Technical aspects in the qualification of free-field environments

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Introduction

Free-field environments are usually qualified according to annex A of ISO 3745 [1]. The sound pressure decrease with an increasing distance from a point source is compared to the ideal free-field behaviour. The tolerated deviations of ±1.0 dB to ±3.0 dB depend on the frequency but also on the type of the room to be qualified.

In the last few years, PTB has developed and optimised a measurement setup to perform this test. In view of the small tolerances, measurement uncertainties must be small. This requires special technical solutions which will be overviewed in this paper. One focus is on the different sources and the directivities and stabilities achieved. Other important points are the development of the multi-sine test signals, the automatic microphone traversing system, the background noise handling and the data acquisition and analysis. Finally, some selected results of room qualification measurements will be presented.

Traversing unit

The microphone is traversed by a cable car (Figure 1). It is made of 1.5 mm steel wire with a circular cross section in order to minimise possible reflections. The bearer cable consists of 1.0 mm fishing line which is very flexible and can be mounted with enough tension to bear the weight of the preamplifier and the cable. A drive cord is used as the traction rope to ensure a reliable positioning of the cable car. The traction rope is guided by deflection rollers to a belt sleeve which is driven by a stepping motor with 400 steps on the circumference. It usually stands in the room to be tested and is covered by insulating and damping material. It is also possible to place the stepping motor outside the room to be tested. The diameter of the belt sleeve is 31.85 mm which gives a continuous traversing speed of 15 mm/s for the stepping frequency of 60 Hz usually used.

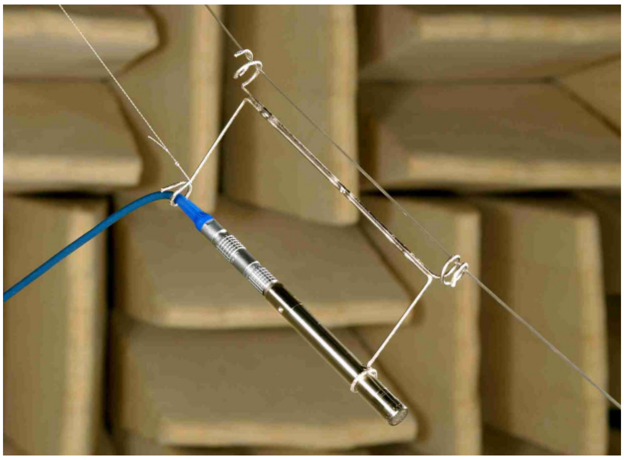

Figure 1 Microphone traversing unit

Sources

Different sources are necessary to ensure an omnidirectional radiation and a sufficient power output in the frequency range of interest. An overview of all the sources used at PTB is given in Table 1 and Table 2. The frequency range is usually divided into three ranges. For each range, different sound sources and different excitation signals are applied.

Table 1 Sources for anechoic rooms

f / kHz	Source description
0.04 – 0.4	Dodecahedron, 200 mm diameter
0.5 – 4	Compression driver with 1/2'' probe
4.8 – 20	Compression driver with 1/4'' probe

Table 2 Sources for hemianechoic rooms

f / kHz	Source description
0.04 – 0.4	Half-dodecahedron, 400 mm diameter
0.5 – 4.0	Compression driver with 1/2'' probe
4.8 – 20	Compression driver with 1/4'' probe
0.04 – 2.0	Flush mounted speaker, 75 mm diameter
2.5 – 40	Piezo driver working on an exponential horn with a 3 mm flush mounted opening

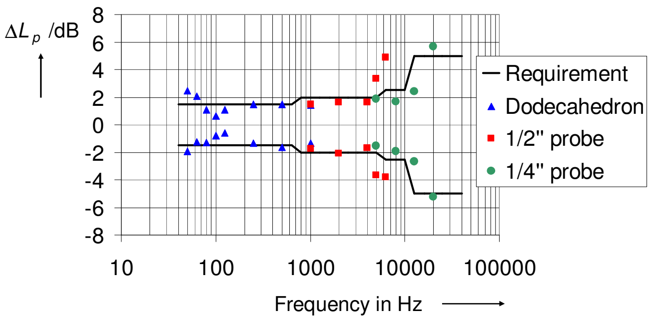

Figure 2 Directivity of the sources for anechoic rooms

The distribution of the sound pressure levels on an enveloping surface was measured to determine the directivity according to [1]. Since a tonal excitation is used by PTB in general, this excitation was also used for the directivity test. The difference between the extreme levels and the average level on the surface

$$\Delta L_p = L_{p,\max/\min} - \overline{L_p} \qquad (1)$$

must be smaller than a requirement given in [1]. This comparison is displayed in Figure 2 and Figure 3. For anechoic rooms, this test has been passed well (Figure 2). At nearly all the frequencies at least one of the sources can be used. For low frequencies, a small dodecahedron is used, whereas for the higher frequencies a compression driver with a ½'' or ¼'' probe is adequate. Only at 50 and 60 Hz and at 20 kHz, the requirement is not met (Figure 2). The deviation at low frequencies is probably due to the properties of the room in which this test was carried out, whereas the deviation at 20 kHz is due to the fact that the opening of the probe is not sufficiently small compared to the wavelength.

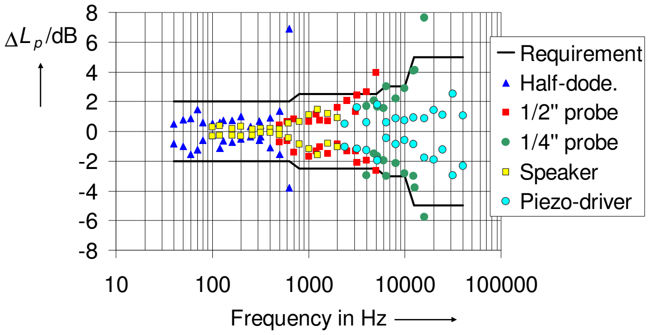

Figure 3 Directivity of the sources for hemianechoic rooms

The situation is slightly different for hemianechoic rooms. When there is an opening in the floor of the room to be tested, the preferred sources are a flush mounted speaker for frequencies up to 2 kHz and a piezo driver working on a horn with a 3 mm opening for the higher frequencies. These sources show an excellent directivity in the whole frequency range up to 40 kHz (symbols with yellow and light blue filling in Figure 3). Without a hole in the floor, the test sources have to be put on the floor. In this case, the compression driver is used with the opening of the tube as close as possible to the reflecting plane (Figure 4). At low frequencies, a half-dodecahedron is in use (Figure 5). Nevertheless, the directivity criterion is not fulfilled by these sources for frequencies above 12.5 kHz.

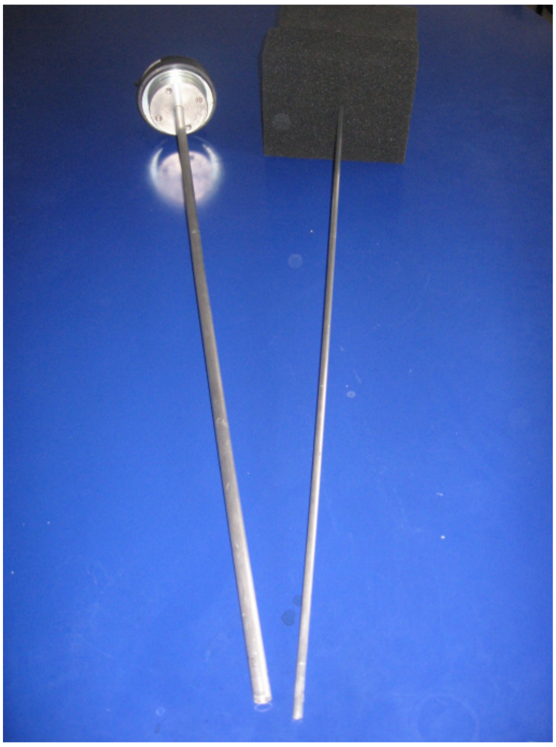

Figure 4 Compression drivers with ½" and ¼" probe to be used in anechoic or hemianechoic rooms

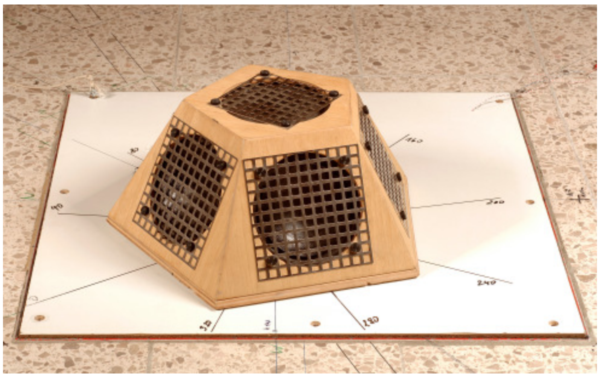

Figure 5 Half-dodecahedron for low frequencies in hemianechoic rooms

Another requirement of [1] is that the sources must be stable to within ±0.5 dB during one traverse. This requirement is generally met by the sources. Furthermore, a stationary reference microphone is always placed in the test room, and the remaining drift is compensated. This seems to be advisable in view of the very small tolerances between ±1.0 dB and ±3.0 dB to be detected.

Test signals

The current standard can be used with broadband or tonal excitation. Since broadband tests seem to miss some room deficiencies [2], PTB always advises the use of tonal excitation. The application of a multi-sine signal thereby reduces the measurement effort considerably, since many frequencies can be measured in parallel. Nevertheless, one has to take into account that the compression driver with the probe is a heavily non-linear source. Thus, using standardised one-third octave midband frequencies leads to a multitude of emitted frequencies which have not been excited (Figure 6). If such a signal is analysed by one-third octave band filters, each frequency band contains several tones which works like a broadband excitation. Therefore, it was decided to excite the sources at exact multiples of a base frequency which leads in the emission of the source to an exact line spectrum (Figure 7). If this signal is analysed by an FFT with a rectangular window and a matching length, each FFT line represents exactly one frequency.

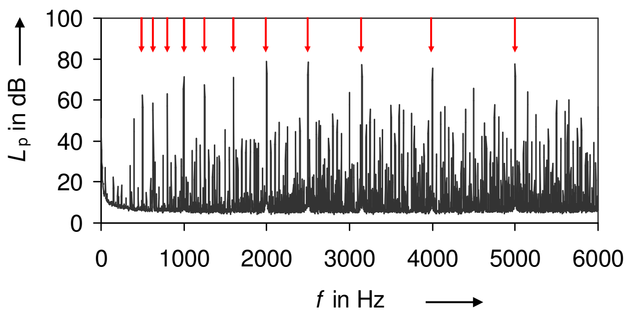

Figure 6 Sound pressure level at 1.5 m distance from the ½'' probe, FFT, Hanning window, 6400 lines between 0 and 10 kHz, multi-tone excitation at standardised one-third octave midband frequencies (marked)

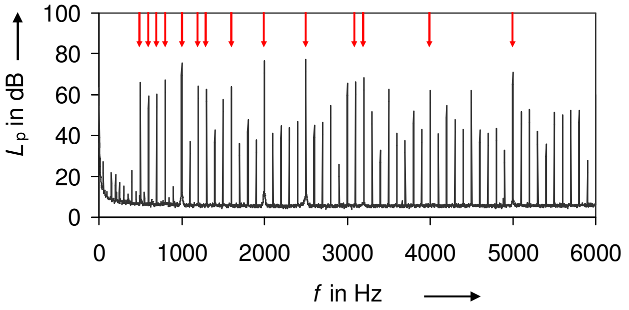

Figure 7 Sound pressure level at 1.5 m distance from the 1/2'' probe, FFT, Hanning window, 6400 lines between 0 and 10 kHz, multi-tone excitation at exact multiples of 100 Hz at marked frequencies

Another point to be obeyed with parallel one-third octave band filtering is the limited steepness of the filters. The damping between neighbouring filters is only about 20 dB. This may not be sufficient because the frequency response of the sources is often not really flat and the crosstalk leads to a correlated disturbing signal.

Background noise handling

The handling of the background noise is another important point with room qualification tests. Since a continuous traverse is used, the traversing system is one major source of disturbing noise. Therefore, all the measurements are carried out twice, once with and once without excitation. If the compression driver is used, only the opening of the probe is sealed when the background noise is measured. This leaves the radiation from the body of the compression driver unchanged which may sometimes be an important source of background noise. Since the background noise may be correlated or uncorrelated, a correction would need some further clarifications. Therefore a background noise correction is usually not applied by PTB.

Data processing

The measurement is carried out by a two-channel real time analyser (Figure 8). The analysis is identical for both channels representing the moving and the fixed microphone. A 100 line FFT with a uniform window is applied with no overlap. The FFT length is adjusted in a way that each frequency line contains only one tone. Hence, for the signal shown in Figure 7, the analysis bandwidth is 100 Hz.

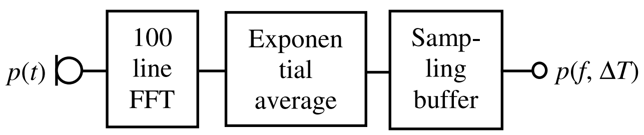

Figure 8 Data processing

FFT results are exponentially averaged with a time constant T_{av} (Table 3). This averaging corresponds to a line average r_{av} which is much smaller than the acoustic wavelength. The results are then sampled at ΔT = 650 or 65 ms which gives the final sound pressure as a function of frequency and time or respectively as distance from the source. The time step is chosen to give one sample per cm or per mm. A typical measurement result is shown in Figure 9.

Table 3 Details of the setup

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	650	10
0.5 – 4.0	10	260	4	86	65	1
4.8 – 40	2.5	32.5	0.5	8.6	65	1

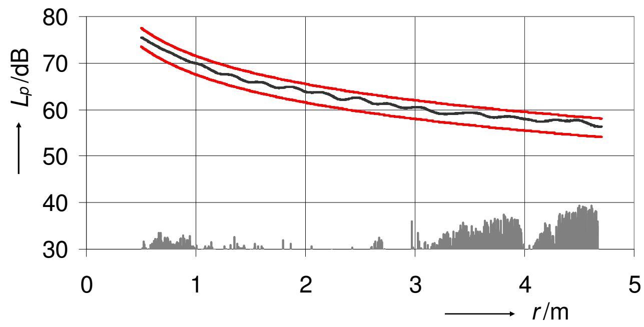

Figure 9 Measured sound pressure levels on a path (black), tolerance range from [1] (red) and background noise (grey), f = 800 Hz, compression driver with 1/2'' probe

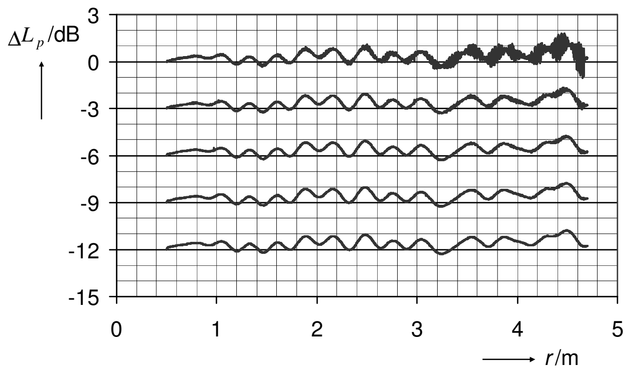

Figure 10 Normalised sound pressure levels with T_{av}=10; 30; 60; 100 and 300 ms (from upper to lower curve), f = 800 Hz, compression driver with 1/2'' probe

The measurement is carried out on at least five paths in the room to be qualified. The deviation between the measured sound pressure level decrease and the ideal free-field behaviour is calculated by

$$\Delta L_p(r,n) = L_p(r,n) - L_{p,0}(n) + 20\lg\left(\frac{r}{r_0}\right)\text{dB} - K_{abs}(r,n) - K_S(r,n) - 3(n-1)\,\text{dB} \qquad (2)$$

where $L_p(r,n)$ is the measured sound pressure level, r is the distance from the source, n is the path number, K_{abs} is a correction for air absorption according to ISO 9613 [3] and K_S is a correction for the emission change of the source. The sound pressure level $L_{p,0}(n)$ is adjusted in such a way that the measured curve fits as well as possible into the tolerance range given by [1]. The last term in eq. (2) is introduced to shift the measured curves by 3 dB per path which allows the results from different paths to be shown in one graph. An example is shown in Figure 10. In this case, the same path was measured with different averaging times. It can be clearly seen that the basic shape of the curves remains unchanged, but larger averaging times lead to smoother curves. This is especially the case at the end of the paths where the background noise is about 20 dB smaller than the signal in the example (see Figure 9).

Finally, the distance at which the measured sound pressure levels leave the tolerance range given by [1] is calculated. The volume described by these distances is then considered to be in compliance with the standard [1].

Repeatability of the results

To increase confidence in the results, measurements were repeated five times at short intervals using the same equipment. From the five repetitions, the standard deviation of repeatability of the normalised sound pressure levels (eq. (2)) was calculated. From these values, the mean and the maximum standard deviation of repeatability s_r were determined for the whole path. At frequencies between 50 and 2000 Hz, s_r is mostly below 0.2 dB (Figure 11). Larger values are observed at 40 Hz and at higher frequencies. In comparison to the tolerances of ±1.0 dB to ±3.0 dB, these standard deviations seem to be sufficiently small in the central frequency range. An application of the method at the very high frequencies is questionable due to the large s_r.

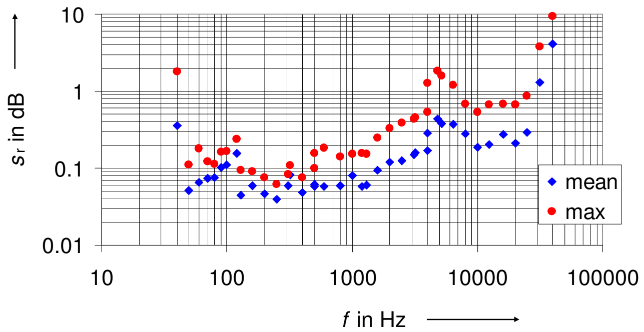

Figure 11 Standard deviation of repeatability

Results obtained with different sources

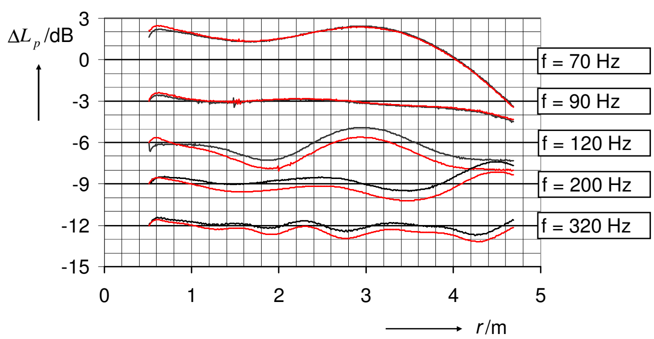

Figure 12 Comparison between the results from the flush mounted loudspeaker (black) and half-dodecahedron (red)

Another interesting aspect is whether different sources yield the same results. This has been tested for the low and medium frequencies (Figure 12, Figure 13). It turns out that measured sound pressure level decreases are nearly identical for different sources. Only the sound pressures in the vicinity of the sound sources are slightly different which leads to a systematic shift due to the normalisation used (eq. (2)).

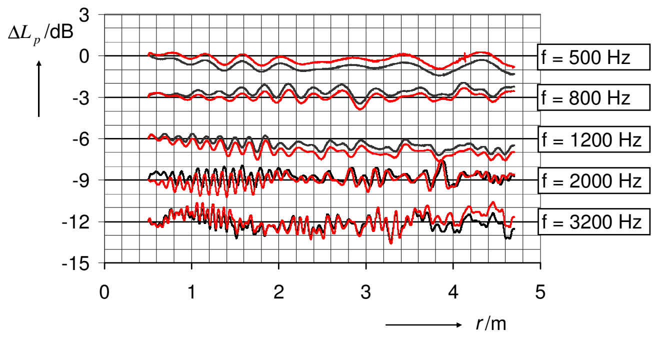

Figure 13 Comparison between the results from the flush mounted loudspeaker (red) and the compression driver with the 1/2'' probe (black)

Uncertainty of the results

The measured quantity is a sound pressure level difference by nature. Therefore, many uncertainty components like the frequency response of the microphone or the calibration of the equipment have only a minor influence. It is estimated that the sound pressure level differences have an uncertainty of about 0.1 dB in the central frequency range and reach values up to 0.2 or 0.3 dB at the lower and the higher frequencies, respectively. However, the effect of these uncertainties on the final result, the qualified volume, is still to be determined. Here, the choice of the paths, the sound source, the number of paths and many other aspects have to be taken into account.

Summary

The equipment and the procedure set up at PTB enable a reliable qualification of hemianechoic and anechoic rooms.

Acknowledgements

The design of the cablecar was inspired by Hans-Joachim Milz from G+H Schallschutz.

References

[1] ISO 3745:2003 Acoustics - Determination of sound power levels of noise sources using sound pressure - Precision methods for anechoic and hemi-anechoic rooms

[2] Cunefare, K. A.; Badertscher, J.; Wittstock, V.; *On the qualification of anechoic chambers; Issues related to signals and bandwidth*. J. Acoust. Soc. Am., Vol. 120, No. 2, August 2006, 820–829

[3] ISO 9613-2:1996 Acoustics -- Attenuation of sound during propagation outdoors -- Part 2: General method of calculation