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Investigation on a possible future qualification of free field environments with impulse methods

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ABSTRACT

Currently, free field environments are qualified according to ISO 26101. This method relies on the inverse-square-law. Ideally, the sound pressure level decreases by 6 dB per doubling of the distance to a point source. Real free field environments however exhibit deviations to this ideal case caused by remaining reflections from the room walls and built-in devices. To detect the reflections at high frequencies, it is necessary to measure the sound pressure level with high spatial resolution, which results in a large experimental effort to verify the free field condition.

Therefore, it was investigated at PTB if transfer functions or impulse responses can be used to qualify free field environments. The measured transfer functions are analysed in a similar way as the current standard requires. For impulse responses, the different run-time can be used to separate direct and reflected sound. The ratio between these two components is then used as a criterion to test the free field. Results obtained by both methods are compared with results yielded by the current qualification method.

In this paper, the general idea of the measurement method, a possible practical implementation including a data analysis and some results obtained with this new method are discussed.

1. INTRODUCTION

Anechoic and hemi-anechoic environments are used for many different purposes such as sound power determinations, tests of equipment for measuring or producing sound, or psychoacoustic tests. Despite this vast field of application there is only one method standardised to test the acoustic properties of such environments [1]. It is based on the idea that –in free field– the sound pressure level has to drop by 6 dB when the distance to a point source is doubled. This method has been used for decades, and it used to be standardised in Annex A of [2]. Several changes have been made in the latest versions of this test which improved its significance considerably but the general concept has not been altered.

This so-called draw-away test has been investigated in great detail, see e.g. [3] to [6]. In addition to this there is also the idea of measuring the statistical spread of sound pressures in a room and calculate from this the reflection coefficient of the lining [7], [8]. This work was further on extended by applying numerical calculations to determine the properties of the lining [9]. Another approach was chosen in [10] where the room influence on the measured sound power level was used as the main criterion. This is reflected by the two-surface method as standardised in Annex B of [2].

Since qualifying an anechoic environment is basically about discriminating between reflected and direct sound it is straightforward to apply impulse response methods employing run-time effects. Such an approach is reported in [11], however only a qualitative comparison between different linings is presented there.

Based on the notion of a general applicability an inclusion of impulse response methods to qualify (hemi-) anechoic environments was originally intended when the development of ISO 26101 [1] started. Nevertheless, during the standardisation process it turned out that there was no real experience available and nobody developed such a method. It was therefore not possible to include an impulse response method in [1].

This contribution is therefore dedicated to the development of a general concept and a first implementation of an impulse method for qualifying (hemi-) anechoic environments. The important question is whether it is possible to define a method which gives results comparable to the existing standardised method and whether an application of this new method could be advantageous.

2. METHODS FOR ROOM QUALIFICATION

2.1. Current method for room qualification

For the current method [1], the decrease of the sound pressure level is measured in relation to the distance of a sound source. This is compared with the theoretically expected course of the free field, and deviations from the ideal conditions must be within the required tolerances. The sound pressure level should be measured at different microphone traverse paths within the room. They must be chosen so that they cover the area in the room intended for measurements, at least. [1] prescribes five straight paths, starting from the acoustic centre of the sound source, used in different directions. One or more sound sources approximating a point source over the frequency range of interest shall be used, and for the measurement either pure tone signals or a broadband signal are allowed.

There are several requirements for the test sound source. For example, it should be compact in design and be able to radiate sound power with great consistency. At the frequencies of interest, the sound pressure level should be sufficiently high, i.e. at all points along the microphone paths at least 6 dB above the noise level. It is also required that the sound source radiates omnidirectionally. A test method for the last-mentioned attribute is described in Annex B of [1]. Preferably, the sound source must be in the centre of the test environment. However, for some environments or rooms it could be necessary to measure with several different sound source locations.

According to [1], the sound pressure level drop needs to be inside a defined tolerance band. In principle, the tolerance band depends on the envisaged usage of the room. For sound power level determination, a tolerance band is given in ISO 3745, annex A [2]. Other standards may define other tolerance bands. A general purpose tolerance band is given in ISO 26101 table A1 [1].

Whether the measured sound pressure levels are inside the specified tolerance is checked as a function of the distance from the source for each frequency. The larger the distance to the source, the larger the deviations to the ideal free-field. So, at some distance the measured sound pressure levels will leave the specified tolerance range. The last microphone position before this happens is the maximum distance from the source for which free-field conditions are observed. This is the test result, a maximum distance from the source r_{max} as a function of frequency.

2.2. Transfer function method

The transfer function method relies on the measured transfer function between the source and the microphone. Different transfer functions are measured on a path, which starts at the source and ends at the lining of the walls. Stepped measurements are performed. Therefore, the microphone does not move while the measurement of the transfer function is in progress.

In post processing, the measured transfer functions L_{meas} are compared to a reference measurement L_{ref}. The measurement nearest to the source is the reference measurement. For each frequency, the corrected difference L_{corr} of both transfer functions is calculated by equation (1).

$$L_{corr} = L_{ref} - L_{meas} - L_{air} - 20 \log\left(\frac{l_1}{l_2}\right) \text{dB} \qquad (1)$$

The different distances during the measurements l_1 and l_2, as well as a correction of the air absorption according to ISO 9316-1 [3], are included in the calculation of L_{corr}.

Since the corrected difference L_{corr} is physically identical to a sound pressure level drop it can be used to calculate an r_{max} as a function of frequency in analogy to 2.1.

2.3. Impulse response method

The impulse response method uses the measured impulse response between source and microphone. The impulse responses are measured on each path at different distances to the source. As for the transfer function measurement, the microphone is at rest for each impulse response measurement.

In post processing, the impulse response is divided into direct sound pressure p_d, reflected sound pressure p_r and background noise. With the direct and reflected sound pressures a new quantity

$$L_{\text{ref}} - L_{\text{dir}} = 20 \cdot \log_{10}\left(\frac{p_r}{p_d}\right)\text{dB} \qquad (2)$$

is calculated by equation (2), which is then used to evaluate the quality of the free field at the measurement position.

$L_{\text{ref}} - L_{\text{dir}}$ can be mathematically converted into the current quantity for the qualification of free field environments. The current qualification is based on standing waves and therefore, on the addition and subtraction of the direct and the reflected sound pressure. The maximum and minimum sound pressure levels on a measurement path are defined in equation (3) and (4).

$$L_{\max} = 20 \cdot \log_{10}\left(\frac{p_d + p_r}{p_0}\right)\text{dB} \qquad (3)$$

$$L_{\min} = 20 \cdot \log_{10}\left(\frac{p_d - p_r}{p_0}\right)\text{dB} \qquad (4)$$

The difference between $L_{\max}$ and $L_{\min}$ can be considered to be identical to the width of the tolerance band used for the current test.

It can be converted into the newly defined quantity $L_{\text{ref}} - L_{\text{dir}}$ by equation 5.

$$L_{\text{ref}} - L_{\text{dir}} = 20 \cdot \log_{10}\left(\frac{\left(10^{\frac{L_{\max}-L_{\min}}{20\text{ dB}}} - 1\right)}{\left(10^{\frac{L_{\max}-L_{\min}}{20\text{ dB}}} + 1\right)}\right)\text{dB} \qquad (5)$$

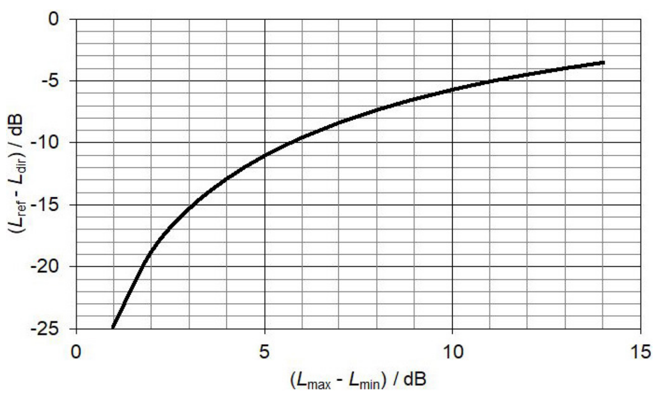

Figure 1. Difference between reflected and direct sound as a function of the width of the tolerance range given in ISO 26101.

With equation (5), it is possible to convert the tolerance band of ISO 26101 [1] into a tolerance for the newly developed impulse response method.

This new tolerance is not a band, but a tolerance line. Every datapoint underneath the tolerance line does fulfil the free field condition, every datapoint above the tolerance line does not fulfil the condition for a sufficient free field. An $r_{\max}$ can be calculated similar to the method mentioned in 2.1.

3. PRACTICAL IMPLEMENTATION

3.1. The test room

All measurements for this contribution were performed in the hemi-anechoic room at PTB Braunschweig. According to the guidelines in [1], three traverse paths were installed in the room. Always starting from the centre of the room, one of the traverse paths runs into a dihedral corner and another into a trihedral corner. The last one runs to the wall where the door is located (Figure 2). Path 1 and path 2 have a length of about 5 m, and path 3 has a length of about 4 m. Each path is technically realised by a tensioned, 1 mm thick fishing line to which a cable car is attached (Figure 3). A microphone is mounted to the cable car and a motor-controlled cable pulls the cable car together with the microphone along the chosen path [12].

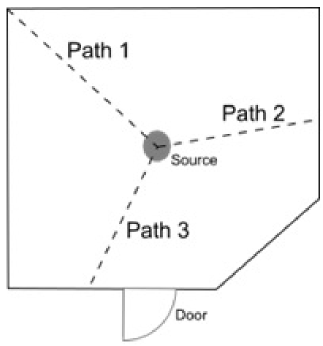

Figure 2. Sketch of the room with marked paths.

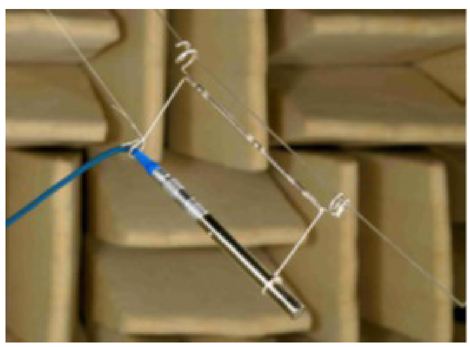

Figure 3. Microphone cable car[3].

To make a valid comparison between the current method and both new methods, the hemi-anechoic room is measured in two states. First, measurements are made under normal room conditions. Afterwards, two reflecting panels are added to the room. They have a size of 1,4 m by 1,0 m each and are made of 8 mm coated chipboard. This way a "good" room is compared to a "bad" room, and it is tested whether the change in conditions can be detected by all three measuring methods. Furthermore, it can be determined to which degree the results of all three measuring methods are equivalent.

3.2. Sound sources

Two sound sources are used to ensure an omnidirectional radiation of sound in the frequency range of interest. A loudspeaker, 75 mm in diameter, is used for frequencies from 40 Hz to 4 kHz. The other sound source is a piezo driver, which works on an exponential horn with a 3 mm opening [12]. It is used for frequencies from 4 kHz to 20 kHz. In the middle of the hemi-anechoic room, there is an opening in the floor. The sound sources are installed in this opening so that they are flush mounted with the floor. It is important that an adhesive foil is applied over the sound source with the piezo driver. The adhesive foil is supposed to mend acoustically hard edges in the area between the sound source and the floor opening and therefore prevent scattering of high-frequency sound. An opening in the foil allows sound radiation, otherwise the foil would seal the sound source.

3.3. Current Method

To provide a sufficient power output for the measurement, three different multi-sine signals are used in the frequency range of interest. The use of multi-sine signals also reduces the amount of pure tone test signals and the corresponding measurement effort [12]. Two of the three signals were generated to cover the frequency range from 40 Hz to 4 kHz and are used with the loudspeaker. The piezo driver is fed by a multi-sine signal, which provides tones between 4 kHz and 20 kHz. The signals are supplied to their intended sound source via an additional amplifier.

Aside from the microphone, which is mounted on the cable car, there is also a fixed microphone in the room. This microphone serves as the reference required by the standard in [1]. At the start of the measurement, the moveable microphone is 0.5 m away from the sound source. Both microphones are connected to a real time analyser. The Fast-Fourier-Transform (FFT) -analyser settings are identical for both microphone channels. A uniform window is applied with no overlap. The FFT length is adjusted, so that twice as many lines are used as tones in the test signal (Figure 4). Using this method, the FFT-result contains the tones of the test signal and the background noise during the measurement. This also reduces the effort of the measurements, because a second measurement for the background noise is not needed. The FFT analysis is started at the same time as the cable pull, which moves the microphone along the chosen path for a predefined distance. The results of the FFT are exponentially averaged with a time constant T_{av}. Since the microphone is moving during the measurement, this corresponds to an averaging over a certain distance r_{av} (Table 1). For resolving maxima and minima in the sound pressure level, r_{av} must be considerably smaller than the smallest wavelength λ_{min}. Every ΔT, a spectrum is written into a waterfall diagram [12]. The sound pressure is provided at the end of the analysis as a function of frequency and time or respectively at spatial steps Δr.

The described process is repeated three times per path, twice for the loudspeaker, as it is supplied with

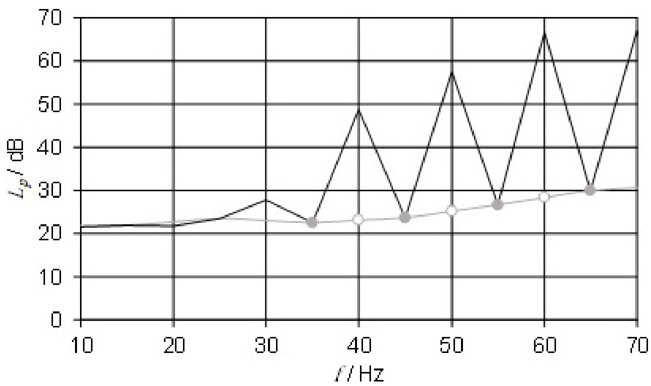

Figure 4. Noise level determination by linear interpolation using a multi-sine. The test signal contains tones at exact multiples of 10 Hz. It is analysed with an oversampled FFT and the surrounding noise levels (grey dots) are used to determine the noise level $L_{p,noise}$ at frequency f (white dots).

Table 1. Setup for the real time analyser [12].

f kHz	FFT length ms	T_{av} ms	r_{av} mm	λ_{min} mm	ΔT ms	Δr mm
0.04 - 0.4	100	2600	40	860	6500	10
0.5 - 4.0	10	260	4	86	65	1
4.0 - 20	2.5	32.5	0.5	17	65	1

two different test signals, and once for the piezo driver source.

To handle the background noise, a correction based on linear interpolation is applied. Thanks to the extra FFT-lines, each spectrum contains information about the background noise. Using the noise levels surrounding the multi-sine tones (Figure 4), the noise level that would occur at the frequency of the sine tones can be calculated by linear interpolation.

For each path and every frequency of interest, the deviation between the measured sound pressure level decrease and the ideal free-field behaviour is calculated by equation (6) for both room states [12]. To present the results of all paths at one frequency in a chart, the curves are shifted by 6 dB.

$$\Delta L_p(r,n) = L_p(r,n) - L_{p,0}(r,n) + 20\log\left(\frac{r}{r_0}\right)\text{dB} \\ - K_{abs}(r,n) - K_s(r,n) - 6(n-1)\text{dB} \quad (6)$$

$L_p(r,n)$	Measured sound pressure level
$L_{p,0}(r,n)$	The sound pressure level adjusted in such a way that the measured curve fits as well as possible into the tolerance range
$K_{abs}(r,n)$	Correction of air absorption according to ISO 9613-1 [13]
$K_s(r,n)$	Correction for the emission change of the source
r	Distance to the source
n	Number of the path

Typical results are shown in Figure 5 for both room states at 1 kHz. The distance at which the sound pressure levels leave the tolerance specified in the standard is also calculated according to [1] and [12].

3.4. Transfer function method

The transfer functions are measured with two different sine sweep signals. One sweep, ranging from 10 Hz to 4 kHz, is used with the low frequency sound source. The other sweep covers the frequencies between 4 kHz and 20 kHz and is used with the high frequency sound source.

The transfer function is measured on all three paths in 10 cm steps. The measured transfer functions in this specific case are calculated by setting the microphone output voltage in relation to the source input voltage.

The analysis method of the data is mentioned in chapter 2.2. Figure 6 shows a compact summary of this method.

3.5. Impulse response method

Since the impulse response is the inverse Fourier transform of the transfer function, the impulse response

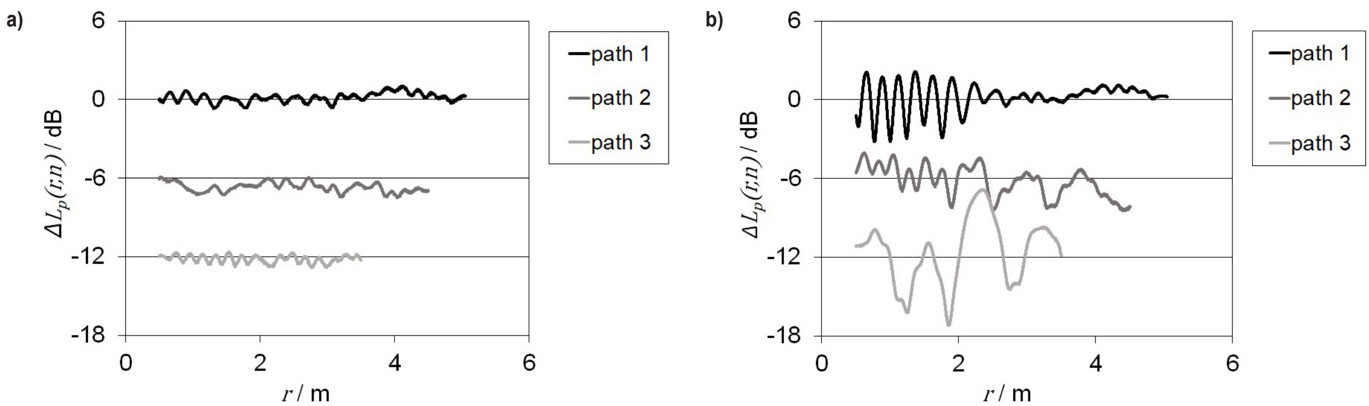

Figure 5. Typical results of the evaluation for a) the "good" room and for b) the "bad" room at 1 kHz.

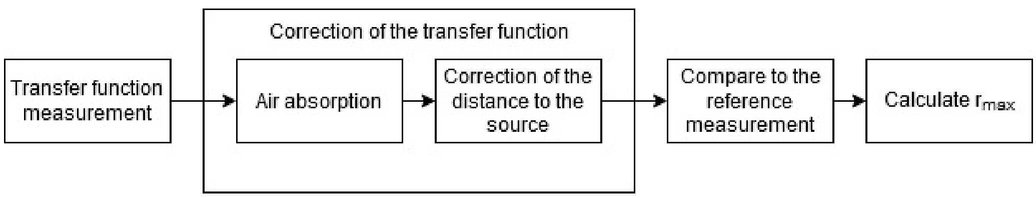

Figure 6. Analysis steps for the transfer function method.

method uses the same measurement data as the transfer function method. Therefore, measurement conditions are identical.

The analysis of the data is performed after the measurements along the path have been finished. Measured transfer functions are filtered using a high-pass filter with a cutoff-frequency of 8 Hz for the lower frequency measurement signal and 1 kHz for the higher frequency measurement signal.

Then the Inverse Fast Fourier Transforms (IFFT) of the transfer functions are calculated to acquire the impulse responses. For each impulse response, the noise level is calculated. The impulse response is then divided into direct sound, reflected sound and noise. The start of the direct sound and the reflected sound are calculated using the shortest travelled path, that the sound wave could possibly travel inside the room. In this calculation, only the room walls are considered, built-in devices are ignored to keep the calculations simple.

The direct and reflected sound are windowed using a rectangular time window. Then, the FFT of both components is calculated. An additional zero padding ensures that both calculated spectra have the same frequency resolution.

Based on these spectra, $L_{ref} - L_{dir}$ is calculated for each measurement point on the path from which r_{max} is finally calculated in analogy to the method mentioned in 2.2. A summary of the approach is given in Figure 7.

4. RESULTS

4.1. Results for the transfer function method

Figure 8 shows four different transfer functions which were measured on one path in a hemi-anechoic room without additional reflectors. The distance to the source was doubled for every measurement. The red dashed line indicates the transition between the measurements with the source for low frequencies and the source for high frequencies. The characteristics of the sources are clearly visible, especially for the high frequency source.

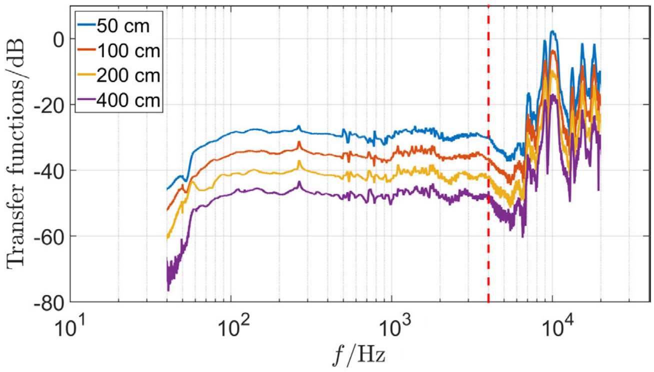

Figure 8. Transfer functions measured in different distances to the source on one path in a hemi-anechoic room without additional reflectors.

The frequencies below 40 Hz were cut off, because the measured signal is strongly influenced by background noise. The source does not emit enough sound power in that frequency range. Above 100 Hz, the transfer functions are equidistant which indicates that the measured room provides a free field.

For further analysis, the measurement at 50 cm distance to the source is used as the reference measurement.

Figure 9 shows the corrected difference calculated by eq. (1) for all 45 measured transfer functions on that path. The two red dashed lines represent the tolerance band which is defined in [1].

At most frequencies the corrected difference is inside the tolerance band for every measured distance to the

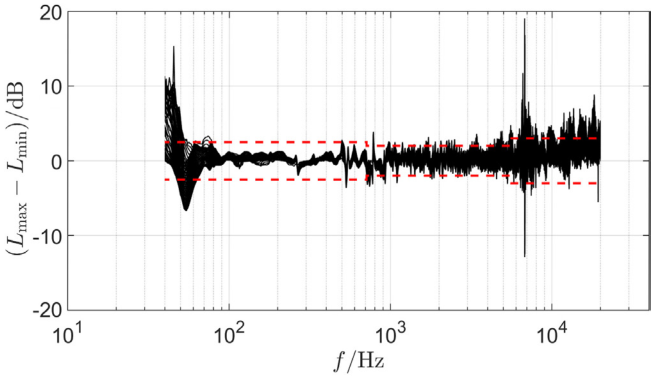

Figure 9. Corrected differences between the reference measurement and a normal measurement on one path in the hemi-anechoic room without additional reflectors.

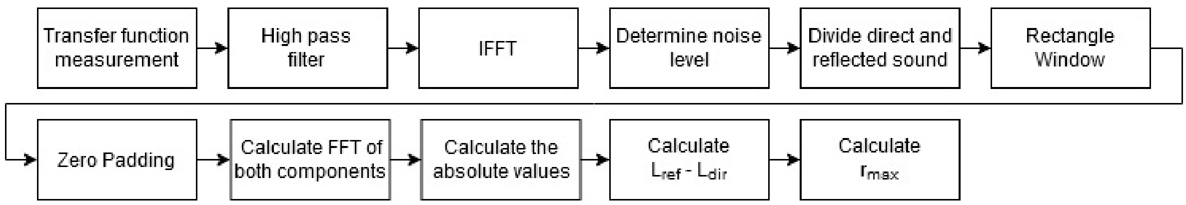

Figure 7. Analysis steps for the impulse response method.

source. However, it is outside the tolerance band at single frequencies for certain distances. At frequencies below 60 Hz, differences also leave the tolerance band which is the expected behaviour for this specific room.

On the same path, measurements were also performed with reflecting surfaces placed inside the hemi-anechoic room. The corrected difference is shown in Figure 10.

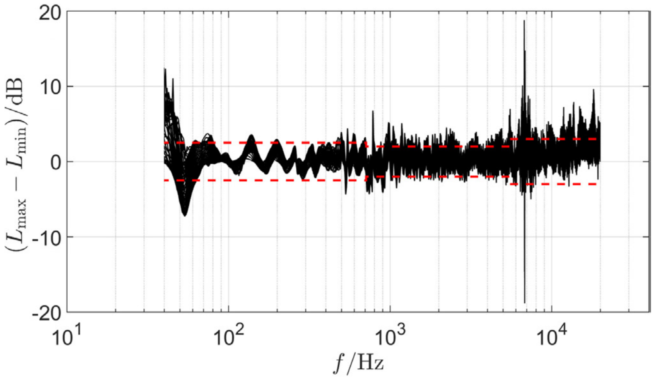

Figure 10. *Corrected differences between the reference measurement and a normal measurement on one path in the hemi-anechoic room with additional reflectors.*

In this case, the corrected difference violates the free field condition at more frequencies. At frequencies below 80 Hz the additional reflectors influence is marginal. At frequencies above 100 Hz the difference between both room conditions is clearly visible.

With these differences and the tolerance band, it is possible to calculate the corresponding r_{max} for each frequency. Figure 11 shows the calculated r_{max} for the hemi-anechoic room without reflectors. As expected, the calculated r_{max} is small at low frequencies. Between 60 Hz and 500 Hz it stays nearly constant at 4.5 m, which is the maximum measured distance. For higher frequencies the calculated r_{max} depends strongly on the specific frequency.

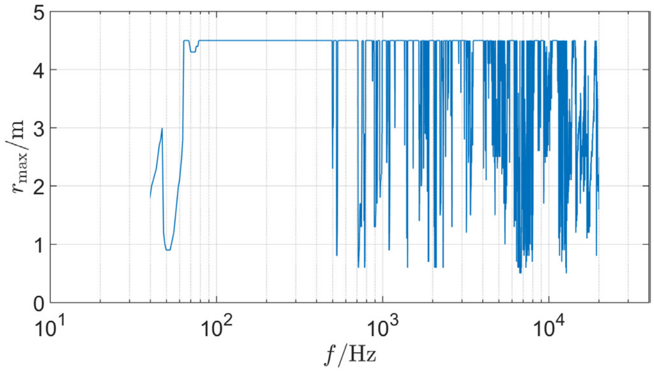

Figure 11. *r_{max} for the hemi-anechoic room without additional reflectors using the transfer function method.*

The r_{max} for the hemi-anechoic room with reflectors looks similar to this function. To compare both r_{max}, the mean of r_{max} for every one-third octave band was calculated. The results are shown in Figure 12 for both room conditions. These results are not meant to have any physical meaning. They are presented to demonstrate that the hemi-anechoic room has a much smaller r_{max} inside a specific frequency range, if the reflector is inside the room. Only in the one-third octave band at 40 Hz, the room quality improves with the reflectors inside the room possibly due to the fact that the reflectors act as absorbers at that specific frequency band.

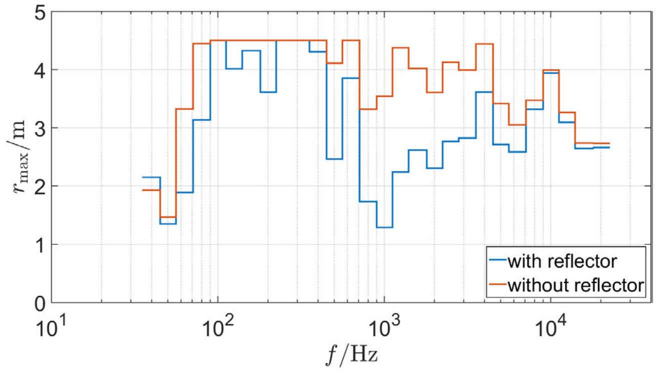

Figure 12. *r_{max} in one-third octave bands for the hemi-anechoic room with and without additional reflectors using the transfer function method.*

4.2. Results for the impulse response method

Figure 13 shows the difference between reflected and direct sound pressure levels $L_{ref} - L_{dir}$ for a distance of 50 cm to the source on one path in the hemi-anechoic room with and without the additional reflectors. The red dashed line is the calculated tolerance line, which is mentioned in 2.3. The measurement results for both sources are shown in this diagram.

Especially between 100 Hz and 600 Hz, a huge difference between both functions is observed. For

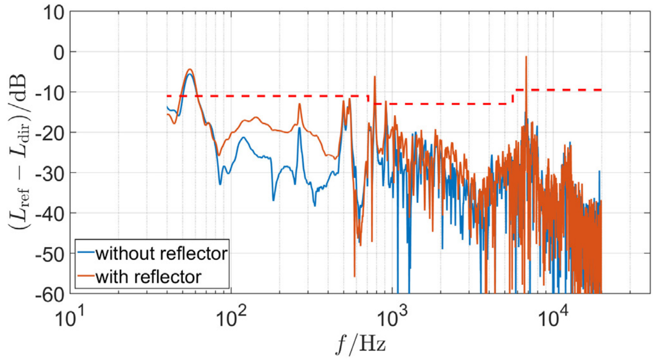

Figure 13. *$L_{ref} - L_{dir}$ for a distance of 50 cm to the source with and without additional reflectors.*

higher frequencies the functions get much more sensitive, so the result is much more frequency dependent. For most frequencies both functions are underneath the tolerance line. The room fulfils the free field condition at 50 cm for nearly all frequencies.

Figure 14 shows the same type of data as Figure 13 but with a distance of 200 cm to the source. Both functions are now closer to the tolerance line at nearly all frequencies. The difference between the measurements with and without reflectors is still clearly visible.

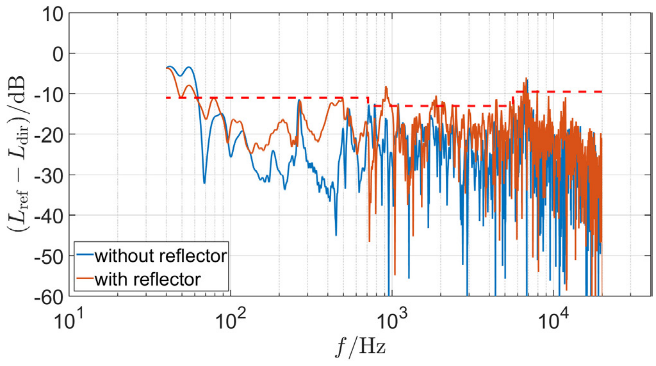

Figure 14. *$L_{ref} - L_{dir}$ for a distance of 200 cm to the source with and without additional reflectors*

The r_{max} for both room cases are shown in Figure 15. It is plotted in one-third octave bands, because the FFT results vary strongly with frequency and thus can't be compared in one graph. The sudden drop at 4 kHz is caused by the change to the physically smaller source for higher frequencies. It seems that the signal to noise ratio is too low at these frequencies for this kind of measurement.

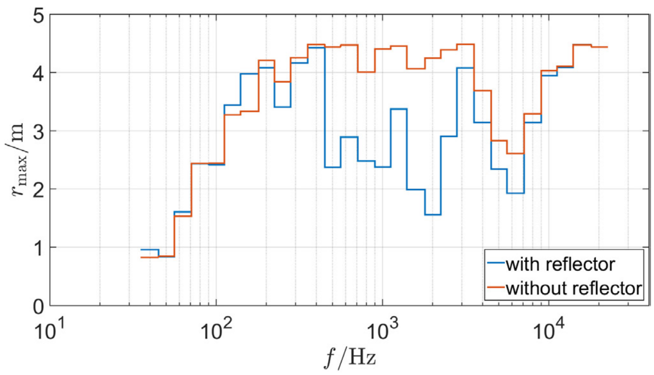

Figure 15. *r_{max} for the hemi-anechoic room using the impulse response method.*

4.3. Comparison between the different methods

The calculated r_{max} for the three different methods in the hemi-anechoic room without reflectors is shown in Figure 16. Since the measurements according to ISO 26101 were performed at specific frequencies, only discrete points are plotted for all three methods.

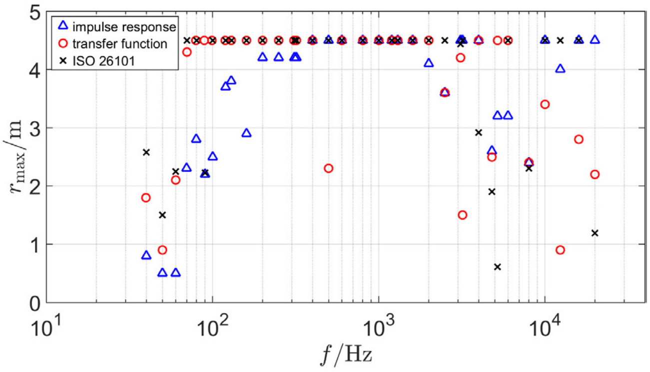

Figure 16. *r_{max} for all three methods in the hemi-anechoic room without additional reflectors.*

Especially the transfer function method gives similar results as ISO 26101. For most of the frequencies, the result is the maximum distance of 4.5 m. The smallest r_{max} is 0.5 m which is the smallest distance to the source measured.

The impulse response method generally seems to give a smaller r_{max} than the transfer function method, especially at low frequencies. Therefore, it does not give the same results as ISO 26101.

In Figure 17, the results for the room with the built-in reflectors are shown. The resulting r_{max} are lower on average for all three methods. This is especially obvious for the ISO 26101 results. For the other two methods it is hard to notice the difference, especially at high frequencies.

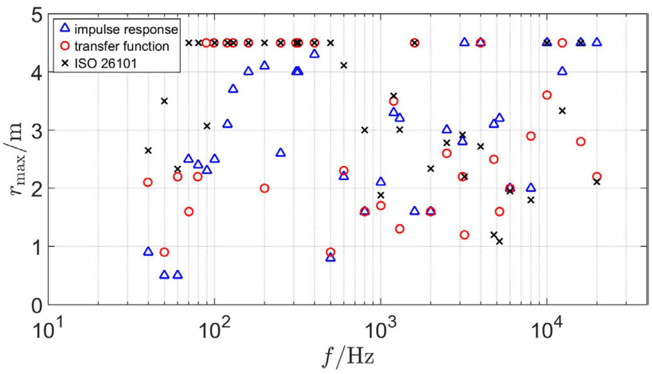

Figure 17. *r_{max} for all three methods in the hemi-anechoic room with additional reflectors.*

It is difficult to see, if the results of ISO 26101 and the newly developed methods match at high frequencies. The huge range of the values for the r_{max} at neighbouring frequencies results in a band which includes the whole area of possible values.

Therefore, at the specific measurement frequencies of the ISO 26101 measurements, the r_{max} of the newly developed method was divided by the r_{max} of the ISO

26101 measurements. This ratio should ideally be 1. For the room without the reflectors, the ratio is shown in Figure 18.

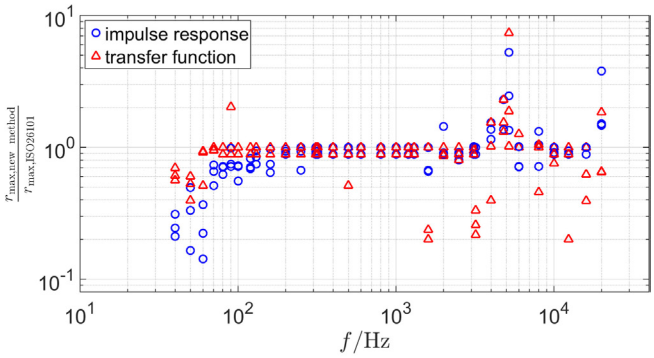

Figure 18. r_{max} *for the new methods set in relation to the* r_{max} *of ISO 26101 for the hemi-anechoic room without additional reflectors.*

For most of the frequencies, both newly developed methods give nearly identical results as ISO 26101. Especially the transfer function method shows an excellent agreement over the entire frequency range. The impulse response method exhibits some deviations at lower frequencies. At individual frequencies, deviations can be very high for both methods.

The same can be done for the room with the built-in reflectors. This is shown in Figure 19. The deviations are higher than in Figure 18, but most values are concentrated around 1.

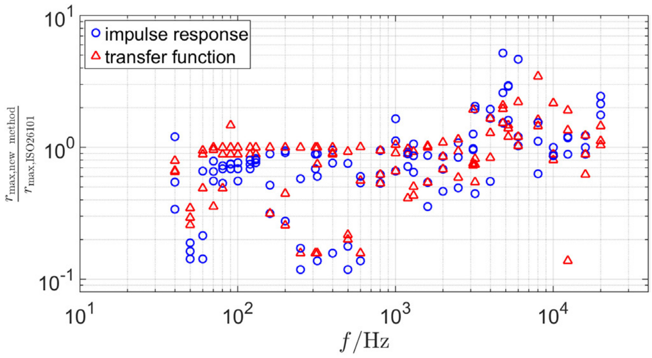

Figure 19. r_{max} *for the new methods set in relation to the* r_{max} *of ISO 26101 for the hemi-anechoic room with additional reflectors.*

5. CONCLUSION

A first attempt was made to qualify the free-field inside a hemi-anechoic room by transfer functions and impulse responses. Both methods turned out to be capable of detecting deviations from the ideal free-field behaviour with a similar measurement effort as for the currently standardised method. The qualified distances from the source up to which free-field conditions are applicable were used to compare the results of the new methods with the currently standardised method. Both new methods show promising results, in principle, even though deviations occurring at certain frequencies are significant. Further research is needed before these alternative qualification methods could be applied or even standardised.

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