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THE SOUND FIELD DISTRIBUTION PROBLEM OF ROOM EQUALIZATION

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0. ABSTRACT

The success of room equalizing depends on the solution of several different problems. A central problem is the sensitivity of the system to changes in the position of the sound source as well as the receiver. The position of the sound source and the receiver is crucial to the performance of a room equalizing system. Using computer simulations and physical measurements, the distribution of the low frequency sound field is analyzed. A special analysis was caried out to explore the use of 2 microphone positions around the listening position. This lead to the conclusion, that the position sesitivity is less when using 2 microphone positions compared to 1 microphone position.

1. INTRODUCTION

The purpose of room equalization is to minimize the influence of the listening room when reproducing sound through loudspeakers. Several different approaches have been reported, including manually tuned graphic and parametric equalizers. These approaches, however popular to the common user, are not sufficient in any way to solve the complex problem of room equalizing. Automatic tuning of a graphic equalizer is reported in [1], but this does not circumvent the limitations of graphic and parametric equalizers, namely the inability to produce any causal transfer function.

In automatic systems, the basic approach is based on measuring the transfer function from the loudspeaker to an omnidirectional microphone, placed on the prefered listening position. Theoretically the transfer function of the ideal room equalizing filter can be calculated as the inverse of the measured transfer function. A transfer function measured in a room, however, is under normal conditions related to a non minimum phase system [2]. The inverse of a non minimum phase system can not be both stable and causal because the inverse system contains poles outside the unit circle. A common solution to this problem involves the design of a minimum phase equalizing filter that corrects the amplitude characteristic, and leaves part of the phase characteristic uncorrected. The use of modern digital signal processing enables a causal and stable approximation to the ideal equalizing filter, which corrects the phase characteristic to a certain extent, as well as correcting the amplitude characteristic. Doing this, a pure delay must be accepted beside the acoustic delay due to the fact that a pure delay can not be removed by a causal system.

The type of filter used for room equalization commonly is Finite Impulse Response (FIR). The use of FIR filters ensures a stable realization. Most widely used methods for deriving the coefficients of the filter involve optimization using the criterion of Least Mean Squares (LMS). A LMS solution can be found in several different ways, but the straight forward algebraic solution involves inversion of a matrix. When the order of the FIR filter increases this inversion becomes unpractical and other approaches are used.

The references [3], [4], [5] and [6] describe room equalization systems based on adaptive filters. Traditionally an adaptive filter is created in an iterative process, in which the coefficients of the FIR filter converge towards the optimum filter by means of a performance function. Normally the LMS criterion is applied in this process, but in acoustic applications the LMS criterion is not satisfactory because it is not psychoacoustically optimal [5]. Due to this fact some systems use another criterion in an attempt to utilize the present psychoacoustic knowledge [5].

Most reported methods for designing an equalizing filter are based on fixed positions of loudspeaker and microphone. When the position of the loudspeaker or the microphone is changed, the transfer function from the loudspeaker to the microphone also changes. If the equalizing filter is not changed according to the change in position, the effects can be severe colouring, preechoes, etc. The placement of the loudspeaker does probably not change very often, and the equalizing filter can be adjusted after every movement. But the position of the receiver/listener does indeed change, e.g. when the listener moves his/her head. In addition the ears of the listener are not placed in the same position but separated about 18 cm.

Elliott et al [7] reports about "Multiple-Point Equalization in a Room",. which is an equalizing system that is based on measuring the transfer functions from the loudspeaker to 4 different microphones. The initial approach in [7] consists of an equalizing filter constructed from the measurement at only one microphone position, and the result is evaluated at all 4 microphone positions. At the first microphone position, the amplitude characteristic was approximately perfect after equalization, but at the 3 other microphones the results were worse after equalization compared to the initial situation without equalizing filter [7]. Based on these results a new attempt was made [7] in which the transfer functions to all 4 microphones were used to design the equalizing filter. This leads to some improvement at all 4 microphones, but the results at any of the 4 microphones were far away from a perfectly flat amplitude characteristic. This is a result of the sensitivity to change in the position of the receiver.

The importance of the loudspeaker position is well demonstrated in [8], [9] and [10], where different effects, caused by moving the loudspeaker, are studied. From these studies it can be seen, that some placements near reflecting planes are more ideal than others.

From this it can be seen that the sensitivity to changes in position of sound source and receiver is a central problem of designing a room equalization system.

2. METHODS

The aim of this work is to achieve new knowledge about the sensitivity problem described above. The frequency range of interest is set to the low frequency range from 10 Hz to 500 Hz. The strategy is to analyze the sound field. First of all this is done by analyzing the sound pressure distribution of the low frequency sound field in one listening room. Following this, an analysis of the sensitivity to changes in the position of the receiver is performed. A major part of the analysis is carried out using a room simulator programme, developed on the principle of mirror images. Using this tool it is possible to handle a large number of transfer functions. To verify the results from the simulations, a series of physical measurements was carried out in an IEC 268-13 standard listening room. Finally the position sensitivity is analysed when using 2 microphones near the prefered listening position.

2.1 Room simulator programme. As a tool used in the rest of the work, a room simulator programme was developed. The programme was written in the programming language ANSI C. Execution time varies depending on the length of the impulse response to be calculated and on the dimensions of the room. The programme was run on an IBM compatible PC, which is based on the Intel Pentium processor (150 MHz). Using this PC, one simulation lasted 5 minutes (impulse response length = 400 ms, room shown in figure 2). The impulse response is calculated at discrete times, corresponding to time sampling at 4 kHz sampling rate.

The basic principle of this room simulator programme involves modelling a room by an infinite number of virtual sound sources, each operating in free field. The impulse response, from the physical source to the receiver, is then calculated by summing up all the contributions from each of these virtual sources. The positions of the virtual sound sources can be found as iterative mirror images of the physical sound source. This principle is known as the image method or the principle of mirror images.

A number of restrictions and approximations form the basis of a simplification of the programme:

- The Programme does only apply to rectangular rooms.
- The loudspeaker is modeled as a point source, i.e. omnidirectional characteristic and constant sound pressure at all frequencies when operating in free field.
- No furnitures can be modeled and included in the simulation.
- The reflection factors of the surfaces of the room are independent of the angle of incidence and of the frequency. Different reflection factors in different frequency subbands are, however, possible by multiple runs of the programme.
- The reflection factors are real valued and constant over the entire surface of a wall, floor or ceiling, but the reflection factors can be specified independently for each of the 6 surfaces of a rectangular room.
- The sound absorption in the air is neglected.
- The number of included virtual sources are finite.

Time sampling is performed by calculating the values of the output at times, which are an integer multiple of the sampling period. Together with a 10. order Butterworth lowpass filter, this eliminates the time quantizing error, which is introduced in [11], when the delay is rounded off to the nearest integer multiple of the sampling period. Transfer functions are calculated using a 4096 point FFT to transform the calculated impulse responses to frequency domain. The characteristic technique, used in the developed programme, is the continuous time simulation of rooms, which lead to the programme title: "ConSim". Additional information about the simulation program, "ConSim", is avaible from [12].

2.2 Sound pressure distribution. The room dimensions and the reflection factors, in this analysis, are chosen to approximate the IEC standard listening room, which is to be used later during the measurements. Figure 1 shows these properties of the simulated room. The reverberation time, T_{60}, of this simulated room is estimated to 315.4 ms by the formula of Sabine [13].

The first step in analyzing the influence of the sound pressure distribution is carried out by simulating the transfer functions from 2 different loudspeaker positions to 17 listening positions along the middle of the room. The listening height was constantly 1.10 m, which is a typical height of listening, when a person is seated. Figure 2 shows the 2 different loudspeaker positions. The first 200 ms of the impulse response was calculated using ConSim. The length of the impulse response corresponds to a frequency resolution of approximately 4 Hz [3]. Care was taken when choosing the length of the impulse response, making sure that the response has decreased to a satisfactory small value before truncating.

All virtual sound sources, within a distance corresponding to a delay of 200 ms, are included in the calculations. In this case, the number of virtual sound sources were 28152. The pressure distribution is then analyzed at several frequencies using a 4096 point FFT to transform the impulse responses to frequency domain.

A more extensive analysis of the sound pressure distribution was performed as well. The transfer functions from loudspeaker position A to 81 listening positions were simulated. The listening positions were distributed uniformly in a plane 1.10 m above the floor, which can be seen in figure 3. These 81 transfer functions made it possible to achieve a general view of the pressure distribution in a rectangular room at different frequencies.

2.3 Position sensitivity based on simulations. In this analysis of the position sensitivity, "ConSim" was used to simulate the same room as in section 2.2, which is shown in figure 2. The loudspeaker was fixed in position A. The position sensitivity is analyzed by examining the effects of changing the listening position away from a reference position. The reference listening position forms the third angle in an equilateral triangle, of which the loudspeaker and a hypothetical symmetrical placed loudspeaker form the two other angles. The height of the reference listening position was 1.10 m. The listening position was changed 10 cm in each of the directions: up/down, left/right and forward/backward. Figure 4 contains a table of both reference listening position as well as relative listening positions.

The length of the calculated impulse response was 200 ms. After calculation of the 7 transfer functions, the 6 positions away from the reference position were normalized according to the reference position, i.e. the amplitude characteristic, simulated at the reference position, was subtracted on a logarithmic scale (dB) from each of the 6 other amplitude characteristics. This operation is similar to inserting an equalizer, which is designed to correct the amplitude characteristic in the reference position, in the signal path. In other words, this operation enables an evaluation of room equalization systems, which are based on measuring the transfer function from the loudspeaker to a fixed microphone position.

2.4 Position sensitivity based on measurements. To verify the results in section 2.3, a series of physical measurements were carried out in an IEC 268-13 standard listening room. This room has similar dimensions and approximately equal reverberation time to the room simulated in section 2.3. The loudspeaker placement was fixed at a position 1 m from the nearest end wall and 1 m from the nearest side wall. The loudspeaker position and the reference microphone position are shown in figure 5, at which the dimensions of the used IEC standard listening room are shown as well.

The loudspeaker was a KEF model 107 - type SP3059 placed directly on the floor. The used microphone was a pressure calibrated measure microphone, Brüel & Kjær 4166, placed vertically as shown in figure 6. The height of microphone, at the reference position was 1.10 m above the floor. In the frequency range of interest this microphone has an omnidirectional characteristic.

The position of the microphone was changed 10 cm in each of the directions: up/down, left/right and forward/backward. Including the reference position the total number of microphone positions were 27. Figure 7 shows the positions of these 27 listening positions. Finally the position of the microphone was moved 30 cm and 60 cm away from the reference position in the direction left/right to set the perspective of the measurements.

The transfer functions were measured using a MLSSA measuring system, installed in an IBM compatible PC. The MLSSA system uses a Maximum Lenght Sequence (MLS) as test signal, and uses crosscorrelation to calculate the impulse response. To prevent time aliasing, the length of the MLS was set to 1024 ms, which is far beyond the reverberation time of the room. The sampling rate was set to 4 kHz, and the on board programmable antialiasing filter was set to Butterworth characteristic, while the cut-off frequency was set to 1 kHz. A time window of 200 ms was used to reduce influence of noise. This corresponds to a frequency resolution of 4.45 Hz [3]. The transformation to frequency domain was carried out using a 4096 point FFT.

2.5 Position sensitivity using 2 microphone positions. As mentioned in section 1,"Multiple-Point Equalization in a Room" by Elliott et al [7] is a method, where 2 or more microphone positions are selected around the prefered listening position(s). This method had shown some improvements compared to using only one microphone position. From Elliot et al, it can bee seen that using more microphones enables some improvement at a number of listening positions, while using only one microphone position lead to nearly perfect equalization at that particular microphone postion and both improvements and deteriorations at other positions.

Considering this, it is interesting to investigate the position sensitivity of 2 microphones near the listening position moved in parallel 10 cm both in the right and left directions relative to the reference positions of the two microphones. This should then be compared to 1 central placed microphone, again moved 10 cm left/right. The separation of the 2 microphones is chosen to 0,50 m, which is reasonable considering the frequency range of interest. Figure 8 shows the simulated listening room, which is identical to the one used in section 2.1 and 2.2, and the postion of the loudspeaker. Also included in this figure are the reference positions of both the 2 microphones (LEFT MIC and RIGHT MIC) moved in parallel and the central placed microphone (CENTER MIC). Short vertical lines indicate the movements of +/- 10 cm relative to each microphone reference position. The position sensitivity of 1 microphone is also investigated around the LEFT MIC and RIGHT MIC reference positions.

"ConSim" was then used to simulate the first 400 ms of the impulse response at all the different microphone positions. This corresponds to a frequency resolution of approximately 2 Hz [3]. A 4096 point FFT was used to transform the time signal to frequency domain. In the case of 2 microphones, the 2 individual amplitude responses are averaged by summing the squares of the individual amplitude responses, dividing by 2 and then calculating the squareroot. This is done at one frequency at a time. This means, that only 1 single amplitude response results from 1 simulation of the sound pressures at the 2 microphones. After moving the 2 microphones, in parallel, this procedure is repeated.

Similar to the procedure, described in section 2.2, each simulation result from positions away from the microphone reference position, are normalized acording to the simulation result from the reference position. I.e. the amplitude response simulated in the reference position is subtracted from the amplitude responses simulated in the +10 cm and -10 cm positions.

3 RESULTS

The results of this work are grouped into 4 logical sections, namely results concerning the sound pressure distribution, the results concerning position sensitivity, based on simulations and measurements and finally a section containing the results of using 2 microphone positions compared to 1. The sound pressure distribution has been analyzed, and knowledge about the unevenness of the sound pressure at different listening positions was achieved. The position sensitivity has been analyzed using both simulations and physical measurements, and results are available, which enable an evaluation of room equalization systems, based on 1 fixed receiver position. In addition to this, it is possible to evaluate the effect of using 2 microphone positions near the prefered listening position compared to 1 microphone position.

3.1 Sound pressure distribution. The sound pressure distribution was first examined at 17 positions along the middle of the room. 22 Hz is frequency of the standing wave, which is characterised in that, half of the wave length is equal to the length of the simulated room. Figure 9 shows the amplitude of the sound pressure at the 17 positions along the middle of the room. Both the results concerning loudspeaker position A and B, referring to figure 2, are included in figure 9. The amplitude of the sound pressure is normalized to the maximum amplitude, which is found at the nearest end wall relative to the loudspeaker.

Figure 9 clearly shows the standing wave at 22 Hz, when the loudspeaker is at position A as well as at position B. As expected, the amplitude is minimum, i.e. includes a notch, half through the room, i.e. 3.90 m from either end wall. At this point, the amplitude is between 16 and 18 dB below the level at the nearest end wall. This indicates the extent of the unevenness. It should be noted that the greatest changes are centered around the notch of the amplitude distribution. i.e. small changes in position lead to the greatest changes in sound pressure amplitude, when the reference position is placed near the notch, at the middle of the room. No significant difference is noted between loudspeaker position A and B.

At 44 Hz the standing wave appears in figure 10, as expected, but in this case the difference between loudspeaker position A and B becomes significant. Generally, the sound pressure amplitude, corresponding to position B, is approximately 7 dB below the amplitude, corresponding to position A. At the second minimum, i.e. 6 m from the end wall nearest to the loudspeaker, the difference between position A and B reaches 17 dB.

To achieve a general view of the sound pressure distribution along the middle of the room at several frequencies, a graphic presentation is used, in which frequency and geographical placement, along the middle of the room, are combined. The geographical placement along the middle of the room is measured in the direction: up to the left, in figure 11, and the frequency increases in the direction: up to the right. From figure 11, it is possible to evaluate the unevenness of the sound pressure distribution at frequencies from 10.7 Hz to 96.7 Hz in steps of 0.977 Hz. The standing waves at 22 Hz, 44 Hz and at 66 Hz appear clearly in figure 11. The unevenness tends to be more significant at frequencies near the normal-modes, i.e. at 22 Hz, 44 Hz, and 66 Hz but not at 88 Hz. Generally the unevenness tends to decrease at higher frequencies.
Figures 12 to 14 show the results from simulating the transfer functions from a loudspeaker to 81 different receiver positions. In figure 12 the standing wave at 22 Hz is verified to vary only in the direction: forward/backward, and not in the direction: left/right. In figure 13 the standing wave at 66 Hz is

generally verified to have a similar property, but the amplitude is increasing towards the corner, at which the loudspeaker is placed. The real unevenness of the sound pressure distribution is best shown in figure 14, in which the complex standing wave at 60.4 Hz is recognized (mode 2 - 1 - 0).

3.2 Position sensitivity based on simulations. The amplitude of the sound pressure at the reference position, is shown in figure 15. This figure shows that the amplitude characteristic is dominated by a number of peaks and notches. The notches seem to be generally more narrow than the peaks.

After calculation of the deviation from the reference position, limiting curves of the deviation were made, based on comparison of the 6 deviation curves, corresponding to the 6 positions away from the reference position. I.e. these deviations are caused by movement of the receiver position, away from the reference position. The limiting curves are shown in figure 16. The typical deviation in the frequency range of interest is approximately 3 dB above or below the reference, but several peaks, in both directions, of the limiting curves are observed. The diviation at these peaks is from 15 dB to 30 dB. The worst deviations tend to be observed at frequencies, where the reference amplitude includes a notch.

3.3 Position sensitivity based on measurements. A series of physical measurements were performed to verify the results, achieved from the simulations in section 3.2. The presentation of the results is chosen to be similar to that in section 3.2, i.e. presentation of the amplitude characteristic at the microphone reference position, and limiting curves of the deviation from that reference position. The used IEC standard listening room has similar dimensions and approximately the same reverberation time as the room simulated in section 2.2. The loudspeaker position, L1, is placed near position A, refering to figure 2 and figure 5. This is the reason, why figure 15 and figure 16 are used, when comparing simulations and measurements.

Figure 17 shows the sound pressure amplitude characteristic, at the reference position, and it should be noted that 0 dB, in figure 17, corresponds to 20 μPa/V. The voltage is referring to the terminals of the loudspeaker. The RMS value of the sound pressure, at the reference position, when continuously leading the MLS signal to the loudspeaker, was measured to 74.7 dB re 20 μPa. When comparing figure 17 and figure 15, good agreement is found, especially at low frequencies. Note the peaks at approximately 22 Hz, 42 Hz and 66 Hz at both figures. The characteristic notch at approximately 50 Hz is also found at both figures. The highpass properties of the physical loudspeaker are clearly notable below approximately 40 Hz.

Figure 18 shows the limiting curves of the deviation from the amplitude at the reference position, and the pattern is comparable with figure 16, i.e. the deviation becomes more fluctuating at higher frequencies. The most important similarity of figure 18 and figure 16 is the tendency that the worst deviations are found at frequencies, where the amplitude characteristic, at the reference position, includes a notch.

To set the perspective, measurements were performed at 30 cm and 60 cm distance from the reference position, in the direction: left/right. Figure 19 shows the deviation from the amplitude at the reference position when the microphone is moved 30 cm to the right and to the left. This shows the seriousness of the sensitivity problem because deviation exceeds 10 dB in relatively wide areas of the frequency range. At 60 cm distance, figure 20, the results only confirm this property of the sensitivity problem, while several peaks, of 20 dB or more, are observed.

3.4 Position sensitivity using 2 microphone positions. The simulated amplitude characteristic at the microphone reference position "CENTER MIC" is presented in figure 21. "CENTER MIC" is defined in figure 8. The amplitude characteristic is dominated by a number of peaks and notches. Figure 22 shows the deviations caused by moving 1 microphone 10 cm to the left and to the right relative to "CENTER MIC". It should be noted, that the curves in figure 22 are not limiting curves, but 2 actual deviation curves - one from the left movement and one from the right movement. Below 100 Hz deviations of 5 dB are found, and a very narrow peak of 13 dB is discovered. Above 100 Hz levels of 10 to 15 dB are found a number of times. These results agrees with the results found in section 3.2 and 3.3 when the different impulse response length (time window) is considered.

The amplitude characteristic at the microphone reference position "RIGHT MIC" is found in figure 23. From this figure it can bee see, that some similarities are found when comparing figure 21 and 23. The deviations caused by moving 1 microphone 10 cm to the left and to the right relative to this reference postion, "RIGHT MIC", are found in figure 24. Below 100 Hz deviations tends to be a bit lower than in figure 22, i.e. levels of 4 dB or more are found a number of times. Above 100 Hz levels of 5 to 10 dB are found a number of times, which is a lower level than in figure 22.

Figure 25 shows the simulated amplitude characteristic at the microphone reference position "LEFT MIC". Again some similarities are found, when comparing to figure 21 and figure 23. When moving 1 microphone 10 cm to the left and to the right relative to the microphone reference postion "LEFT MIC", deviations below 100 Hz tend to be equal or less compared to figure 22. This can be seen from figure 26. The levels are kept below 5 dB in the frequency range below 100 Hz with only 2 narrow exceptions of 7 dB. Above 100 Hz the deviations found are compareble to the levels in figure 22, while levels of 10 to 20 dB are found a number of times.

The average of "LEFT MIC" and "RIGHT MIC", calculated as described in section 2.5, is found in figure 27. This average amplitude characteristic is characterised by some of the same proberties as the amplitude characteristics in figure 21, 23 and 25. But the averaging proces (geometric) leads to a bit more smooth curve, which can be seen, when comparing figure 21, 23, 25 and 27. Figure 28 presents the deviations caused by moving 2 microphones in parallel 10 cm to the left and to the right. As can be seen in this figure the level of the deviations below 100 Hz is reduced to approximately 1.6 dB. Above 100 Hz the general level is approximately the same as below 100 Hz with only a few exeptions. When coparing figure 28 to any of the figures 22, 24 or 26 it is clear, that the general level of deviations caused by movements of +/- 10 cm is reduced significantly by employing 2 microphone positions compared to 1.

4 DISCUSSION

The performance of "ConSim" can be evaluated in several different ways, but the most direct method is to compare the simulated amplitude characteristic to a measured one. This is done in section 3.3, and the results were in good agreement at lower frequencies. Less or no agreement, at higher frequencies, can be explained by a number of circumstances. E.g. the loudspeaker position, used in the measurements, differed about 25 cm in distance to the nearest 2 side walls compared to position A, which was used in the simulations. The reflection factors, used in ConSim, were intuitively set to values, which correspond to a reverberation time approximately equal to that in the physical IEC standard listening room. In addition to this, approximations were made when developing "ConSim", as described in section 2.1. It should, however, be noted that the intention never was to perform a simulation of the exact same room as the physical IEC room. The intention was to simulate some room with approximately the same properties.

Results in section 3.1 strongly agree with the theoretical knowledge about normal-modes, which suggests that "ConSim" is able to simulate the essential properties of the sound field. Based on this consideration "ConSim" is concluded to be able to simulate essential properties of a transfer function in a rectangular room. These facts also suggest that the restrictions and approximations in "ConSim" do not prevent "ConSim" from providing usable and accurate information.

The results in section 3.2 indicate that position sensitivity is primarily related to notches of the standing waves. This makes sense because the steepest areas of the sound pressure distribution of a standing wave are located around minima points, i.e. notches. Combining this result with the tendency that the unevenness is decreasing at higher frequencies makes an important point: avoid listening positions near minima points of the standing waves, which possess the lowest frequencies. In other words, a listener should avoid selecting a listening position near the minima points of the first few standing waves. A similar advice applies to the loudspeaker placement as well, while the transmission from a minimum point of sound pressure was found to be less effective in section 3.1. Here it should be noted that loudspeaker position B is placed near the first minimum of the standing wave at 44 Hz. The major irregularities of the sound pressure amplitude distribution are found at the frequencies of the standing waves in a room. This observation shows the significant influence of the standing waves at low frequencies.

A remarkable finding in sections 3.2 and 3.3 is that the worst and most significant position sensitivity is found at frequencies, at which the sound pressure amplitude at the reference position includes a notch. The practical impact of this is that notches should not be exactly equalized using a filter, which includes a peak at the same frequency, as this will lead to serious problems in practical use. Advicing not to perform exact equalization of notches is not new, but the arguments are usually of the psychoacoustic type [5]. Now the position sensitivity is an argument, not to perform exact equalization of deep notches, as well.

This analysis of the position sensitivity problem explains the unsatisfactory results achieved by Elliott et al [7], when evaluating the resulting transfer functions to 4 different microphone positions, after designing an equalization filter from the transfer function to only one of the microphone positions. Elliott et al tried to solve this problem by measuring the transfer function at several different listening positions, and to take all of them into account, when designing the room equalization filter. This principle prevents equalization of deep notches, provided that the notches are not common to the selected listening positions.

An important fact consists of the agreement between the results achieved in section 3.2, based on simulations, and the results achieved in section 3.3, based on physical measurements. In both sections, position sensitivity was found to be related to notches, and position sensitivity was proved to be a significant problem, which can not be neglected when designing room equalization systems. In other words, future development of room equalization systems, based on only one microphone position, seems to be useless.

The significance of the achieved results is related to the development of room equalization systems, which offers an improvement in practical use. Such a system demands a wide degree of insensitivity to changes of the listening position, which could enable the listener to move around in the listening room as well as enable several listeners to experience the improvement simultaneously. This is found to be a demanding task, due to the serious effects caused by small movements of 10 cm, not to mention 30 cm, 60 cm or even longer distances.

One point in this matter is the results found in section 3.4, where the use of 2 microphone position is compared to the use of only 1 microphone position. This strongly indicate, that the avarage pressure from 2 microphone positions is less plagued by position sensitivity than the sound pressure picked up by only 1 microphone placed at either of the 2 positions or in the middel of the two. This means that sound pressure used to design the room equalization system is less influenced by choice of the microphone positions, when using 2 positions compared to 1 position. But after designing the room equalization system, the listener is still left with the fact, that the actual listening position choosen is significant to the experience due to the distribution of the sound field, which is not changed by employing 2 microphones during the measurements.

5 CONCLUSION

The conclusions of this work can be summarized in the following statements:

- The developed room simulation programme, "ConSim", was proved to be a valuable tool to handle a large number of transfer functions

- Good agreement was found between simulations and measurements.

- The listening position should not be placed near minima points of the standing waves, which are well spaced in frequency. This follows from the finding that position sensitivity is related to the minima points of standing waves.

- Significant differences in the amplitude of the sound pressure were found, depending on the listening position, due to significant unevenness of the amplitude distribution in a listening room.

- The major irregularities of the sound pressure amplitude distribution were found near the frequencies of the standing waves of the room.

- Notches should not be exactly equalized, as the worst and most significant position sensitivity was found at frequencies, where the amplitude characteristic at the reference listening position includes a notch.

- Position sensitivity can not be neglected, while significant changes occur because of 10 cm movements of the listening position. The situation is even worse at 30 cm and 60 cm distances.

- Future development of room equalization systems, based on only one microphone position, seems to be useless because of the position sensitivity problem.

- The position sensitivity of the avarage sound pressure at 2 microphone positions is less compared to 1 microphone position.

6. REFERENCES

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(length · width · height) = 7.80 m · 4.14 m · 3.00 m

Surface	R
Near end wall	0.9
Rear end wall	0.9
Right side wall	0.9
Left side wall	0.9
Floor	0.8
Ceiling	0.5

Figure 1: Dimensions and reflection factors, R, of the simulated IEC standard listening room. The near end wall is the nearest end wall relatively to the loudspeaker, and the left side wall is the side wall nearest to the loudspeaker.

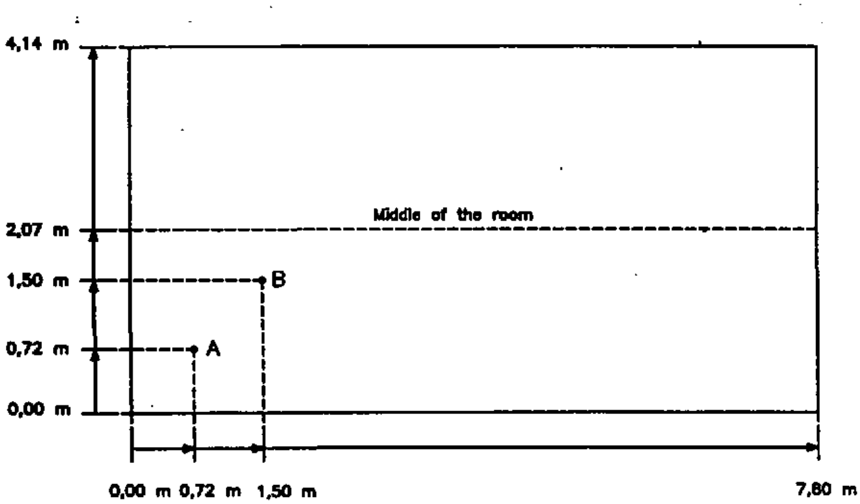

Figure 2: Ground plan of the simulated IEC standard listening room, and positions of loudspeaker placement A and B. The loudspeakers are placed 0.46 m above the floor. Note the indication of the middle of the room. The position along the middle of the room is measured from the end wall, which is depicted to the left.

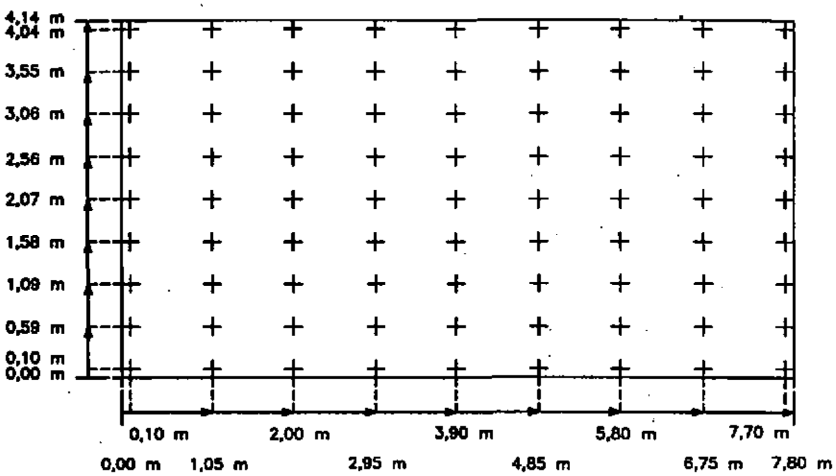

Figure 3: Positions of 81 different receiver points, depicted on the ground plan. Note that all points are placed 1.10 m above the floor.

Reference position (x,y,z) = (2.07m, 3.06m, 1.10m)

Position No.	Relative position		
	x [m]	y [m]	z [m]
1	0	0	0
2	0.1	0	0
3	-0.1	0	0
4	0	0.1	0
5	0	-0.1	0
6	0	0	0.1
7	0	0	-0.1

Figure 4: Relative positions of the receiver point, i.e. 6 receiver points, No. 2 - 7, away from the reference position, No. 1.

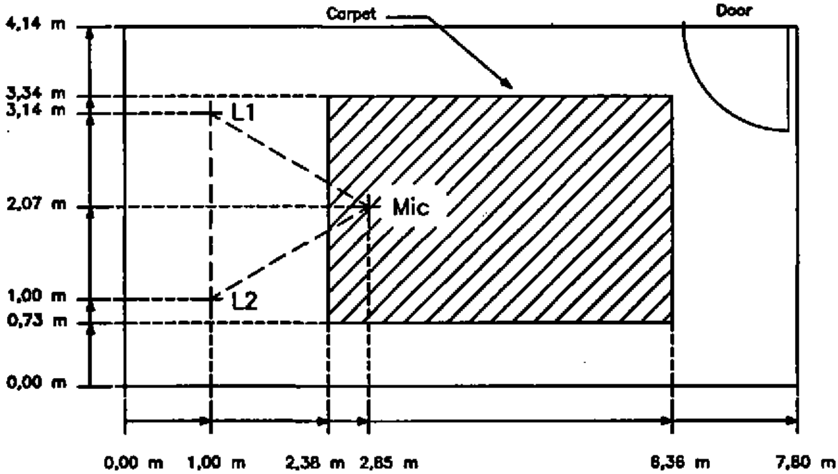

Figure 5: Ground plan of the IEC standard listening room, used in the measurements in section 2.4. "Mic" is the microphone, "L1" is the physical loudspeaker and "L2" is a hypothetical symmetrical placed loudspeaker. Note that "L1", "L2" and "Mic" form an equila-teral triangle. The height of the microphone is 1,10 m above the floor.

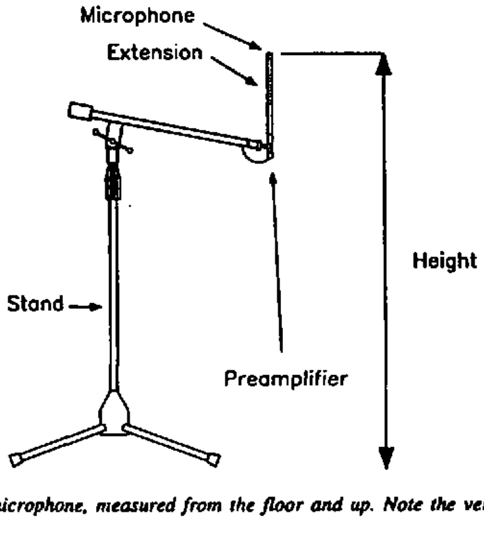

Figure 6: Height of the microphone, measured from the floor and up. Note the vertical placement of the microphone on the stand.

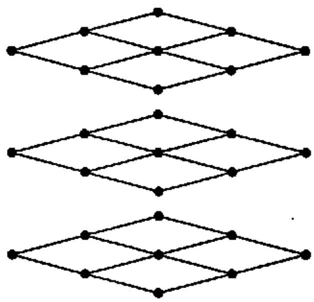

Figure 7: Placement of the 27 receiver points, used in the measurements in section 2.4. The 3 planes are located 1.00 m, 1.10 m and 1.20 m above the floor. The reference position is located in the center of the middle plane. The distance between 2 receiver points is 10 cm in each of the directions: up/down, left/right and forward/backward.

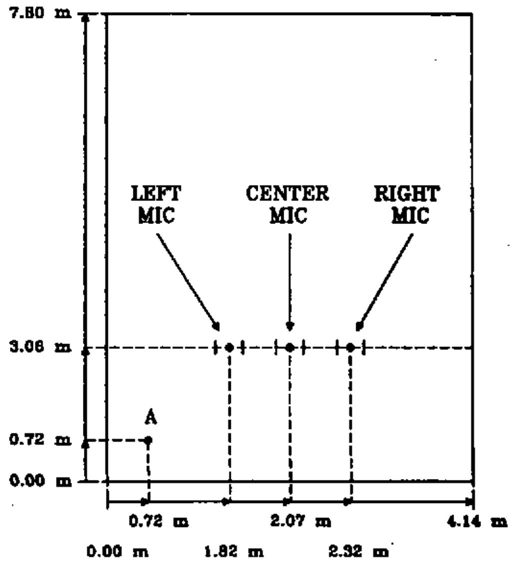

Figure 8: Ground plan of the simulated IEC standard listening room, and the position of loudspeaker placement A. The loudspeaker is placed 0.46 m above the floor. LEFT MIC and RIGHT MIC refers to the 2 microphone positions, which are moved in paralel. The short vertical lines indicate the movements of +/- 10 cm relative to each microphone reference position. All microphone positions are located 1.10 m above the floor.

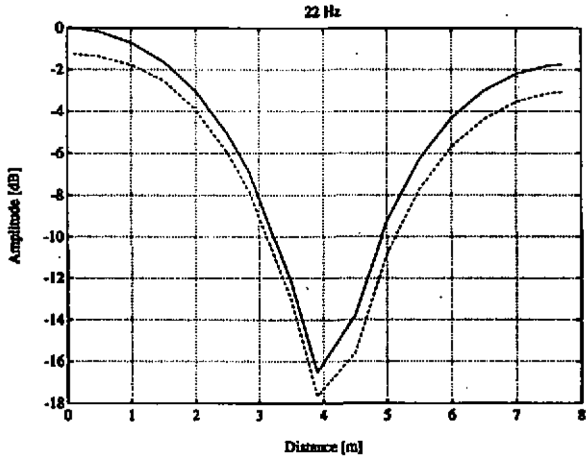

Figure 9: The amplitude of the sound pressure along the middle of the room at 22 Hz. The height is 1,10 m. The continuous line refers to loudspeaker position A, and the dashed line refers to position B. The sound pressure is normalized to the maximum amplitude, which is found at the nearest end wall.

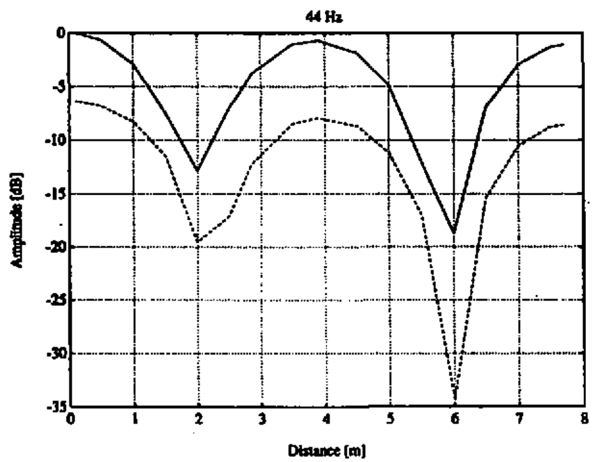

Figure 10: The amplitude along the middle of the room at 44 Hz. The continuous line refers to position A, and the dashed line refers to position B. The height is 1,10 m and the amplitude is normalized to the maximum amplitude.

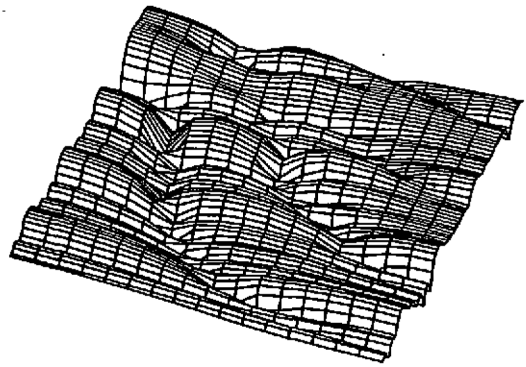

Figure 11: Amplitude of the sound pressure at different positions along the middle of the room and at different frequencies. The geographical placement along the middle of the room is measured in the direction: up to the left. The frequency increases in the direction: up to the right in steps of 0.977 Hz from 10.7 to 96.7 Hz.

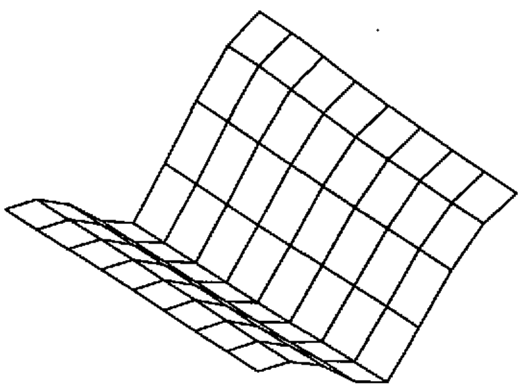

Figure 12: Amplitude of the sound pressure at 22 Hz, simulated at 81 different positions in the room. The length of the room is measured in the direction: up to the right, and the width is measured in the direction: up to the left. The height of all points is 1,10 m.

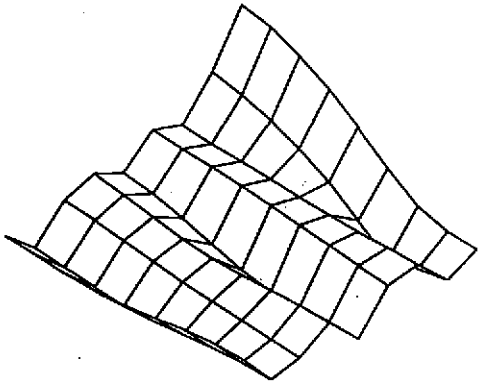

Figure 13: Amplitude of the sound pressure at 66 Hz, simulated at the same 81 positions as in figure 16. Note the increasing amplitude at the corner, which is depicted at the top of this figure. The loudspeaker is located in this corner.

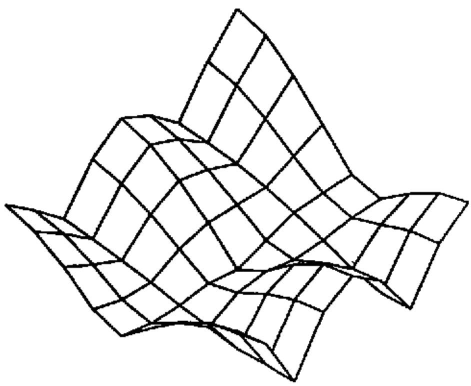

Figure 14: Amplitude of the sound pressure at 60.4 Hz, simulated at the same 81 positions as in figure 16. This is a complex standing wave of order 2 along the length of the room and of order 1 along the width of the room.

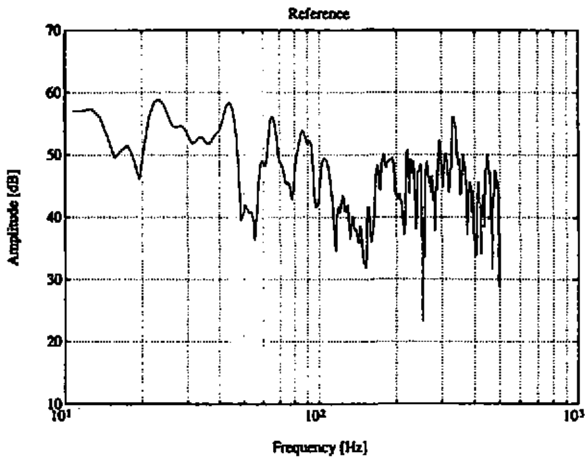

Figure 15: Amplitude of the sound pressure in the reference position. The loudspeaker is placed in position A.

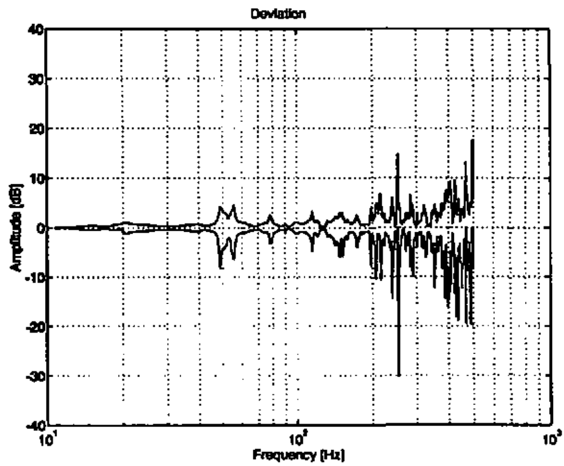

Figure 16: Limiting curves of deviation of the amplitude, caused by moving the receiver point. The loudspeaker placement is position A.

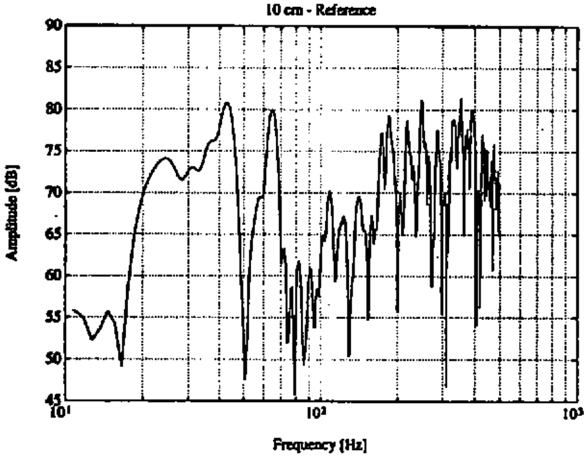

Figure 17: Amplitude of the sound pressure in the reference position of the physical IEC standard listening room.

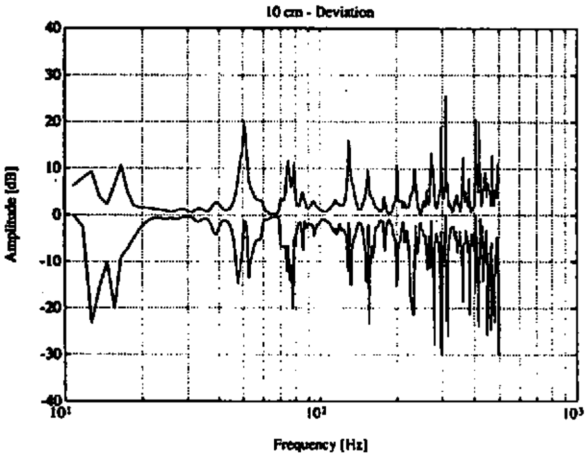

Figure 18: Limiting curves of deviation of the amplitude, caused by moving the receiver point. Measurements at 27 different microphone positions were performed to produce this figure.

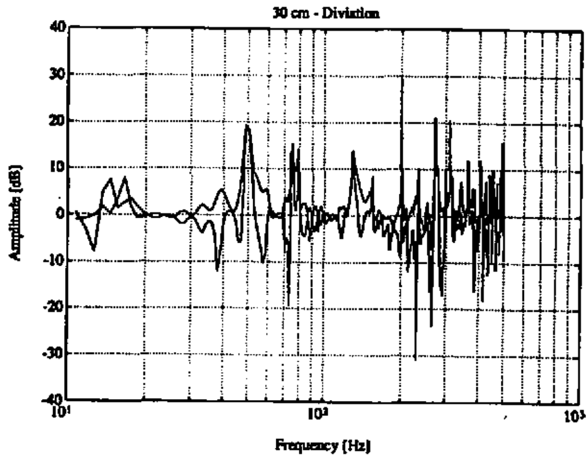

Figure 19: Curves of deviation of the amplitude, caused by moving the microphone position 30 cm to the left and to the right of the reference position.

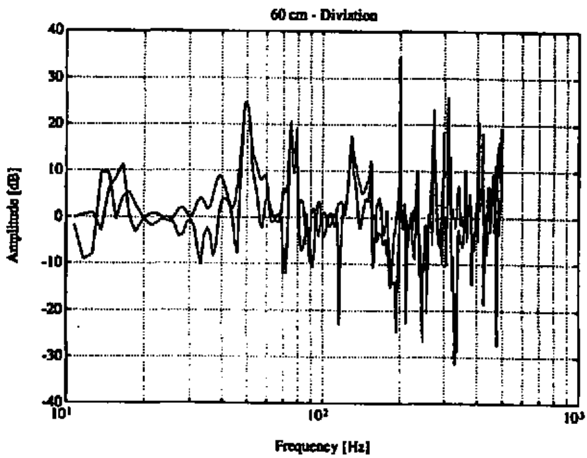

Figure 20: Curves of deviation of the amplitude, caused by moving the microphone position 60 cm to the left and to the right of the reference position.

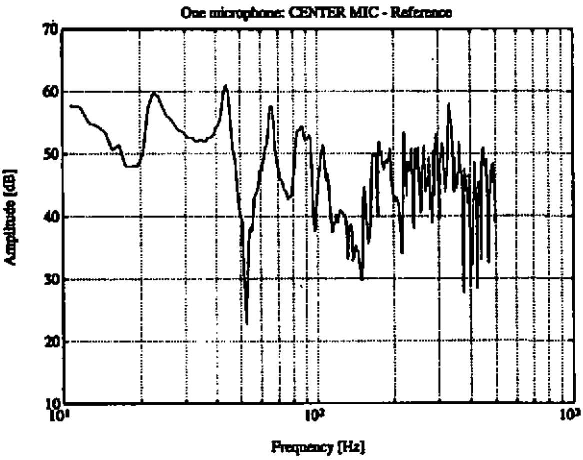

Figure 21: Amplitude of the sound pressure, one microphone position: CENTER MIC - reference position.

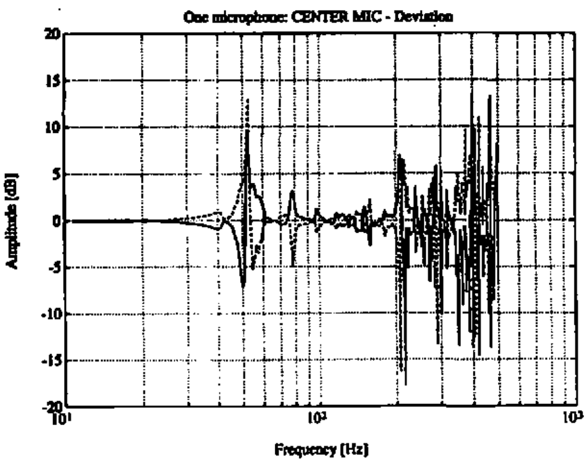

Figure 22: Curves of deviation of the amplitude, caused by moving one microphone position 10 cm to the left and to the right of the reference position: CENTER MIC.

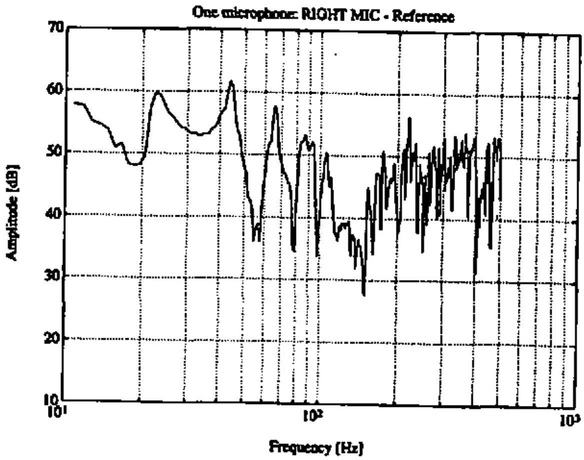

Figure 23: Amplitude of the sound pressure, one microphone position: RIGHT MIC - reference position.

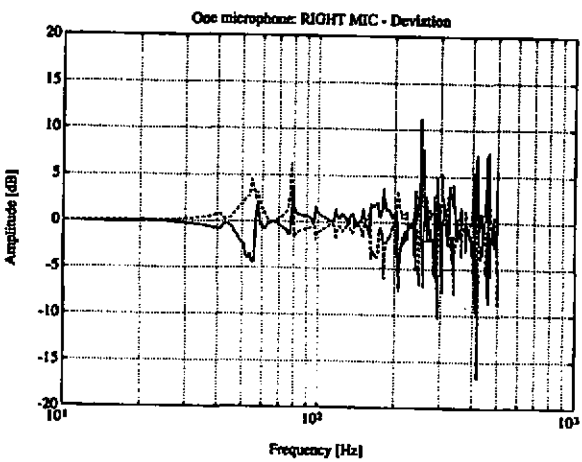

Figure 24: Curves of deviation of the amplitude, caused by moving one microphone position 10 cm to the left and to the right of the reference position: RIGHT MIC.

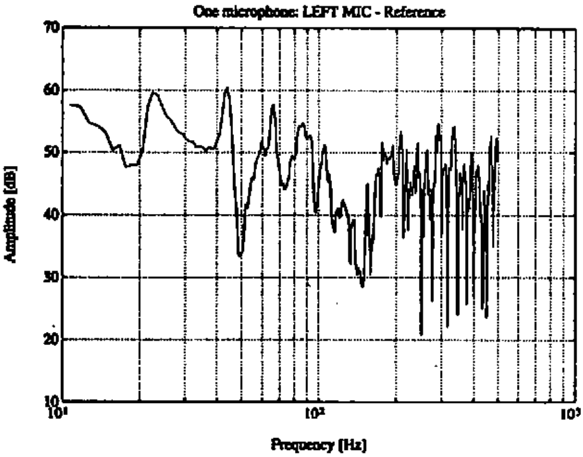

Figure 25: Amplitude of the sound pressure, one microphone position: LEFT MIC - reference position.

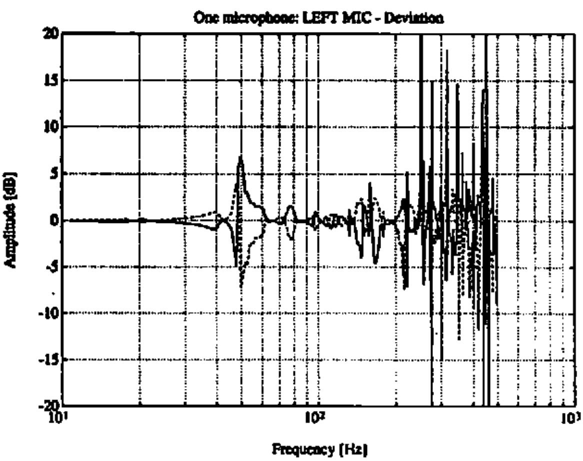

Figure 26: Curves of deviation of the amplitude, caused by moving one microphone position 10 cm to the left and to the right of the reference position: LEFT MIC.

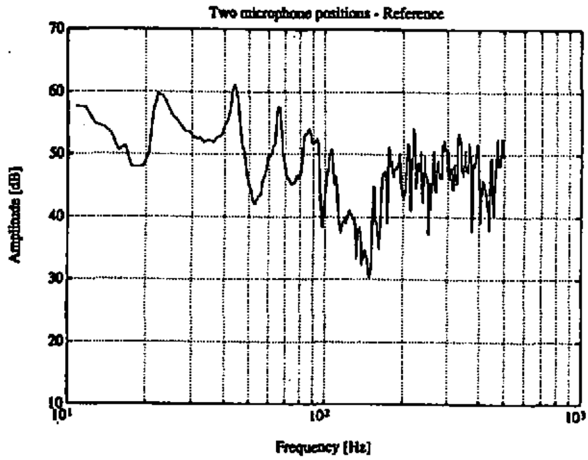

Figure 27: Amplitude of the sound pressure, average of two microphone positions: RIGHT MIC and LEFT MIC - reference positions.

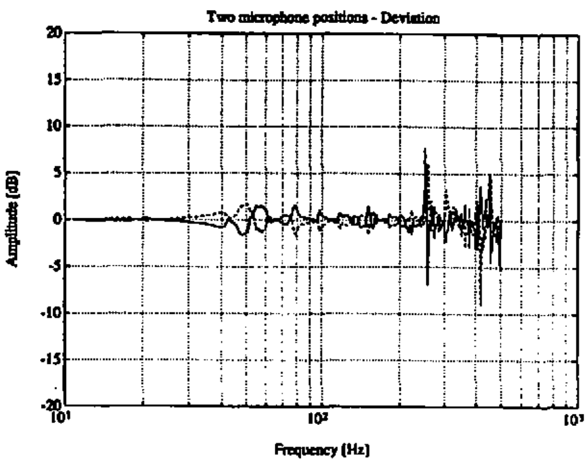

Figure 28: Curves of deviation of the amplitude, caused by moving two microphone positions 10 cm to the left and to the right of the reference positions: RIGHT MIC and LEFT MIC.