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A Novel Approach for Prototype Extraction in a Multipoint Equalization Procedure

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ABSTRACT

Multipoint equalization is a useful procedure used to enlarge the zone to be equalized in sound reproduction systems by measuring the room impulse responses in multiple locations and deriving a prototype function capable to represent the real environment. This paper deals with the introduction of a novel prototype function derived from the combination of quasi-anechoic impulse responses with the impulse responses recorded in the real environment to be equalized. This is motivated by the fact that at mid and high frequencies the timbre perception and localization is dominated by the direct sound, thus, the measurable, but mostly inaudible magnitude deviations due to reflections should not be equalized. Several experiments have been conducted in order to validate the proposed approach, considering a real environment and reporting objective and subjective measurements in comparison with the state of the art.

1. INTRODUCTION

Room response equalization is a technique aimed at improving the objective and subjective quality of sound reproduction systems. The technique has found application in cinema theaters, home theaters, and car HiFi systems. The sound reproduction improvement is obtained by shaping the room transfer function from the loudspeakers system to listener with a suitably designed equalizer. Room response equalizers can be divided into single position and multiple position equalizers. In single position room equalizers, the equalization filter is designed on the basis of a measurement of the room impulse response (IR) in a single location [1, 2, 3, 4, 5, 6]. These equalizers can achieve the room equalization only in a reduced zone around the measurement point with the size of a fraction of the acoustic wavelength. Indeed, the room IR varies significantly with the position. On the other hand, a variety of approaches on multipoint room equalization are documented: these approaches enlarge the equalized zone by measuring the room impulse response in multiple locations [7, 8, 9, 10, 11, 12, 13, 14, 15, 16, 17].

The approach presented in this paper is related to the room response equalization systems of [7, 10, 14, 15] and [4]. In [7], S. Bharitkar and C. Kyriakakis introduced a multiple position room response equalization technique based on fuzzy c-means clustering and frequency warping. Specifically, given a set of room impulse responses measured at different positions, the mentioned technique applies a fuzzy c-means algorithm for clustering them on the basis of their similarity. A prototype impulse response, obtained by combining the cluster centroids, is then used for designing a low order equalization filter by means of Linear Predictive Coding (LPC) analysis. In order to obtain a better fit of the LPC model to the room response in the low frequency region, the measured room responses are frequency warped using a psychoacoustically motivated Bark scale. Although this technique is able to obtain only a magnitude equalization of the room response, it is effective and robust against displacement effects. This approach has been subsequently improved in [10, 14, 15] by performing most of its operations in the frequency domain. The computational complexity is reduced by applying the fuzzy c-means clustering directly to the room amplitude frequency responses in order to obtain a prototype magnitude response. Frequency warping is done by suitably sampling the magnitude responses. The extraction of the minimum-phase component of the IRs is avoided, and the computation of the autocorrelation of the IR of the prototype filter is also simplified by working in the frequency domain. Furthermore, in [15] it was shown that the prototype magnitude response can also be computed by averaging the magnitude responses measured at the different positions. The performance achieved with this simpler prototype extraction method are similar to those obtained with the fuzzy c-means clustering. A further improvement to this method was introduced in [14] by replacing the equalizer design based on LPC analysis with an approach based on frequency deconvolution with regularization [18] to avoid excessive gains at high frequencies.

Starting from the approach of [14], the definition of a novel prototype response for multipoint audio equalization is presented taking into consideration some aspects of the single position equalization technique proposed in [4]. The novel prototype function is derived from the combination of quasi-anechoic IRs, derived from a gated (i.e., truncated) version (up to the first reflection) of the responses, with the IRs recorded in the real environment. In particular, given a set of impulse responses measured in the to-be-equalized zone, their magnitude spectrum is derived considering the quasi-anechoic IR spectrum for frequency greater than a certain transition frequency, and the original (un-gated) IR spectrum below the same transition frequency. This is performed in order to equalize the direct sound only in mid-high frequency range, while performing full equalization in the modal frequency range. In fact, at mid and high frequencies the timbre perception and localization is dominated by the direct sound, and thus, the measurable, but mostly inaudible, magnitude deviations due to reflections should not be equalized [4]. Then, the magnitude spectra are smoothed considering fractional octave bands: this method simulates a well-known property of the auditory system which presents a poorer frequency resolution at higher frequencies. In this way, it is possible to consider a non-uniform resolution, which decreases with increasing frequency, to obtain a less precise equalization at higher frequency resulting in a broader equalized zone. At this point, a representative response of the considered acoustic environment is derived taking into account all smoothed IRs. The prototype frequency response is obtained using an arithmetic mean of the smoothed frequency responses. Then, the inverse model of the prototype function is obtained. The equalization filter is computed by frequency deconvolution with regularization to avoid excessive gains especially at high frequencies.

Given a set of impulse responses measured in a real environment, tests have been conducted in order to evaluate the effectiveness and perceptual usefulness of the proposed approach in comparison with similar state of the art approaches. In particular, the approach has been compared with those of [19] and

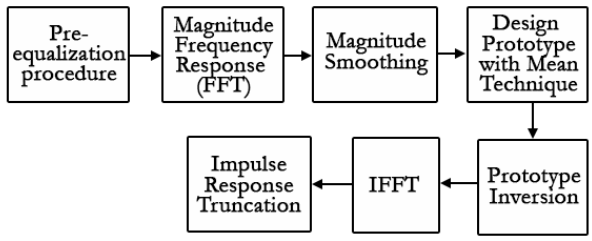

Fig. 1: Room response equalization diagram.

[14] taking into consideration objective and subjective measurements. The results of a preliminary listening test session are reported giving a subjective point view of the effects produced by the multipoint equalization procedure.

The paper is organized as follows. Section 2 provides a description of the proposed algorithm, taking into consideration each mathematical steps of the approach. Section 3 discusses the results in terms of objective and subjective measurements considering a real environment, in order to illustrate the performance, the robustness, and the perceived subjective quality of the proposed room response equalization procedure. Finally, Section 4 contains some concluding remarks and future works.

2. ALGORITHM DESCRIPTION

The proposed approach is based on a multipoint techniques [15] capable to enlarge the listening sweet spot. It takes into consideration a quasi-anechoic approach [4] that is capable to produce also a general equalization of the used loudspeakers. The novel prototype function is derived from the combination of quasi-anechoic IRs, derived from a gated version (up to the first reflection) of the responses, with the IRs recorded in the real environment.

Figure 1 shows each step of the equalization approach. In particular, given a set of impulse responses measured in the to-be-equalized zone, a pre-equalization procedure is applied considering the quasi-anechoic IR spectrum for frequency greater than the transition frequency, and the original (ungated) IR spectrum below the transition frequency. It is performed applying the following equation to each room impulse response (RIR), in the frequency domain:

$$H(k) = H_i(k) \cdot w_{\mathrm{lf}}(k) + H_f(k) \cdot w_{\mathrm{hf}}(k), \qquad (1)$$

where $H_i(k)$ is the frequency response of original RIR, $H_f(k)$ is the frequency response of gated RIR, $w_{\mathrm{lf}}(k)$ and $w_{\mathrm{hf}}(k)$ are half Hann windows used for selecting the low-pass and high-pass frequency bands, respectively. The linear combination in Equation (1) is used to equalize the direct sound only in mid-high frequency range, while applying full equalization in the modal frequency range. Figure 2 shows the behaviour of a measured impulse response and of the impulse response obtained following the proposed procedure in comparison with the third octave complex smoothing. A transition frequency of 700Hz has been considered with about 1 ms half Hann window.

Then, starting from the magnitude spectra of the IRs, a smoothing operation is applied: this method simulates a well-known property of the auditory system which presents a poorer frequency resolution at higher frequencies. The basic equation for performing non uniform frequency magnitude spectrum smoothing of a frequency response $H(k)$ is given by [20]

$$H_{sm}(k) = \sum_{i=0}^{K-1} W_{sm}\left(M(k), i\right) \left| H\left((k-i) \bmod K\right) \right|, \qquad (2)$$

where K is the total number of frequency bins, $M(k)$ is the half-window length, which is a monotonically increasing function of the frequency index k, and W_{sm} is a zero-phase window function, defined as

$$W_{sm}(m,i) = \begin{cases} \frac{b-(b-1)\cos(\pi i/m)}{2b(m+1)-1}, \\ \qquad i = 0, \ldots, m \\ \frac{b-(b-1)\cos(\pi (K-i)/m)}{2b(m+1)-1}, \\ \qquad i = K-m, \ldots, K-1 \\ 0 \quad \text{otherwise} \end{cases} \qquad (3)$$

where $b = 0.5$ and m is the smoothing index, i.e., the half-window length. In this way, it is possible to consider a non-uniform resolution, which decreases with increasing frequency, to obtain a less precise equalization at higher frequencies resulting in a broader equalized zone.

At this point, a representative response of the considered acoustic environment is derived taking into

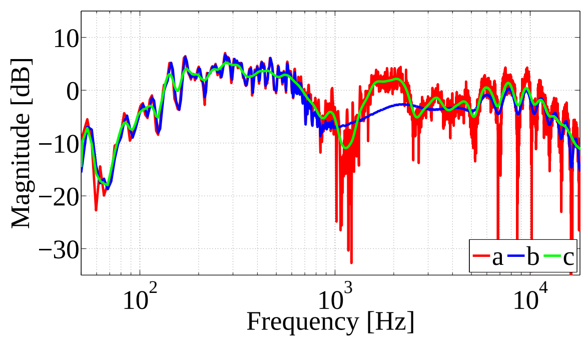

Fig. 2: Comparison between (a) the measured IR and (b) the pre-equalized IR and (c) one third octave smoothed IR.

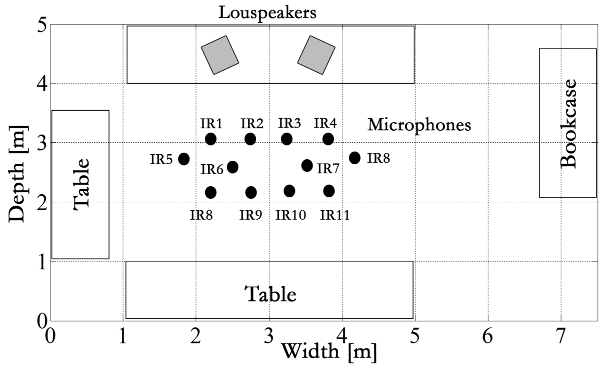

Fig. 3: Microphones and loudspeakers positions for the experimental results.

account all smoothed IRs. The prototype frequency response is obtained using an arithmetic mean of the zero-phase smoothed frequency responses, as follows

$$H_p(k) = \frac{1}{M}\sum_{j=1}^{M} H_{sm_j}(k) \tag{4}$$

with $k = 0, \cdots, K-1$ and H_{sm_j} the smoothed frequency response at the j-th measurement position.

Some tests have been performed considering the Fuzzy c-means approach [10], however the same results in terms of final prototype function have been obtained considering the mean approach. This could imply that the pre-equalization method is already capable of extracting the main characteristics of the analyzed environment.

Then, the inverse model of the prototype function is obtained using a frequency deconvolution with regularization [18] that is capable to avoid excessive gains, especially at high frequencies. Since the prototype is zero-phase, it is applied as follows:

$$H_{\mathrm{inv}}(k) = \frac{H_p(k)}{|H_p(k)|^2 + \beta} \tag{5}$$

where β is the regularization factor and $k = 0, \cdots, K-1$. For the experimental results, a regularization factor with value 0.00001 is considered. The computational complexity of the fast deconvolution method is essentially that of the inverse FFT which is an $O(K \log K)$ algorithm [18]. The corresponding time domain filter is truncated by an appropriate window function with a length L, in order to have a more compact representation. Its length can be limited due to the beneficial effect of the pre-equalization procedure and thus reducing the delay introduced by the filtering operation.

3. EXPERIMENTAL RESULTS

Several tests have been carried out in order to evaluate the effectiveness of the presented algorithm taking into account a real environment, and also providing comparison with the competing techniques in terms of objective and subjective measures. Loudspeakers and microphones positions are shown in Figure 3 together with room size: the distance between the microphones in the positions IR1, IR2, IR3 and IR4 has been set to 17 cm, taking into consideration the median distance between human ears, since they have been used to calculate the equalizer responses. Considering the other positions (from IR5 to IR12), they have been considered in order to test the validity of the proposed approach in positions different from those considered to derive the equalizer. Measurements have been performed using a professional ASIO sound card and microphones with an omnidirectional response. The loudspeakers were Genelec 6010A with free field frequency response 74 Hz - 18 kHz. A personal computer running NU-Tech platform has been used to manage all I/Os [21]. The impulse responses have been derived using a logarithmic sweep signal excitation [14] at 48 kHz sampling frequency.

3.1. Objective Analysis

First of all, the performance of the equalization has

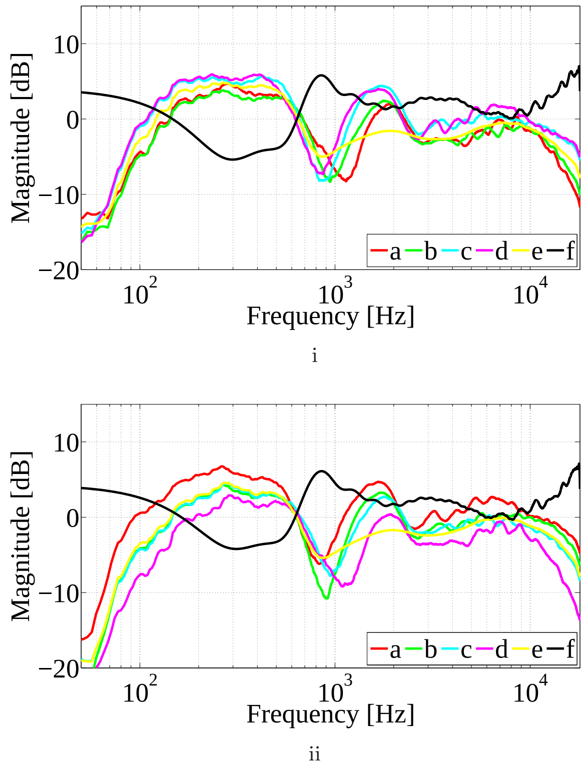

Fig. 4: Magnitude frequency response of (a), (b), (c), (d) smoothed frequency response, (e) prototype function, (f) equalizer considering (i) left and (ii) right channels.

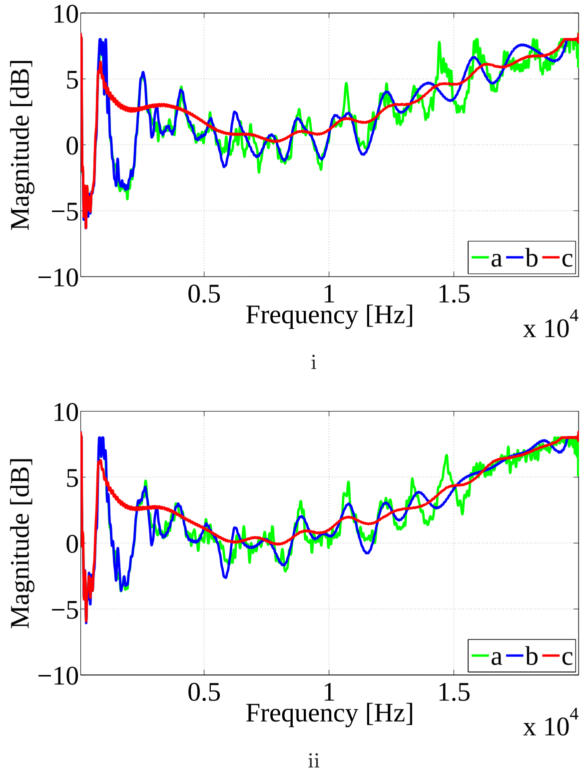

Fig. 5: Magnitude frequency response of equalizer functions obtained with (a) the approach of [14] , (b) the method of [19], (c) the proposed approach considering (i) left and (ii) right channels.

been tested considering the aforementioned scenario. Figure 4 shows the magnitude responses of the identified IRs at different microphone positions, of the prototype function, and of the equalizer, considering the left and right channel. It is evident that the prototype follows the behaviour of the identified room impulse responses, in both cases.

Figures 5 show comparison between the proposed approach and the methods of the state of the art used as terms of comparison. The plots are reported without a logarithmic scale in order to underline the behaviour of the proposed approach in the mid-high frequency spectrum that shows a lower equalization effect. The use of a non-uniform resolution which decreases with increasing frequency to obtain a less precise equalization at higher frequency is an important aspect since it results in a broader equalized zone [22]. In order to verify this issue, the equalization effect is evaluated taking into consideration other positions different from those used for the prototype extraction, i.e., IR5, IR6, IR7, IR8, IR9, IR10, IR11, IR12 of Figure 3. In particular, two measures have been selected to analyze the equalization results:

- the spectral deviation [23, 24], which gives a measure of the deviation of the magnitude frequency response away from a flat one [24], considering the single IRs before and after the equalization;
- the Sammon map [25], which takes into account all the measured IRs at a time, and it is a useful representation of impulse responses and environment response equalization.

Table 1: Spectral Deviation calculated over the entire set of IRs considering the left channel, where Method 1 is the proposed method, Method 2 is the approach of [19], Method 3 is the approach of [14].

	IR1	IR2	IR3	IR4	mean
Original	2.999	2.077	1.555	1.678	2.077
Method 1	2.246	1.384	1.186	1.359	1.544
Method 2	2.166	1.267	1.149	1.237	1.455
Method 3	2.090	1.215	1.006	1.172	**1.371**
	IR5	IR6	IR7	IR8	mean
Original	3.571	2.570	2.371	2.467	2.745
Method 1	2.818	2.273	2.188	1.792	**2.268**
Method 2	2.763	2.454	2.301	1.663	**2.295**
Method 3	2.855	2.450	2.377	1.733	2.354
	IR9	IR10	IR11	IR12	mean
Original	2.41	1.847	1.851	2.97	2.109
Method 1	2.127	1.663	1.688	1.881	**1.840**
Method 2	2.252	1.887	1.863	2.003	2.001
Method 3	2.259	1.880	1.891	2.013	2.011

Table 2: Spectral Deviation calculated over the entire set of IRs considering the right channel, where Method 1 is the proposed method, Method 2 is the approach of [19], Method 3 is the approach of [14].

	IR1	IR2	IR3	IR4	mean
Original	1.567	1.606	1.862	2.879	1.979
Method 1	1.166	1.113	1.220	2.022	1.380
Method 2	1.026	1.089	1.108	1.946	1.292
Method 3	0.945	0.994	1.062	1.907	**1.227**
	IR5	IR6	IR7	IR8	mean
Original	2.260	2.130	2.437	3.396	2.556
Method 1	1.656	1.833	2.136	2.570	**2.049**
Method 2	1.527	1.964	2.283	2.554	**2.082**
Method 3	1.557	2.006	2.318	2.594	2.119
	IR9	IR10	IR11	IR12	mean
Original	2.186	2.037	2.184	2.524	2.233
Method 1	1.903	1.734	1.828	2.063	**1.882**
Method 2	2.034	1.851	1.968	2.122	1.994
Method 3	2.059	1.909	1.957	2.174	2.025

The spectral deviation (SD) is a well-known parameter used to evaluate the local results of the equalization procedure [24, 10, 11, 23, 26, 27] and to compare different techniques applied to the same environment. In this case, it is used in order to evaluate the performance of the algorithms in different positions of the environment. Tables 1 and 2 show the results obtained considering the twelve IRs of Figure 3. It is evident that increasing the distance from the position used to determine the equalization filter (i.e., IR1, IR2, IR3, IR4), the proposed approach shows better performance with lower spectral deviation values, confirming its validity in increasing the zone where the equalization takes effect. This phenomenon is more evident increasing the listener distance with respect to the loudspeakers, and thus considering IR9, IR10, IR11, IR12 instead of IR5, IR6, IR7, IR8 where the approach of [19] is still valid.

This fact is confirmed also in Figures 6 that depict IR magnitude spectra resulting after equalization procedures at the different positions using the three analyzed methods. The equalization procedure should ideally lead to the target curve. Obviously, this cannot be achieved since the equalizer is derived from a set of IRs and considering a quasi-anechoic procedure. However, the figures show good results with the proposed equalizer for both channels, with a considerable uniformity, considering more distant IRs.

The Sammon map is a non linear projection algorithm method that maps multidimensional data onto lower dimensions (e.g., two or three). The main property of the Sammon map is that it retains the geometrical distances between signals in multidimensional space on the two or three-dimensional space. After equalization, the resulting equalization performance can be determined from the size and shape of the region determined by the equalized frequency responses on the map (circular shape indicates uniform equalization performance at all locations). It is worth noting that, in all reported cases, the equalized magnitude responses are processed by subtracting the individual means, computed between the equalization range ($50 - 20000$ Hz) to give the zero mean equalized magnitude responses. In this way, under ideal equalization, magnitude responses will be located at the origin of the

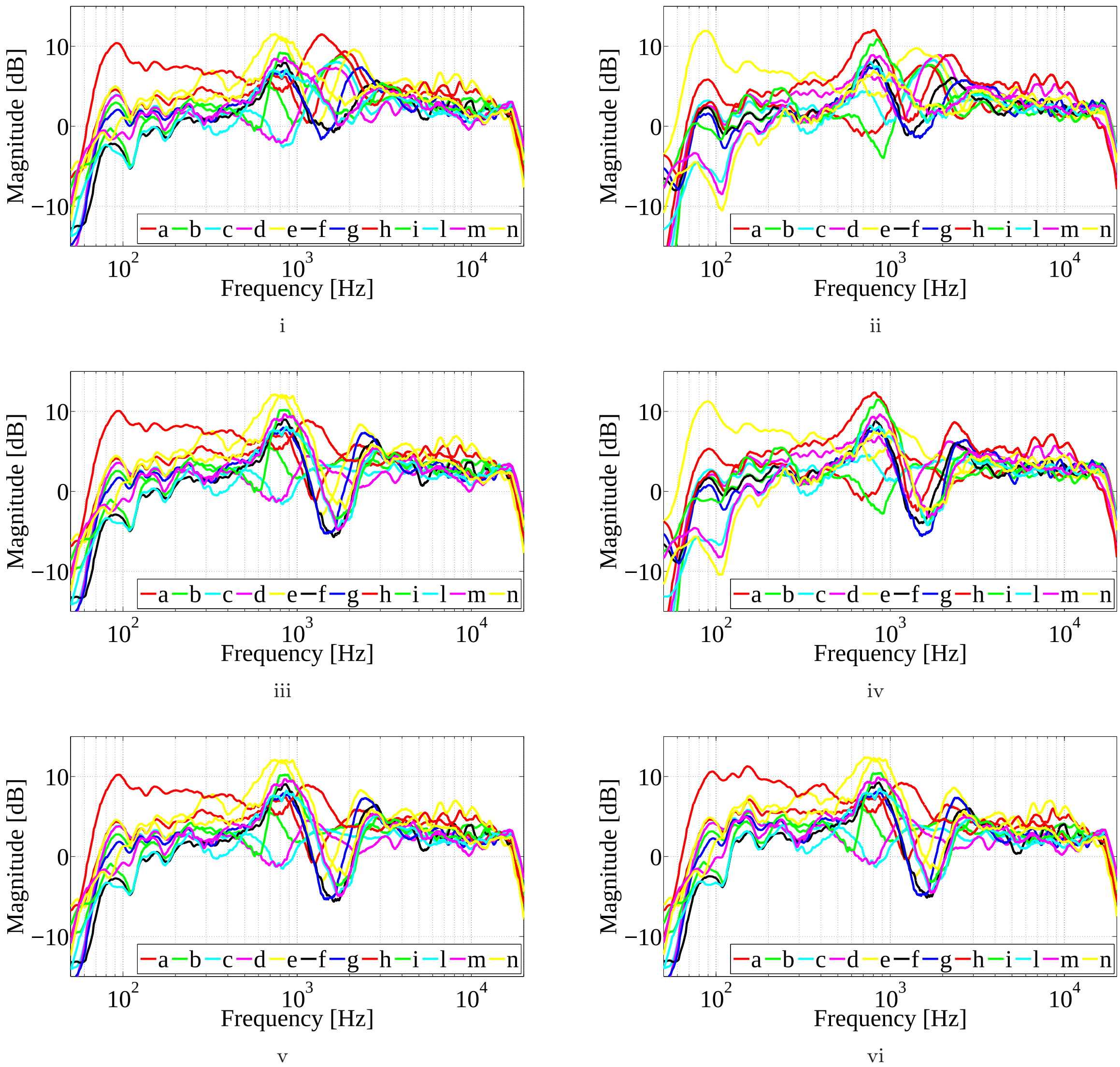

Fig. 6: Magnitude frequency response of the twelve impulse responses (from letter a to n) after equalization procedure considering (i) and (ii) the proposed method, (iii) and (iv) the method of [19], (v) and (vi) the approach of [14]: the first column represents the left channel while the second column represents the right channel.

Sammon map, so that displacement from the origin indicate deviation from ideal flat equalization, as also shown in [24].

Figures 7 show the obtained Sammon map considering the three compared approaches for the twelve positions. It is evident that proposed approach shows the better IRs distribution that is uniform around the origin, while the other approaches point out a worst performance with elliptic distribution rather than circular. This is more evident for the method of [14], as also underlined in the analysis of the spectral deviation.

3.2. Subjective Analysis

Preliminary informal listening tests have been per-

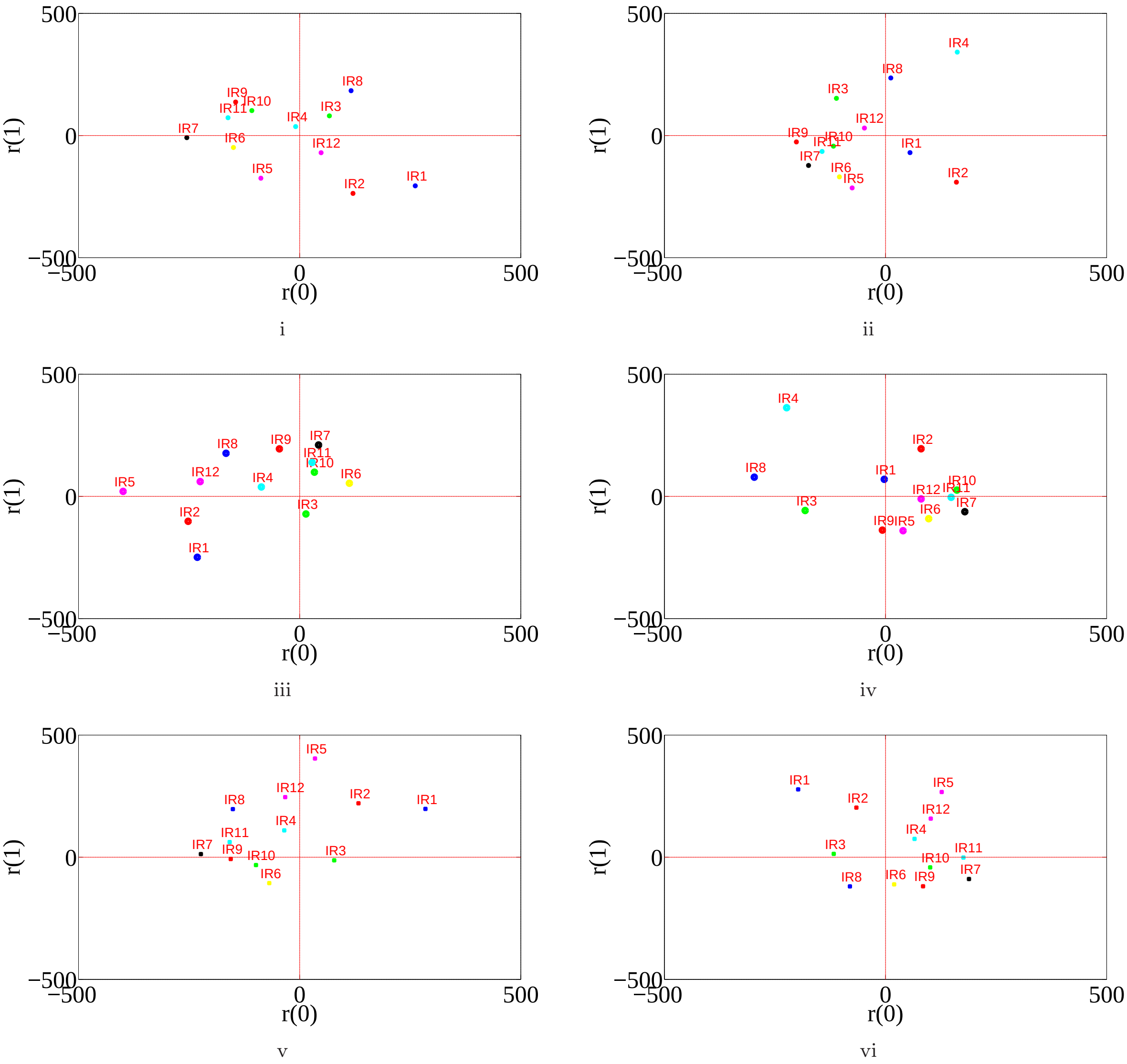

Fig. 7: Sammon map considering the twelve impulse responses after equalization procedure considering (i) and (ii) the proposed method, (iii) and (iv) the method of [19], (v) and (vi) the approach of [14]: the first column represents the left channel while the second column represents the right channel.

formed by reproducing audio material to evaluate the perceptive effect of the equalization, considering expert listeners, i.e., people with a background in acoustics and then not representative of the normal population. Different tracks belonging to different music genres have been considered (i.e., pop, rock, country and soul) and they have been filtered using the equalizer responses of Figure 6. The listeners' position has been fixed in the location where IR1, IR2, IR3, and IR4 were measured. For all the tests, a flat target curve has been considered. The obtained results have confirmed the validity of the proposed approach since all involved subjects have reported positive comments and impressions on the perceived global sound image, including stereo image and spatial representation. Therefore, the sound impression perceived was very similar to those obtained by the methods under comparison i.e., the approach of [19] and [14]. However, as demonstrated in the previous section, the advantage of the proposed method in comparison with the state of the art refers to the enlargement of the equalized zone. For this reason a deeper subjective investigation is now required considering positions different from those used to derived the equalizer response in order to test the robustness of the approach: these results will be presented in a future work.

4. CONCLUSIONS

In this paper, a multipoint fixed equalization approach was presented. The equalizer is designed in the frequency domain to achieve magnitude response equalization of the selected environment. From the set of impulse responses recorded in a real environment, a pre-equalization procedure is performed: it is derived from the combination of quasi-anechoic impulse responses with the impulse responses recorded in the environment to be equalized. This is motivated by the fact that at mid and high frequencies the timbre perception and localization is dominated by the direct sound, thus, the measurable, but mostly inaudible magnitude deviations due to reflections should not be equalized [28]. Then, a suitable prototype is extracted from the set of the pre-equalized IRs by considering a mean approach and the inverse filter is obtained through frequency deconvolution with regularization of the prototype. Some objectives results have been reported and in particular, a comparison with some state of the art methods have been performed. From these results it appears that the proposed method is an efficient and effective solution for dealing with IR equalization. This fact has also been confirmed by informal listening tests. However, further works will be oriented toward evaluations of the system through intensive subjective listening tests taking into comparison different methods.

5. ACKNOWLEDGEMENT

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