equalization aiming at increasing the exciter-panel elongation to a reasonable extent in the frequency range were the panel is able to work properly. Second, a psychoacoustic effect that allows human brain to perceive an octave below the natural MAP low cut-off frequency.

Two MAP configurations have been tested in order to validate the enhancement process in two possible configurations. On the one hand, a rectangular MAP which dimensions allow designers to locate extra exciters that drives the low frequency enhanced signal onto particular points. On the other hand, a wide MAP has also been employed as an example of high aspect ratio panels, present in devices such as flat TV or computer monitors, in which some of the exciters intended to reproduce the WFS signal is used to introduce the processed signal.

The MAP bass enhancement proposed in this paper has been compared to other audio processing in the same MAP prototypes, such as the shelving or peak equalization methods, which have resulted in inadmissible levels of distortion for the same bass frequency loudness perception level.

The combination of two strategies, physical and psychoacoustic, towards a low frequency improvement in MAPs has resulted in a effective extension of the bandwidth in the low end. By carefully selecting the algorithm parameters, it is possible to avoid the use of an external subwoofer in certain applications.

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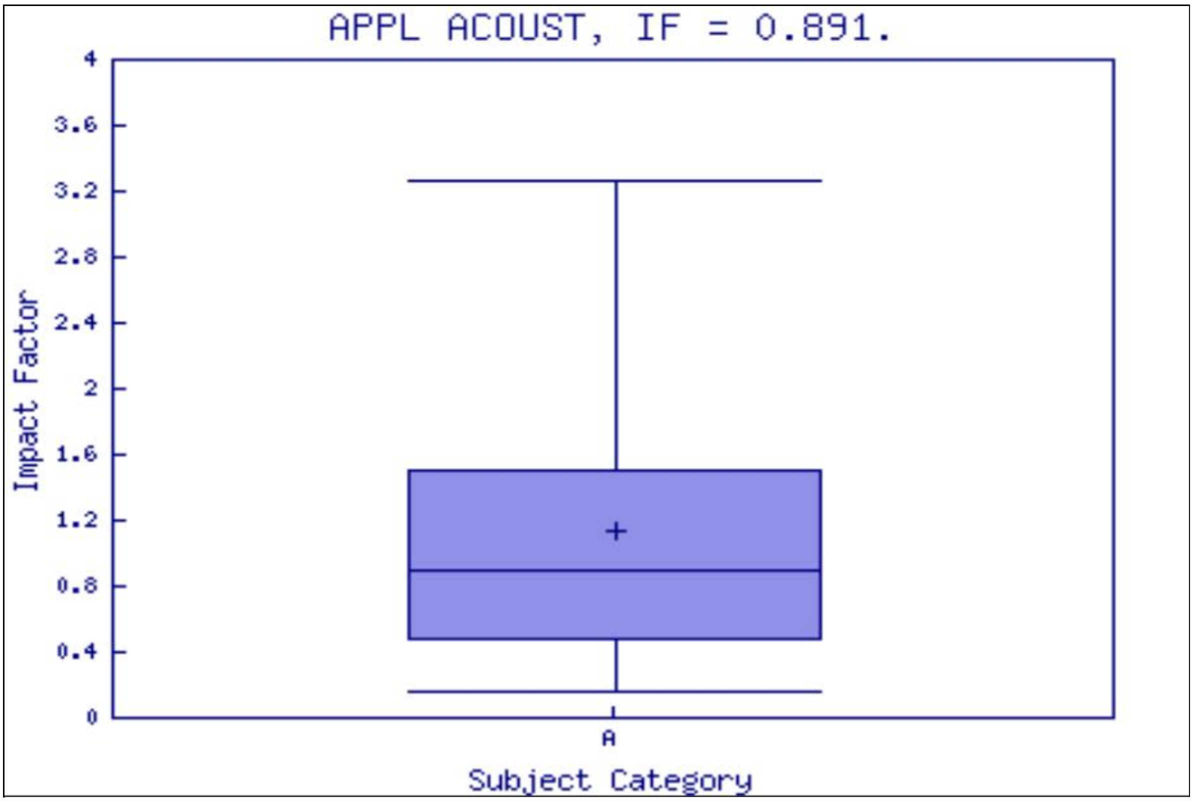

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