Quasi-Anechoic Measurement of Loudspeakers Using Beamforming Method

Quasi-bezechowy pomiar głośników z wykorzystaniem metody kształtowania wiązki

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***Abstract*—This paper describes results of the research on enhancement of loudspeaker impulse response measurement in small room. Proposed enhancement is a combination of MLS (Maximum Length Sequence) algorithm and Delay-Sum Beamforming in order to measure frequency response of loudspeakers even in very small and poorly suppressed rooms. Application of the beamforming algorithm allows the extension measure of impulse response by damping reflections from walls, floor and ceiling.**

Keywords: loudspeaker impulse response, maximum-length sequence, delay-sum beamforming

***Streszczenie* – w artykule opisano wyniki badań dotyczących poprawy jakości pomiaru odpowiedzi impulsowej głośników dokonywanego w małym pomieszczeniu. Proponowana metoda poprawy jakości pomiaru jest połączeniem algorytmu MLS (ang. maximum-length sequence) oraz kształtowania wiązki na zasadzie sumowania opóźnionych sygnałów (ang. delay-sum beamforming) w celu pomiaru charakterystyki amplitudowo-częstotliwościowej głośników nawet w bardzo małych i słabo wyciszonych pomieszczeniach. Zastosowanie algorytmu kształtowania wiązki umożliwia pomiar odpowiedzi impulsowej poprzez wytłumienie odbić od ścian, podłogi i sufitu.**

Słowa kluczowe: odpowiedź impulsowa głośnika, MLS, kształtowanie wiązki

Introduction

Nowadays, different methods of the loudspeaker impulse response measurement are used. Classical methods based on sinusoidal signals require an anechoic chamber in order to eliminate influence of the room acoustic. Other solution is to use a quasi-anechoic methods [1], [2] which give possibility of the measurement in small and poorly suppressed room [3]. Presently one of the most popular quasi-anechoic method seems to be maximum length sequence (MLS) method [4], [5], [6]. The main idea of the method is gating the impulse response what enables to reject reflections from the walls, floor and ceiling. As a result, measurements can be made in the ordinary living room with a sufficiently large dimensions. Therefore MLS has gained immense popularity among amateur teams that build speakers. Moreover, a lot of audio journals has gained the ability to publish characteristics of tested loudspeakers measured in their newsrooms. However the main problem of the MLS method in small rooms is too short gating time of impulse response. It causes decreasing of the overall loudspeaker frequency response reliability in lower frequency range [7]. Possible solution of the problem is increasing room dimensions or dumping walls, floor an ceiling what is often very difficult or impossible. In the paper an alternative novel method of eliminating short gating time problem is presented. Proposed method is based on delay-sum beamforming. The beamforming effect can be achieved using a simple linear array of microphones. Delay-sum beamforming is simplest of all microphone array beamforming techniques [8], [9]. Microphone arrays are used mainly in speech recognition systems for noise reduction in acoustic signal in order to improve recognition accuracy. In problem presented in the paper, delay-sum beamforming is used for suppression of early and late room reflections.
Methods based on the microphone position change eg. by the manipulator [10], [11] are applied in specific electroacoustic measurements. In the research presented here, a single microphone changing its position in space by linear slide has been used instead of linear microphone array. Results of the preliminary research has been presented in [12].

Problem Statement

As mentioned in the introduction, quasi anechoic methods operate by gating the impulse response. The basic problem is the appropriate size of the room in which the measurement is carried out. Gate time of impulse response determines directly lower frequency limit of the frequency response. Increasing the gating time allows to measure the lower frequency band of the loudspeaker. However the lowest frequency part of loudspeaker frequency response should be measured in near field because of room modes [7]. Whole frequency response is obtained by combination frequency responses in near and far field. Nevertheless, in small rooms, the time between direct sound and the first reflection is too short to obtain reliable frequency response in lower and medium frequency range therefore there is a problem of proper combination of frequency responses in middle frequency range. Moreover, regardless of the room size the problem of short time between direct sound and reflection from the floor remains unsolved. This problem results from usual loudspeaker placement close to the floor surface. Some remedy for this situation can be covering the floor with damping material like thick rugs or quilts. Whereas suppressing the floor is relatively simple, in case of walls and especially the low ceiling this operation becomes difficult or even impossible. Not taking into account above problems, only whole-band frequency response can be obtained by measurement combination. Whole impulse response of loudspeaker meaurement is still not possible. Therefore, using the beamforming algorithm seems to be reasonable alternative. In order to explain proposed beamforming methods four cases of MLS measurement will be analyzed in the next sub-chapters: A. Reference loudspeaker measurement in anechoic chamber, B. Standard loudspeaker measurement in room with near and far field results combination, C. loudspeaker measurement in room using beamforming, D. Comparison of measurement results.

A. Reference loudspeaker measurement in anechoic chamber

Reference measurement in anechoic chamber has been accomplished in order to compare efficiency of methods presented beneath. Impulse response measurement has been carried out at a distance of 1m from the front of loudspeaker. Microphone has been positioned on loudspeaker axis at a height of tweeter.

B. Standard loudspeaker measurement in room with near and far field results combination

Loudspeaker and measurement microphone has been positioned approximately in the middle of the room similarly like in the anechoic chamber. Microphone has been placed at a distance of 1m from the front of loudspeaker. at a height of tweeter. The measurement on loudspeaker axis has been accomplished. Afterwards microphone has been placed in front of woofer and near-field measurement has been carried out.

C. Loudspeaker measurement in room using beamforming

General idea of suppressing the reflection with vertical linear array using beamforming is presented in fig.1. Here, desired sound which can arrive from arbitrary chosen direction (in general case) to the reference microphone is cross-corellated with sounds arriving to the other microphones in order to calculate delay times (fig.1.a).

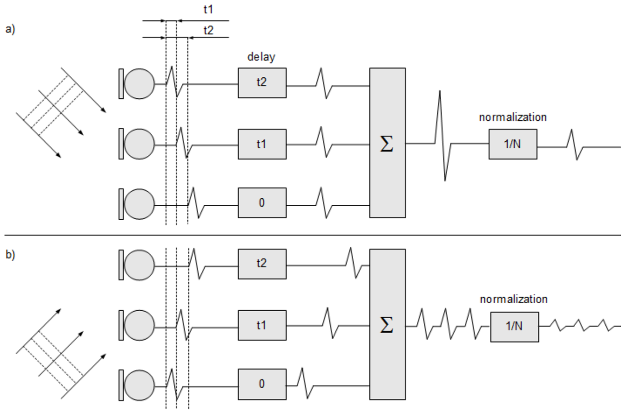

Fig. 1. Delay-Sum Beamforming technique

Rys.1. Wyjaśnienie techniki ksztatowania wiązki na zasadzie sumowania opóźnionych sygnałów

Afterward, signals from all microphones are shifted by the calculated delay times and summed so as to achieve amplified desired signal. Desired signal is N-fold amplified where N is the number of microphones. Unwanted signals from directions different from the chosen one are also summed up but are uncorrelated (fig. 1.b). In effect, after normalization by dividing the signal by N, these uncorrelated signals are suppressed the more the higher number of microphones is.

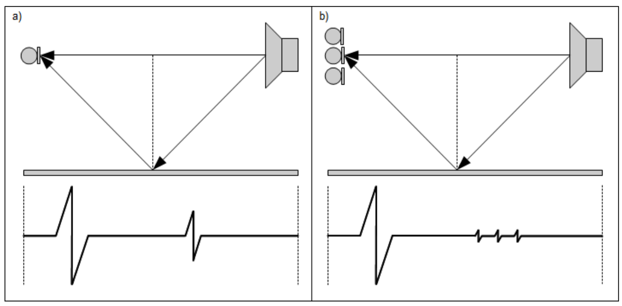

Fig. 2. Suppressing reflection by delay-sum beamforming.

Rys.2. Tłumienie odbić za pomocą kształtowania wiązki poprzez sumowanie opóźnionych sygnałów.

Fig. 2 presents principle of dumping unwanted sound reflection from wall by beamforming algorithm. Dumping is possible because reflected sound reaches microphones at different times and sum of microphone signals is smaller than sum of microphone signals coming from direct sound reaching all the microphone at the same time. Of course signals must be summed up without any delay between each other in this case (so as to amplify direct sound). In case of MLS measurements generated signal is deterministic and can be repeated many times. Also results of measurements are repeatable. Therefore classical microphone array can be replaced by an array with multiplexed channels or by device with appropriately positioned single microphone. Therefore single vertically oriented linear microphone array has been sufficient to carry out beamforming. Another reflections coming from walls and ceiling appear in case of closed small room (fig. 3).

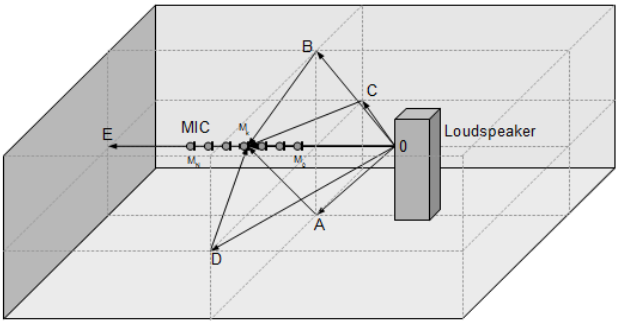

Fig. 3. First reflections in room

Rys. 3. Odbicia bezpośrednie w pomieszczeniu.

In order to suppress all these reflection more sophisticated array of omnidirectional microphones should be used. However in case of MLS measurement there exist more simple solution – longitudinal linear microphone array.

In order to show ability of suppressing reflected signals by the longitudinal linear microphone array we must define several distances between the points from fig. 3. These are following:

$$d_1 = \left|\overline{OM}_k\right| = \left|\overline{OM}_0\right| + k \cdot L \tag{1}$$

$$d_2 = \left|\overline{OE}\right| + \left|\overline{EM}_k\right| = \left|\overline{OM}_0\right| + 2\left|\overline{EM}_0\right| - k \cdot L \tag{2}$$

$$d_n = 2\sqrt{h_n^2 + \left(\frac{1}{2}\left|\overline{OM}_k\right|\right)^2} = 2\sqrt{h_n^2 + \left(\frac{1}{2}\left|\overline{OM}_0\right| + k \cdot L\right)^2}, n = 3,4,5,6 \tag{3}$$

where:

h_3, h_4, h_5, h_6 – heights of triangles OAM_k, OBM_k, OCM_k, ODM_k respectively with the base OM_k,

M_k – position of k-th microphone where $k=0,1, \ldots, N-1$,

N – number of microphones,

L – distance between uniformly spaced microphones in the longitudinal microphone array.

Time between direct sound and given reflection depends on distance differences: $\Delta_R(k)$ for rear wall and $\Delta_{SFC}(k)$ for side walls, floor and ceiling which are obtained from equations (1), (2) and (3) as follows:

$$\Delta_R(k) = d_2 - d_1 = 2\left(\left|\overline{EM}_0\right| - k \cdot L\right) \tag{4}$$

$$\Delta_{SFC}(k) = d_n - d_1 = 2\sqrt{h_n^2 + \left(\frac{1}{2}\left|\overline{OM}_0\right| + k \cdot L\right)^2} - \left(\left|OM_0\right| + k \cdot L\right) \tag{5}$$

It results from equations (4) and (5) that the delay times of the first reflection with respect to direct signal are not constant because depend on k, and in consequence signal reflections are uncorrelated. Due to these relations longitudinal linear array can suppress reflections from side walls, rear wall, floor and ceiling.

The solution with longitudinal array can be simplified furthermore by using only one microphone changing its position by distance L when subsequent measurement is carried out. Besides of significant decreasing of overall cost, single microphone method gives also significant improvement in measurement accuracy because there are no other microphones covering up direct sound.

D. Comparison of measurement results

The efficiency of the method proposed in the paper was tested by comparison of the frequency response obtained by the method with frequency response measured in anechoic chamber. Furthermore, the frequency response measured by beamforming was compared with the response obtained in the classical way, ie by combining measurements: gated on axis and near-field.

Experiments and Results

Four types of experiments have been carried out in the research:

A) Reference loudspeaker measurement in anechoic chamber,

B) Standard loudspeaker measurement in room with near and far field results combination,

C) Loudspeaker measurement in room using beamforming,

D) Comparison of measurement results,

The equipment based on high speed Successive Approximation Register (SAR) analog-to-digital converter with sampling rate 96kHz and resolution 16 bit has been used for MLS measurements. Superior performance of SAR analog-to-digital converters over the sigma-delta ones in audio measurement based on MLS system has been shown in [13], [13] and [15]. For precise MLS signal generation digital-to-analog converter was given up. Output buffer amplifiers are driven directly from processor. Input and output circuits were designed for a wide frequency range. Measurement microphone was constructed using the Panasonic WM-61 capsules. The Behringer uses the same capsule for the construction of the measuring microphone ECM8000.

Fig. 4. Measurement microphone mounted on the linear slide

Rys.4. Mikrofon pomiarowy zamocowany na prowadnicy liniowej

Microphone is moved from measurement to measurement on the linear slide equipped with the electric engine and encoder reading position of the slide. The length of the slide is 1,5m. The controller coupled with measurement card enables achieving of shifting resolution of 0,3mm.Two way speaker has been used in experiments.

A. Reference loudspeaker measurement in anechoic chamber

Measurement of the impulse response has been carried out in anechoic chamber by MLS method.

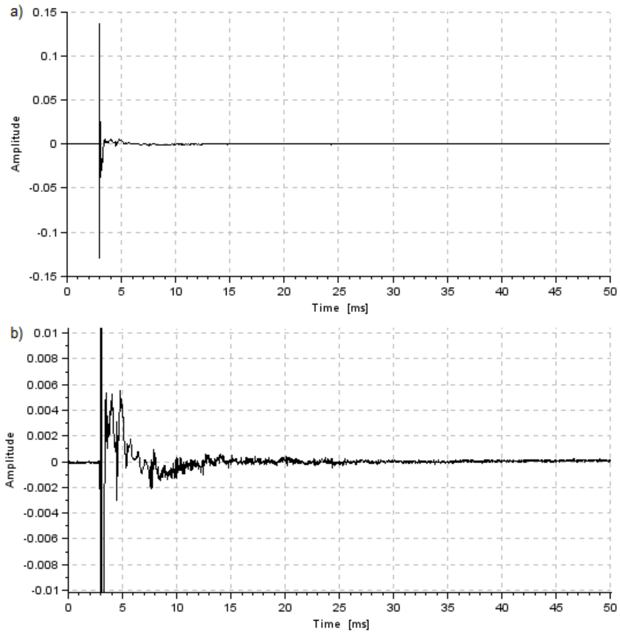

Fig. 5. Impulse response of loudspeaker in anechoic chamber

Rys.5. Odpowiedź impulsowa zespołu głośnikowego w komorze bezechowej

Fig. 5 a) presents impulse response measured in anechoic chamber. The same response zoomed vertically has been shown on Fig. 5 b). Measurement microphone has been positioned on axis at a distance of 1m from measured loudspeaker and at a height of tweeter. Gating time was 50 ms what enabled generation of actual frequency response in the range 20Hz÷20kHz after fast Fourier transform (FFT). Response within entire band has been presented on Fig.6.

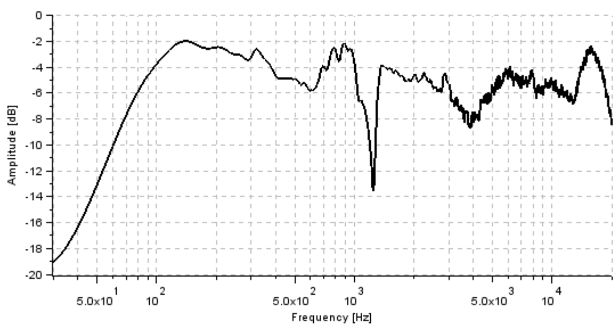

Fig. 6. Frequency response of loudspeaker in anechoic chamber

Rys.6. Charakterystyka częstotliwościowa zespołu głośnikowego w komorze bezechowej

B. Standard loudspeaker measurement in room with near and far field results combination

Another experiment was to measure the same loudspeaker. Measurement microphone was set as in the previous experiment. Fig. 7 shows the impulse response with a duration of 50 ms. It shows clear reflections from the room surfaces.

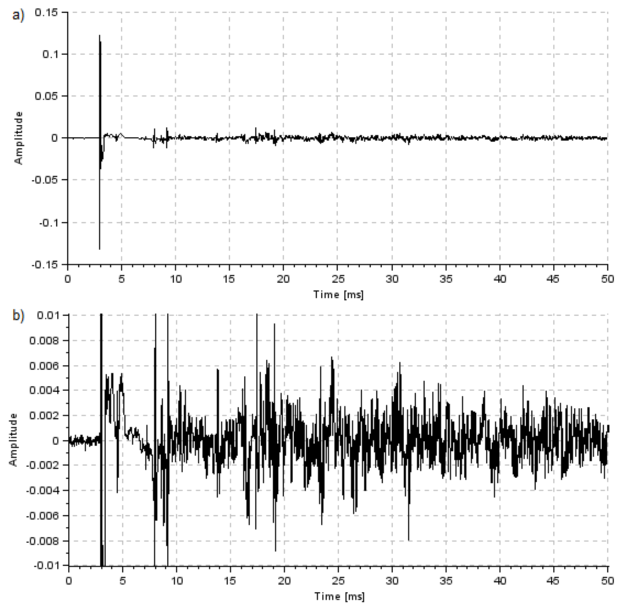

Fig. 7. Impulse response of loudspeaker in room

Rys.7. Odpowiedź impulsowa zespołu głośnikowego w pokoju

The frequency response generated from this impulse response by the FFT is highly distorted by reflections of the room which makes it non-readable (Fig. 8).

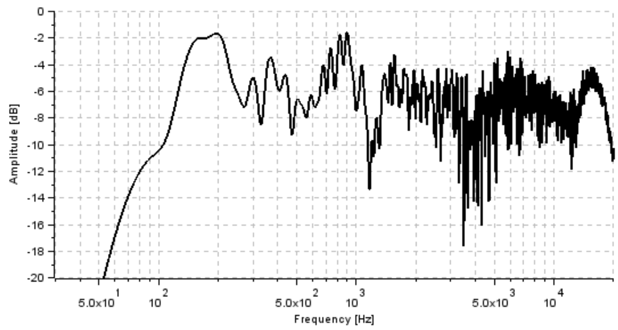

Fig. 8. Frequency response of the loudspeaker in the room

Rys.8. Charakterystyka częstotliwościowa zespołu głośnikowego w pokoju

In order to generate the frequency response, standard procedure has been carried out. The gating of the impulse response part from the impulse beginning to the first reflection was accomplished by the markers (Fig. 9). The time of gated impulse response in this case is approximately 4.5 ms.

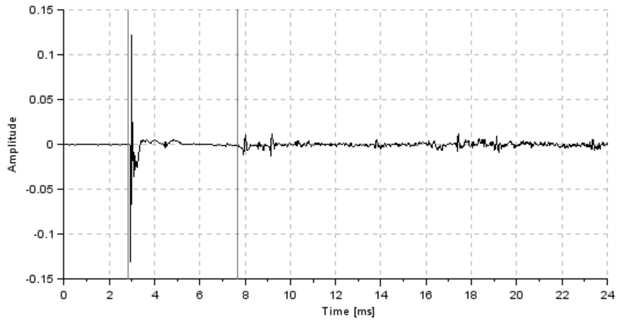

Fig. 9. Gating of impulse response

Rys. 9. Bramkowanie odpowiedzi impulsowej

Such a short period of the impulse response enables to generate the frequency response reliable in the range above 200Hz. Therefore, it is necessary to perform measurements in the near field and attach it on the chart. Near-field measurement is performed at a distance of 1 2 cm from the woofer. In such a case, reflections in the room which is considerably quieter than sounds emitted by woofer have a negligible effect on the measured impulse response.

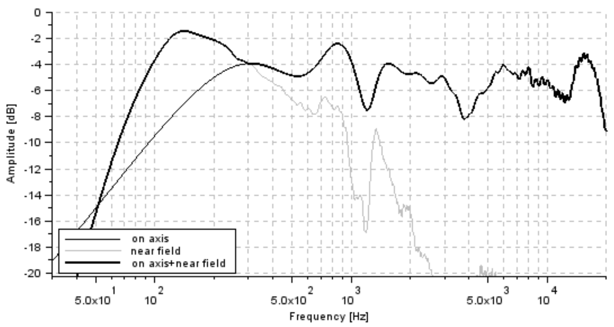

Fig. 10. Frequency response (on axis + near-field)

Rys.10. Charakterystyka częstotliwościowa zespołu głośnikowego złożona z pomiarów w osi i w polu bliskim

The disadvantage of this method is that the operator needs to intuitively find the right place on the chart and adjust the value of the response measured in the near-field to the chart of the response measured on axis. This causes significant errors in the method. Figure 10 shows the frequency response within entire band composed of gated measurement on the loudspeaker axis and the near-field measurement.

C. Loudspeaker measurement in room using beamforming

The experiment was conducted under the same conditions as the previous one except that the microphone was mounted on the linear slide controlled by the measurement card. Reflections has been suppressed by delay-sum beamforming method but using only one microphone which has been placed in 100 different positions along the OM axis (fig. 3). Microphone has been moved within the range 0.5m ÷ 1.5m from loudspeaker with the step of 1cm. One MLS measurement has been carried out in every microphone position. Comparison of impulse responses obtained by single microphone and by the beamforming is presented in fig.11.

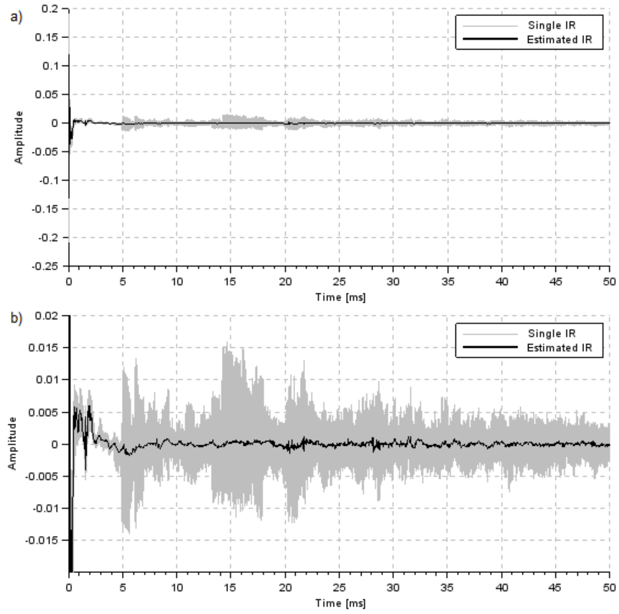

Fig. 11. Comparison of 100 superimposed loudspeaker impulse responses from single microphone, correlated in time domain with respect to direct signal (single IR) with impulse response obtained by beamforming of imposed single microphone impulse responses (estimated IR)

Rys.11. Porównanie 100 nałożonych odpowiedzi impulsowych z pojedynczego mikrofonu, skorelowanych czasowo względem sygnału bezpośredniego (single IR) z odpowiedzią impulsową otrzymaną przez kształtowanie wiązki z nałożonych odpowiedzi impulsowych z pojedynczych mikrofonów (estimated IR)

Estimated frequency response required correction by replacing part of estimated impulse response (up to the first reflection) with signal from microphone placed at a distance 1m from loudspeaker. Fig. 12 shows frequency responses resulting from single measurements and estimated by beamforming.

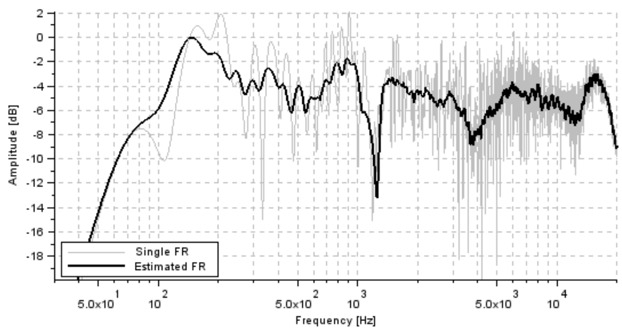

Fig. 12. Speaker frequency responses obtained by single microphone (single FR) and estimated by beamforming (estimated FR)

Rys.12. Charakterystyki amplitudowo-częstotliwościowe otrzymane z pojedynczego mikrofonu (single FR) oraz estymowane poprzez kształtowanie wiązki (estimated FR)

D. Comparison of measurement results

In order to verify the effectiveness of the method, results from previous experiments was imposed to a common plot which is shown in Fig. 13.

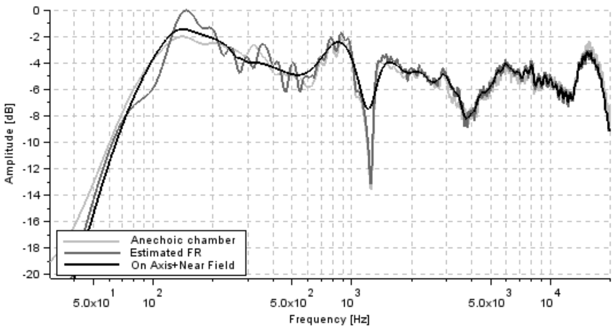

Fig. 13. Comparison of frequency responses measured in anechoic chamber (anechoic chamber) by assembling gated measurement on axis with near-field measurement (on axis + near-field) and by estimation by beamforming (estimated FR)

Rys.13. Porównanie charakterystyk częstotliwościowych zmierzonych w komorze bezechowej przez złożenie bramkowanego pomiaru w osi z pomiarem w polu bliskim (On Axis+Near Field) oraz estymowanej poprzez kształtowanie wiązki (estimated FR)

Reference is the response measured in an anechoic chamber. In a direct comparison more or less significant differences can be seen in responses obtained in different experiments. The standard method of combining gated and near-field measurements leads above all to errors resulting from merging charts by man according to his individual interpretation. Furthermore, a short gating time leads to a significant misrepresentation of the actual response in the range of several hundred Hz to several kHz. Resonance masking is clearly evident on the response shown in Fig.13. Difference with respect to measurement is as much as 6 dB. Response measured using delay-sum beamforming has little ripples at the bottom part, however, the difference with respect to anechoic chamber does not exceed 1.5dB.

DISCUSSION

Experiments described in the chapter "Experiments and Results" need certain discussion. In general, problem of reflection suppression concerns many surfaces therefore more sophisticated omnidirectional microphone array seems to be a must. However results of experiment from the chapter "Experiments and Results" part C confirm that according to the idea described in "Problem statement" part C it is possible to position linear array so as to suppress reflections from all directions. Moreover in case of MLS the array can be replaced with single microphone which change its position during each single measurement.

CONCLUSIONS

In the paper a novel method of the MLS loudspeaker frequency response measurements has been shown. Presented approach is based on the beamforming technique used for eliminating unwanted component in the impulse response of tested loudspeaker. A solution with single microphone changing its position in order to imitate linear longitudinal microphone array has been proposed. Results show around 40 dB suppressing of unwanted components in impulse response. It gives more accurate loudspeaker frequency response in the range of middle and higher frequencies. The method using beamforming does not require operator intervention. With the microphone placed on the controlled slide, measurement procedure is performed completely automatically. Most importantly, the experiment showed that the method gives results very close to these obtained in an anechoic chamber

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