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Subwoofers in Rooms: Equalization of Multiple Subwoofers

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ABSTRACT

The effectiveness of multiple subwoofers in controlling low-frequency room modes can be improved through designing individual equalization for each of the subwoofers. The paper discusses strategies towards equalizer design, including individual equalization of multiple subwoofer responses, minimizing the total energy radiated to the room, and optimization for minimizing sound field variation. The frequency responses and sound field distributions obtained using these methods are compared to the results of conventional equalization and modal control only through loudspeaker placement.

1. INTRODUCTION

It is well established that the coupling of single or multiple subwoofers to room modes can be controlled through loudspeaker placement [1]. Already the optimal placement of a single subwoofer can reduce the frequency response variation in a single listening location, but if control over sound field distribution over any area is needed, then multiple subwoofers are needed.

However, when all the loudspeakers are driven with identical signals there are limits to the degree of sound field control. Optimizing the loudspeaker system performance with respect to a small number of modes can degrade the performance with modes that were not target of the initial optimization, and optimizing with respect to too large number of modes will reduce the improvement for any single mode. Real-life rooms have more complex modal shapes than the theoretical hard-walled highly symmetrical rooms, so achieving control over a group of modes can often prove very difficult. Also, it is quite possible that non-acoustical considerations completely prevent placing the one or more of the subwoofers in optimal locations, so the result is also suboptimal.

These inherent shortcomings of purely acoustical approach can be overcome by increasing the degrees of freedom in the optimization by feeding the loudspeakers with non-identical signals. The paper discusses approaches to multi-speaker equalization.

Methods used here as references to these more complex approaches are straightforward equalization of the response of the multi-loudspeaker group over the listening area, and equalizing each speaker individually at a chosen listening location. These simple approaches do not improve sound field distribution, but on the other hand, use of multiple loudspeakers and optimization of their locations does make the equalization easier by reducing the spatial variation of the sound field.

2. EQUALIZATION METHODS

The examples below were measured in a conventional living room where the listening area is acoustically asymmetrical, with a corridor and openings to adjacent areas on one side of the room, and straight wall on the other side. The two loudspeakers used in the test were placed in the opposite (upper and lower) corners of the listening area. The loudspeaker positions were deliberately chosen to be non-optimal, with point 1 yielding rather usable response but with point 2 less suited for listening on its own, and with sum signal not optimized. This ensured that the possible effects of the optimization would be clearly visible, but the locations were still chosen so that it was possible to couple the excitation to points with different phase for most of the lowest modes. The responses were measured at the normal listening position and at two points at 0.5 m distance from the listening position.

2.1. Individual speaker equalization

The first test to establish the baseline for evaluating other solutions is the behavior of the individual loudspeakers. The graphs below illustrate the response equalized flat at the listening position (0 dB line) and the resulting responses in the adjacent microphone positions.

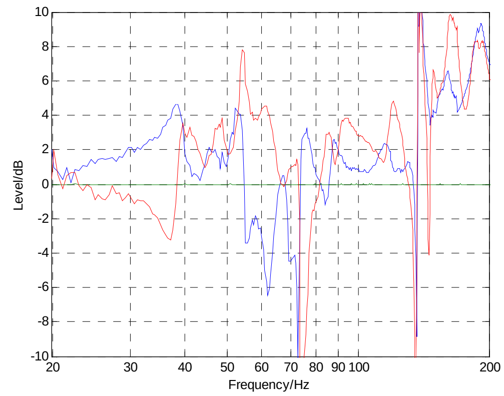

Figure 1 Loudspeaker 1 (lower corner, in front of the listener). Here, as in other frequency response graphs the 0 dB line represents the normalized response at the listening position and the other traces responses at adjacent points.

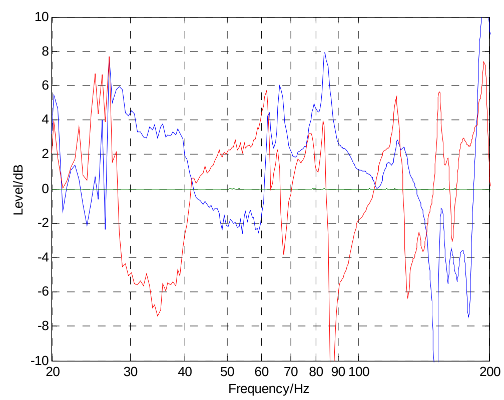

Figure 2 Loudspeaker 2 (upper corner, behind the listener, with corridor starting close to the speaker location).

2.2. Inverse equalization of sum signal

Simple straightforward equalization, where all the speakers are fed with a signal processed with the inverse of the measured sum response may not be a practical approach for low-frequency equalization, but it serves well as a benchmark for other equalization methods. If all the sources are fed similar signals, then it is possible to reduce the frequency response variations, but the differences in the response between the source locations stay the same. It is a good practice to measure the responses in several locations, but the choice or number of measurement locations only helps to avoid clearly non-preferable equalizations, and has no effect on the sound field distribution, which is determined only by the locations of the loudspeakers.

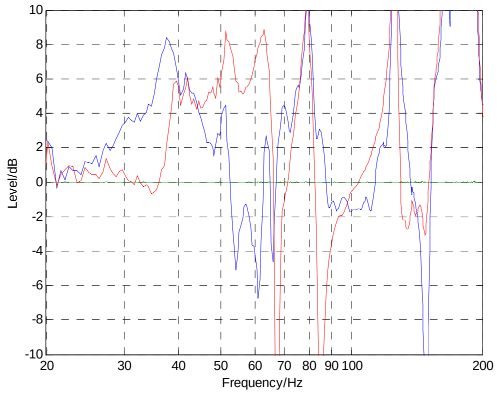

Figure 3 Sum of both loudspeakers.

2.3. Matched response method

A simple extension of the inverse equalization of the whole system is to apply the equalization individually to each speaker so that the responses of each speaker at a selected listening position are equal in magnitude and phase. In this case the responses at the reference location sum to equal value, but the response at other locations is different from the equal-drive case. This method is related to the matched impulse response methods used in underwater acoustics for source localization.

The theoretical benefit of this approach is that the signals should exhibit fully coherent summation only at the intended listening location. However, even in a fully reverberant field the coherence of the signal is significantly reduced only when the distance between observation points is about a quarter of a wavelength, and in a field with strong isolated modes the spatial correlation is strongly determined by the modal shape, so the efficiency of this approach is limited by the sound field characteristics.

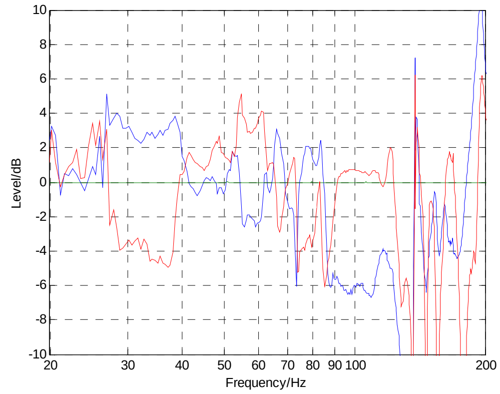

Figure 4 Matched response method.

2.4. Sound field energy minimization

Minimizing the total energy radiated to the room is related to the self and mutual radiation impedances of the loudspeaker. In this method the response of each loudspeaker to the listening location, to the acoustical centers [2] of other speakers, and to the acoustic center to the loudspeaker itself are measured. Assuming that the acoustical impedance of the loudspeaker is high enough so that the diaphragm movement is unaffected by the acoustical load these measurements enable determining the work the loudspeaker does against the sound field excited by all the loudspeakers in the system, and the equalizer design task is then minimizing at each frequency measured the energy needed to produce unity response at the listening position. This approach is closely related to minimum energy principles used in active noise control design, the only difference is that the target is to produce unity response rather than minimizing the sound pressure.

The work done by a loudspeaker to radiate the sound to the room consists of two parts: the work done against the sound field generated by the loudspeaker itself, and work done against the sound field generated by other sound sources. This can be regarded as an extension of the self and mutual radiation impedances discussed by Jacobsen [3], who used self radiation impedance to describe the load seen by a source in a baffle in free space, but here the measured impedance includes also the interaction between the room and the source. The mutual radiation impedance of the source and its reflections has been discussed also by Morse & Ingard [4] and Skudrzyk [5].

The radiation impedances are practically impossible to calculate with any accuracy in a complex room, but their approximate measurement at low frequencies is rather straightforward. Since it is safe to assume that the motion of the diaphragm of a sealed dynamic loudspeaker is essentially unaffected by room interaction, then the self radiation impedance is approximated by the sound pressure p_{ii} generated by volume velocity u at the acoustic center of the i:th loudspeaker, and the mutual radiation impedance is approximated by the sound pressure p_{ij} generated by other speaker(s) at the same location. The sound power radiated by the i:th speaker is then:

$$P_i = p_{tot} u = u\left(p_{ii} + \sum_{i \neq j} p_{ij} \right) \qquad (1)$$

The optimization task is then to minimize the sum of sound powers P_i with the constraint that the sum of the sound pressures generated by all the speakers at the listening locations equals unity. This approach resembles to some extent fundamental principles of active noise control, where it is possible to prove that the optimal controllers for noise field are those minimizing the radiated energy.

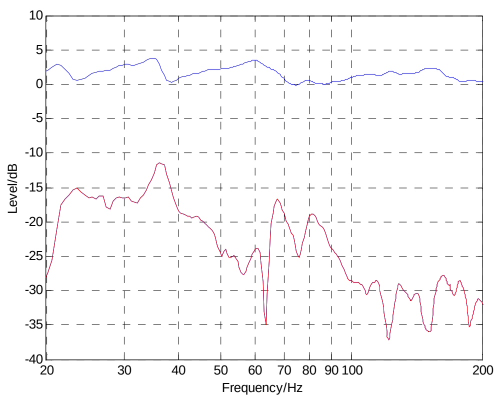

Figure 5 Normalized sound pressure generated by loudspeaker 1 (upper trace), measured 5 cm from the front surface (approximate acoustic center), and by loudspeaker 2 at the same location (lower trace).

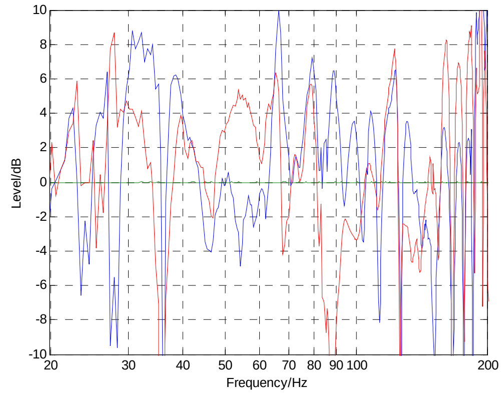

Figure 6 Sound field distribution obtained through minimization of energy radiation

The method is reasonably efficient in controlling higher modes, but, unfortunately, the results in low frequency range, below the first isolated modes, are rather disappointing. This may be due to high sensitivity to noise. Also, the algorithm tends to result in equalization functions that generate sound mostly from one loudspeaker at a time, so minimizing the radiated energy would be probably best used in as an additional criterion in conjunction with other optimization criteria.

2.5. Minimization of deviation

The methods discussed so far have used only one measurement point in the listening area, while the actual target is to minimize the deviation of the sound field over a larger area. Thus it is interesting to see whether it is possible to achieve usable results with direct minimization of the sound field deviation, keeping unity response at the listening location as an additional constraint. Two different optimization criteria were used: minimizing the maximum deviation of sound pressure level from the listening point, and minimizing the RMS variation of sound pressure level across the measurement area. As the results below indicate, the difference between minimizing the maximum error and RMS error is small probably due to the small number of loudspeakers and measurement points.

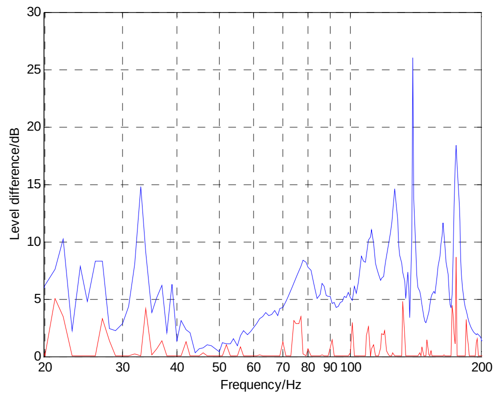

Figure 7 Response deviation across measurement area before (upper, blue trace) optimization, both loudspeakers driven with the same signal, and after (lower, red trace) optimization for minimizing maximum deviation.

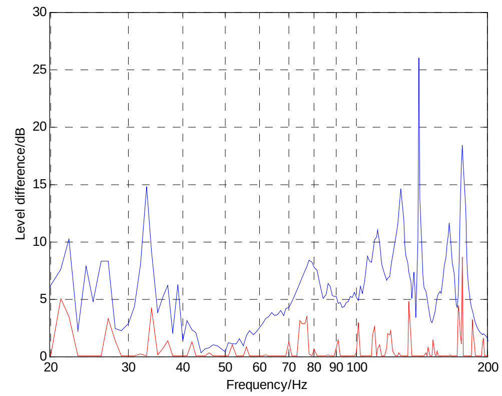

Figure 9 Response deviation across measurement area before (upper, blue trace) optimization, both loudspeakers driven with the same signal, and after (lower, red trace) optimization for minimizing RMS deviation.

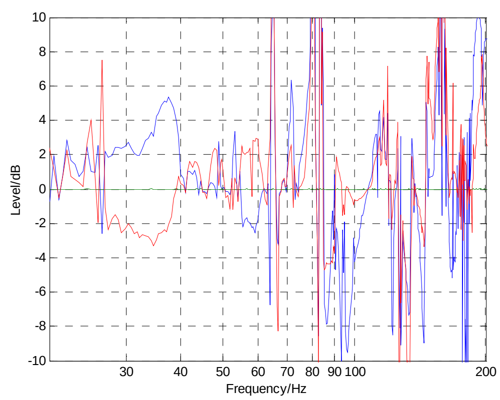

Figure 8 Response at listening point (black) and at adjacent locations (red and blue lines), minimizing maximum deviation.

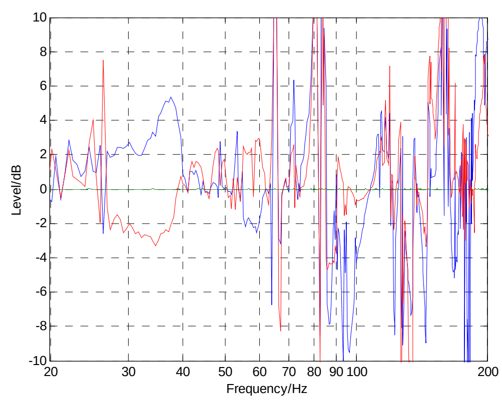

Figure 10 Response at listening point (black) and at adjacent locations (red and blue lines), minimizing RMS deviation.

3. COMPARISON OF DIFFERENT METHODS AND CONCLUSIONS

Although visual inspection of the results gives an impression of the properties, it is useful to make a quantitative comparison of the methods. A single figure of merit that can be used to compare the success of the different methods is the frequency average of spatial standard deviations of sound pressure levels across the measurement points. The number of measurement points in these examples is small for proper statistical analysis, so the results should be regarded only as a coarse indication of the differences. The frequency averages were calculated over the range of 20 – 100 Hz, which is the typical frequency range for subwoofers, and over the range 30 – 65 Hz, which is the sparsely modal range in the listening room used in this examples, and over which range the control of the modes should be the most efficient. The results show that in this case matched response method and minimizing the RMS deviation appear to yield the best results over both frequency ranges.

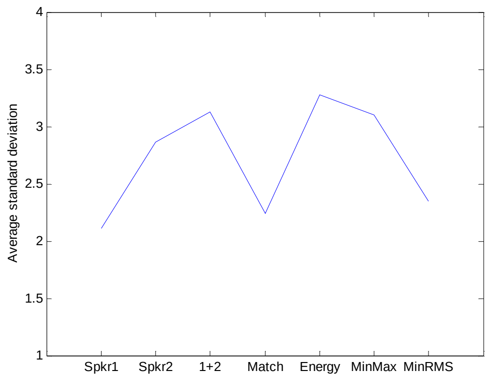

Figure 11 Frequency average of spatial standard deviation, frequency range 20 Hz – 100 Hz.

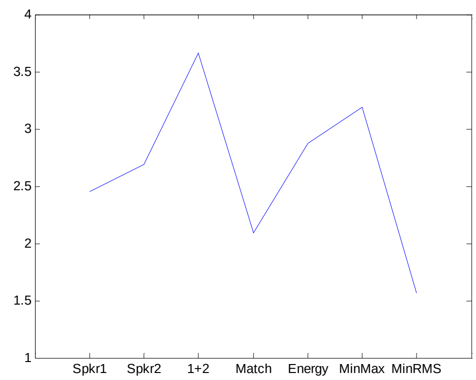

Figure 12 Frequency average of spatial standard deviation, frequency range 30 Hz – 65 Hz.

The properties of the different equalization strategies can be summarized as follows:

- Single speaker: performance can be quite acceptable with careful placement, but full equalization should be attempted only very optimal conditions
- Sum of two speakers: usually improvement over single speakers, but when locations are not optimal, full equalization should be attempted at only very low frequencies.
- Matched responses: some deterioration of low-frequency field distribution, efficient in controlling higher frequencies, easy to implement
- Minimum energy: degrades sound field distribution at lowest frequencies, some success in controlling higher modes, but performance does not justify computational and measurement complexity
- Minimizing maximum or RMS spatial deviation: efficient in controlling the distribution in the sparsely modal region

The results indicate clearly that it is possible to reduce the spatial variation of the sound field through the use of non-identical responses for each speaker. However, the initial results also indicate that the optimization methods used are rather sensitive to noise, so further development of the optimization criteria using e.g. modal decomposition or other more robust system estimation for preprocessing the measured data before optimization would be interesting.

4. REFERENCES

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