Fast Loudspeaker Measurement in Non-Anechoic Environment

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ABSTRACT

The evaluation of the loudspeaker performance requires a measurement of the sound pressure output in the far field of the source under free field condition. If the available test room does not fulfil this condition, it is common practice to generate a simulated free field response by separating the direct sound from the room reflection based on windowing and holographic processing. This paper presents a new technique that performs a filtering of the measured sound pressure signal with a complex compensation function prior to other time and frequency analysis. The influence of room, nearfield and positioning error is compensated in the measured fundamental and nonlinear distortion characteristics. Different methods are presented for the generation of the compensation function based on a reference response measured under anechoic conditions and a test response measured under in-situ conditions. Benefits and particularities are demonstrated by practical measurements using different kinds of test signals.

1. Introduction

Traditional loudspeaker measurements assess the sound pressure output in the far field of the source under free field conditions to generate reproducible and comparable results. Those conditions can approximately be generated in anechoic rooms of sufficient size and with special wall treatment. Some anechoic rooms do not fulfill those requirements and generate an unacceptable error in the measured SPL response of the fundamental component at low frequencies. It is common practice to generate a room correction curve based on reference measurements performed under better conditions that is applied to measured SPL values and to correct the amplitude response of the fundamental component.

There are alternative techniques developed such as windowing of the impulse response and the holographic near field measurement that can be used to separate the direct sound from the room reflections, giving the fundamental frequency response under simulated free field conditions.

All the known compensation techniques are restricted to the measurement of the fundamental response and can not be applied to the measurement of harmonic and intermodulation distortions that are also affected by room reflections. The error can be reduced by performing the nonlinear distortion measurement in the near field where the direct sound is significantly larger than in the far field. However, the measurement of the maximum sound pressure output limited by a certain distortion level requires a reference point in the far field (e.g. r=1m) according to CEA 2010 and other standards.

This paper investigates the influence of a non-anechoic environment on the propagation of the nonlinear distortion components and the problems generated by applying windowing and other straightforward measurement techniques. A new compensation technique will be presented that removes the influence of the room reflections on all signal components in the measured sound pressure signal. The performance of the new technique will be illustrated by reducing the error in measured distortion based on Farina analysis and other transient techniques. The paper also addresses practical issues related to the measurement effort and the verification of the measurement results.

2. Modeling the Room Influence

The acoustical properties of the room, the sensors and other means used for clamping and positioning the loudspeaker during the test may cause a significant difference between the sound pressure $p_{test}(t,r)$ under real test environment and the sound pressure $p_{free}(t,r)$ measured at the same distance and angle under free field condition. The change of the transfer behavior can be described by a linear transfer function $H_{room}(f)$ shown in Figure 1. The loudspeaker under test can be modelled by a linear subsystem $H_{free}(f)$ and a nonlinear subsystem connected in parallel. The nonlinear distortion $p_{dis}(f)$ are negligible in the small signal domain but come comparable in magnitude with the linear output signal $p_{lin}(t)$ at higher amplitudes. The room response $H_{room}(f)$ affects not only the amplitude and phase of the fundamental component but also shapes all harmonic and intermodulation distortion generated by inherent loudspeaker nonlinearities.

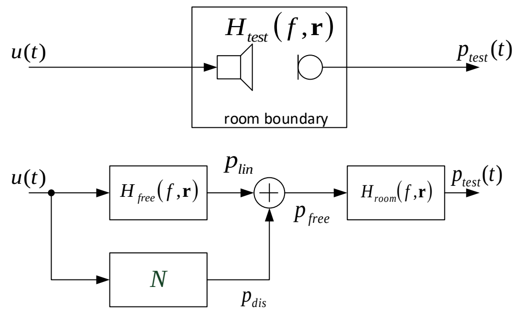

Figure 1: Modeling the loudspeaker in a non-anechoic environment

3. Simulated Free Field Conditions

The influence of the room can be compensated by different measurement and post-processing techniques, providing measurement results that simulate free field conditions.

3.1. *Time Windowing*

The direct sound radiated from the loudspeaker can be separated from early sound reflections and room modes by applying a time window to the original impulse response calculated based on the input signal u(t) and output signal $p_{test}(t)$. This approach assumes that all frequency components of the impulse response $h_{far}(t)$ of the loudspeaker under free field condition have decayed before the first room reflections arrive at the microphone. Manual user interaction is required to place the window at the right position to preserve the direct sound and suppress the reflected components. Inspecting the energy time curve can give some clues for selecting the best window type and for finding the optimum placement of the window.

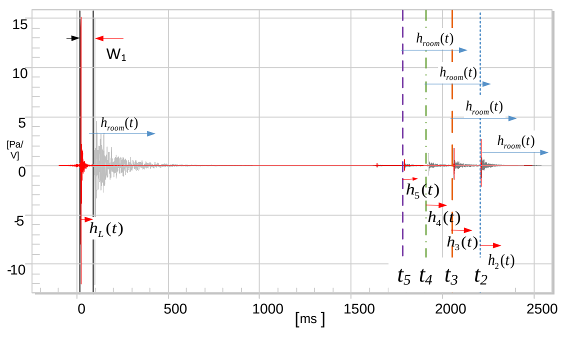

Figure 2: Impulse response measured by exciting the loudspeaker with a sinusoidal chirp with logarithmic frequency-time mapping.

3.1.1. Fundamental component

In practice, a window W_1 is only applied to the impulse response $h_1(t)$ representing the fundamental component, shown on the left side in Figure 2. This window usually provides accurate results of the amplitude and phase response at higher frequencies, where the ringing of the loudspeaker is much shorter than the arrival time of the first room reflection. At lower frequencies, resonances in the loudspeaker system may generate a longer ringing of the impulse response $h_l(t)$ which superimposes with the room response $h_{room}(t)$. A shorter window W_1 can reduce the impact of the room reflections but generates an error in the measured direct sound. The user may not be aware of the reduced resolution in the final frequency response and the error generated in the measured data because zero padding applied to the windowed impulse only generates interpolated data without increasing the frequency resolution.

3.1.2. Windowing of the harmonics

Farina introduced a powerful measurement technique for harmonic distortion that exploits the properties of a sinusoidal chirp with logarithmic frequency-time mapping. The 2^{nd}, 3^{rd} and higher-order harmonic component appear in the impulse response as acausal components on the right side of Figure 2. Turkey windows with a wide flat top region are useful to separate the harmonics and to calculate the frequency response of each harmonic component and the total harmonic distortion (THD).

The position of the turkey window depends on the first arrival time of the nth-order harmonic impulse response, that can be expressed as

$$t_n = T_{stim} - \frac{\log_2(n)}{v_{sweep}} \tag{1}$$

with the stimulus length T_{stim} and the sweep speed

$$v_{sweep} = \frac{\log_2(f_{high} / f_{low})}{T_{stim}} \tag{2}$$

using the lowest and highest frequencies f_{low} and f_{high} of the sweep. The maximum length of the window applied to the nth-order harmonic distortion can be expressed as:

$$T_n = t_{n-1} - t_n \tag{3}$$

For example, a sweep of T_{stim}=1.3 s covering the complete audio band from 20 Hz to 20 kHz limits the window of the 3^{rd}-order harmonic distortion to T_3=76 ms and the window of the 4^{th}-order distortion to T_4=54 ms. Each window should have the maximum length to provide the amplitude response of the harmonic distortion with sufficient frequency resolution. A manual reduction of the window length would reduce the frequency resolution but cannot completely suppress the room influence. The convolution of the room impulse response $h_{room}(t)$ with the nth-order impulse response $h_n(t)$, with $n \geq 2$ representing the nth-order harmonic in free field, will generate a much longer nth-order impulse response in a non-anechoic environment. Thus, the ringing of the combined nth-order impulse response $h_n(t)*h_{room}(t)$ is not negligible at the arrival time t_{n-1} and will superimpose with the impulse response $h_{n-1}(t)$ of the lower order harmonics. For this reason, the measurement of the harmonic distortion based on the Farina approach is performed in most practical applications with maximum window length T_n according to Eq. (3) therefore not suppressing the room reflections.

3.2. Holographic Field Separation

The free-field transfer function $H_{free}(f,\mathbf{r})$ of a loudspeaker can be described as a truncated series expansion of base functions $\mathbf{b}(f,\mathbf{r})$ and weighting coefficients $\mathbf{c}(f)$ as follows:

$$\begin{aligned} H_{free}(f,\mathbf{r}) &= \sum_{n=0}^{N(f)} \sum_{m=-n}^{n} C_{mn}(f) \cdot h_n^{(2)}(kr) Y_n^m(\theta,\phi) \\ &= \mathbf{c}(f) \cdot \mathbf{b}(f,\mathbf{r}) \end{aligned} \quad (4)$$

The base functions $\mathbf{b}(f,\mathbf{r})$ are solutions of the spherical wave equation in spherical coordinates. Spherical harmonics $Y_n^m(\theta,\phi)$ characterize the angular dependency on the radiation angles θ and ϕ with the order n and suborder m. The Hankel function of the second kind $h_n^{(2)}(kr)$ describes the radial dependency of a sound wave radiating from the near field to the far field.

Applying the holographic approach to non-anechoic measurements, an additional expansion term is required that describes the influence of external sound sources (e.g. reflections and room resonances). Those waves are represented by the Bessel function $j_n(kr)$. The superposition of the waves from internal and external sources gives a general solution for the sound field measured in a non-anechoic environment:

$$\begin{aligned} H_{test}(f,\mathbf{r}) &= \sum_{n=0}^{N(f)} \sum_{m=-n}^{n} [C_{mn}(f) \cdot h_n^{(2)}(kr) \cdot Y_n^m(\theta,\phi) \\ &\quad + R_{mn}(f) \cdot j_n(kr) \cdot Y_n^m(\theta,\phi)] \\ &= \mathbf{c}(f)\mathbf{b}(f,\mathbf{r}) + \mathbf{r}(f)\mathbf{b}_{room}(f,\mathbf{r}) \end{aligned} \quad (5)$$

By scanning on two layers surrounding the loudspeaker under test, the direct sound of the loudspeaker can be separated from room reflections and the free-field transfer function can be extrapolated at any point in the 3D space.

The results of this method provides a higher accuracy than traditional far field measurement because the influence of the measure room is removed and the short measurement distance minimizes the impact of air convection and temperature variation on the sound propagation.

However, the holographic technique is a multi-point measurement that requires an automatic robotics system for positioning the microphone with the required precision. The number of points depends on the frequency and the complexity of the sound source. For low frequencies (f<1 kHz), the sound field of a loudspeaker is relatively simple and can be described by a small number of expansion term (N<10) from a limited number of measurement points and time (100-200 points, 15-30 min). A complex sound source (e.g. line array) needs, for high frequencies (f>10 kHz) more expansion terms (N>30) and more measurement points (>3000 points, 8 h). This scanning effort may be justified for measuring the 3D directivity at high angular resolution, but the holographic approach is too time consuming for measurements that are usually performed at one point in the far field, such as the nonlinear distortion measurement.

3.1. Room and Position Compensation

A simulated free-field response $H_{free}(f,\mathbf{r}_r)$ at a defined reference point $\mathbf{r}_r$ can be generated by multiplying the transfer response $H_{test}(f,\mathbf{r}_t)$ measured at a test point $\mathbf{r}_t$ in a non-anechoic environment (in-situ test) with a compensation function $H_c(f)$:

$$H_{free}(f,\mathbf{r}_r) = H_{test}(f,\mathbf{r}_t) H_c(f) \quad (6)$$

The compensation function $H_c(f)$ also compensates for the near field influence and the difference in the position between the test point $\mathbf{r}_t$ and reference point $\mathbf{r}_r$.

The compensation function $H_c(f)$ can be determined by using the room transfer response $H_{room}(f,\mathbf{r}_t)$ and the free field responses $H_{free}(f)$ measured at the test and reference points:

$$\begin{aligned} H_c(f) &= \frac{\underline{H}_{free}(f,\mathbf{r}_r)}{\underline{H}_{room}(f,\mathbf{r}_t)\underline{H}_{free}(f,\mathbf{r}_t)} \\ &= \frac{H_{ref}(f,\mathbf{r}_r)}{H_{test}(f,\mathbf{r}_t)} \end{aligned} \quad (7)$$

A much simpler method is to use a reference transfer function $H_{ref}(f,\mathbf{r}_r)$ measured at a reference point $\mathbf{r}_r$ under anechoic condition.

If an anechoic room of sufficient size and wall treatment is available, a single measurement of the loudspeaker in this room while placing the microphone at the reference position $\mathbf{r}_r$ is suitable for generating a valid reference function $H_{ref}(f,\mathbf{r}_r)$. A measurement in an outside free air environment can provide more accurate results at very low frequencies, but this measurement may be prone to wind, climate and ambient noise. The holographic near field measurement performed only once for a particular loudspeaker generates the coefficients $C_{mn}(f)$ of the wave expansion in Eq. (4) that can be used to calculate the free field response $H_{free}(f,\mathbf{r}_r)$ at any reference point $\mathbf{r}_r$ in the near and far field of the loudspeaker. This calculation gives the flexibility to place the microphone at any point outside the scanning surface later in a in-situ test environment and provide the most accurate reference response for this point $\mathbf{r}_r$. Ideally, the reference measurement is required for each loudspeaker unit. In most cases, the reference measurement may be also applicable to other devices of the same loudspeaker type or even for a class of loudspeakers having similar geometrical and acoustical properties. Thus, the time and effort required to generate a compensation function $H_c(f)$ is worth it for coping with insufficient acoustical properties of the acoustical environment and limited free space to place the microphone in the far field of the loudspeaker.

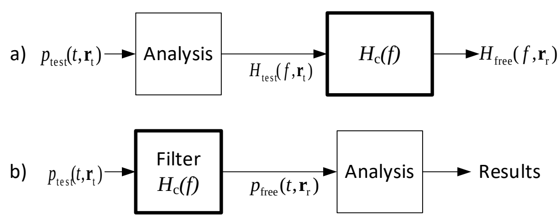

Figure 3: Compensation the influence of room and microphone positioning by correcting the transfer function $H_{test}(f,\mathbf{r}_t)$ after analysis (method a) and by filtering the measured sound pressure signal $p_{test}(t,\mathbf{r}_t)$ before analysis (method b)

There are two different techniques to apply the compensation function $H_c(f)$ to the measured sound pressure signal $p_{test}(t,\mathbf{r}_t)$, as illustrated in Figure 3. The first method a) corrects the results of the spectral and temporal analysis after performing windowing, interpolation, smoothing of the curve shape and other useful processing. This method can easily be used to correct the frequency response of the fundamental component. However, the compensation function $H_c(f)$ depends not only on the room-speaker interactions and microphone position but also on the settings of the analysis. Furthermore, the first method a) faces significant problems in correcting the results of the nonlinear distortion measurement. For example, the artefacts generated by overlapping impulse responses of the harmonic components cannot be compensated after performing the windowing in the Farina Analysis.
The second method b) applies a linear filter $H_c(f)$ to the microphone signal $p_{test}(t,\mathbf{r}_t)$ to generate a virtual signal $p_{free}(t,\mathbf{r}_r)$ at any defined reference point $\mathbf{r}_r$ under simulated free field condition. This filter can be implemented as a digital FIR filter with sufficient length to compensate for any early reflection, room modes and to change the microphone position virtually. Any kind of signal analysis or post-processing can be applied to the filtered signal $p_{ref}(t,\mathbf{r}_r)$. This is important for harmonic distortion measurement based on the Farina technique because the ringing in the impulse response is significantly reduced and separation of the higher-order impulse response becomes possible by windowing.

4. Compensation Function

There are different ways to generate the compensation function $H_c(f)$ and each method has pro and cons:

4.1. *Complete compensation*

There are two methods that compensate the linear transfer behavior at all frequencies in the interested audio band. However, only one of those methods needs a reference function $H_{ref}(f,\mathbf{r}_r)$ at all frequencies while the alternative method only requires reference information at low frequencies.

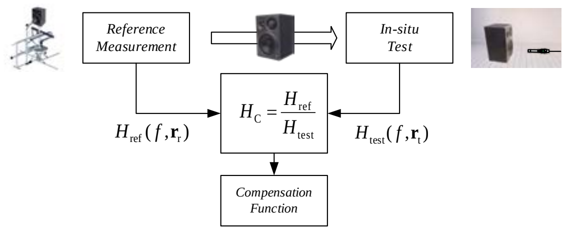

Figure 4: Generating a complete compensation function $H_c(f)$ based on a full-band reference measurement (FBR)

4.1.1. Full-band reference (FBR)

The compensation function defined in Eq. (5) is the most general case that requires a reference curve $H_{ref}(f,\mathbf{r}_r)$ providing data with sufficient accuracy and resolution at all frequencies of interest. It is possible to use a microphone position $\mathbf{r}_r$ during the in-situ test that is different from the reference point $\mathbf{r}_t$. This is beneficial for the measurement of line arrays, sound bars and other large loudspeakers where the microphone is placed in the near field of the speaker to ensure sufficient signal-to-noise ratio (SNR) in the nonlinear distortion measurements and to cope with small dimensions of the measurement room. The compensation function $H_c(f)$ represents the influence of the room reflections at all frequencies and an error in the positioning of loudspeaker and microphone in the reference and in-situ measurement. Averaging the compensation function $H_c(f)$ determined for multiple units of the same type reduces the influence of the positioning error.

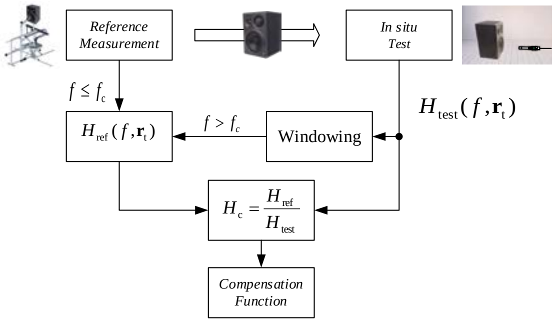

Figure 5: Generating a complete compensation function $H_c(f)$ based on a low-frequency reference function (LFR)

4.1.2. Low-frequency reference (LFR)

Figure 5 shows the alternative method, which only requires a low-frequency reference function $H_{ref}(f,\mathbf{r}_t)$ measured at the same point $\mathbf{r}_t$ as in the in-situ test at frequencies below a cut-off frequency f_c. The function $H_{ref}(f,\mathbf{r}_t)$ can be calculated by using a truncated wave expansion with a low maximum order (N < 5) wherein the coefficients can be identified by a short scan comprising less than 100 points. At higher frequencies ($f > f_c$), windowing is applied to the measured response $H_{test}(f,\mathbf{r}_t)$ to generate valid reference information. The calculated compensation function $H_c(f)$ is very similar for all units of the same loudspeaker type and almost identical for similar loudspeaker types at low frequencies. Thus, the LFR method compensates for the room influence while providing robustness against errors in loudspeaker and microphone positioning. A disadvantage of this method is that the distance between the loudspeaker and a reflecting surface should be larger than the distance to the measurement microphone to apply the windowing. This method also requires that the measurement point $\mathbf{r}_t$ of the loudspeaker is in the far field of the loudspeaker.

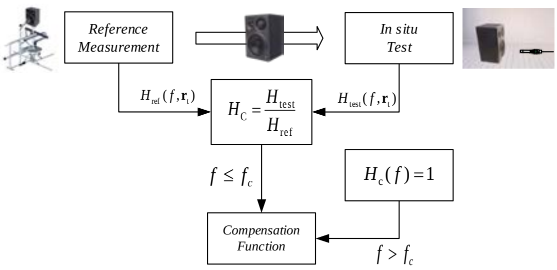

Figure 6: Generating a low frequency compensation function $H_c(f)$ based on a low-frequency reference function (LFC)

4.2. Low frequency compensation (LFC)

Figure 6 shows the calculation of a compensation function $H_c(f)$ that is restricted frequencies below f_c, which is typically about 1 kHz. The purpose is to only compensate the room interaction at low frequencies but keeps the higher frequency band as it is. This kind of compensation is useful in relatively small anechoic rooms with limited thickness of the wall absorption material where early sound reflections are sufficiently suppressed but standing waves generate significant error at low frequencies. Fortunately, the compensation function $H_c(f)$ is valid for a wide variety of loudspeakers, and the LFC method is robust against positioning errors. The reference and the in-situ measurement should be at the same point in the far field of the loudspeaker.

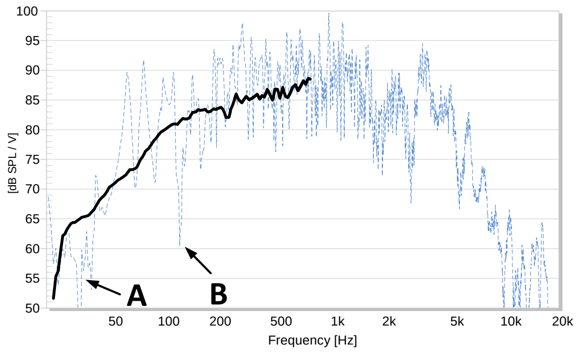

Figure 7: Amplitude response of the unfiltered sound pressure signal H_{test}(f) measured in an office (dashed line) and the low frequency reference curve H_{ref}(f) (solid line) provided by holographic near field scanning.

5. Discussion

The inverse filter technique was evaluated by practical measurements in an office environment. The tests were performed using a full-band compensation filter with a low-frequency reference (LFR) according to section 4.1.2. The low frequency reference curve $H_{ref}(f,\mathbf{r}_t)$ was determined by a previous holographic measurement, which took 15 minutes and provided accurate data up to 800 Hz. The transfer function H_{test} was measured at 1m distance on-axis in front the loudspeaker. Figure 7 shows the measured in-situ frequency response (thin curve) and the free field response extrapolated from the holographic measurement (solid line). The comparison shows that the measurement room changes the magnitude response by more than 20dB.

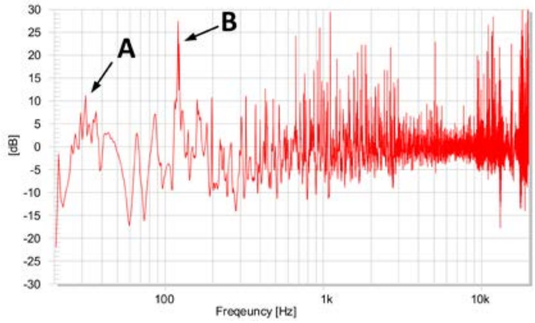

Figure 8: Magnitude response of a full band compensation function $H_c(f)$ determined based on a low frequency reference curve (LFR method).

At 120 Hz (B) and below 40 Hz (A) the measured in-situ response shows distinct cancelation effects. Based on the in-situ measurement and low frequency reference curve, the compensation filter $H_c(f)$ was calculated, which is shown in Figure 8.

5.1. *Reproducibility and Repeatability*

The reliability and robustness of the full band compensation method was investigated by the following experiments.

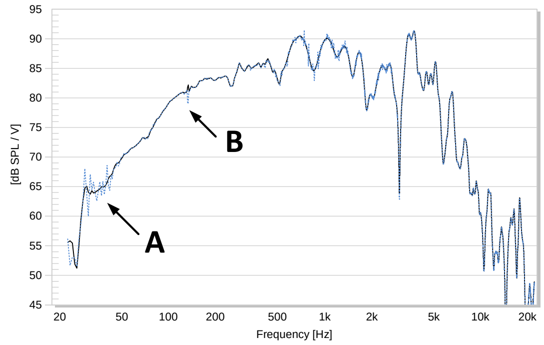

Figure 9: Influence of ambient noise on the simulated free field response based on a full band compensation function using a low frequency reference (LFR method).

5.1.1. Noise Impact

The reproducibility of the method was evaluated by repeating the measurement multiple times not changing the loudspeaker and microphone position and applying the same compensation function $H_c(f)$ generated by the LFR method. As shown in Figure 9, the measurements have small deviations in the magnitude response. An important factor for the reproducibility of the data is the signal-to-noise ratio (SNR) in the in-situ measurement. The cancelation effects are especially critical. For example, at 120 Hz (B), the room reduces the sound pressure by more than 20 dB at the microphone position. Thus, the ambient noise causes a measurement error at this frequency. The same problem occurs below 40 Hz (A), where the SNR is low. Those errors can be reduced by using common techniques like averaging or smoothing.

5.1.2. Positioning Errors

The influence of a positioning error on the reproducibility was investigated systematically. For this test, the loudspeaker position was changed by ±1 cm while keeping the microphone at same position (1 m, on-axis). The compensation filter was calculated once, and all measurements were corrected with the same filter function.

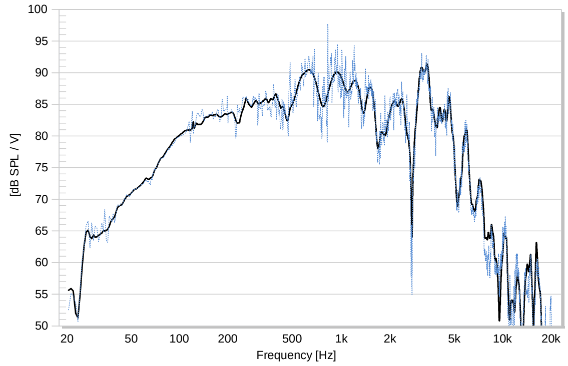

Figure 10: Influence of a positioning error on the simulated free field response based on a complete compensation function $H_c(f)$ using a low frequency reference (LFR method).

Figure 10 shows that the induced positioning error has a minor influence on the free field response below 100 Hz. For higher frequencies, the positioning error causes amplitude variation of approximately ±6 dB in the simulated free field response. This error can be avoided by repeating the measurement of the test response $H_{test}(f,r_t)$ and recalculating the compensation function $H_c(f)$ each time the position of microphone and loudspeaker has been changed.

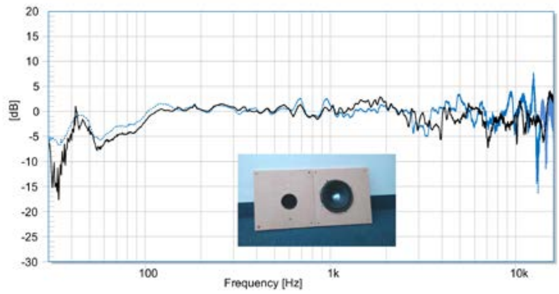

Figure 11: Influence of the transducer geometry on the low frequency compensation function $H_c(f)$ determined by the LFR compensation method.

5.1.3. Device Dependency

Two different transducers, a 3 inch (left) and a 3 inch (right), were measured in the same room at the same position. Figure 11 shows a compensation function $H_c(f)$ calculated for each loudspeaker. Although the room modes causes a common tendency in the curve shape, the size and the particular geometry of the cone shape interacting with the room generating significant changes in the compensation functions $H_c(f)$. Those differences become significantly smaller if different units of the same loudspeaker type or loudspeakers with similar cone geometries are compared.

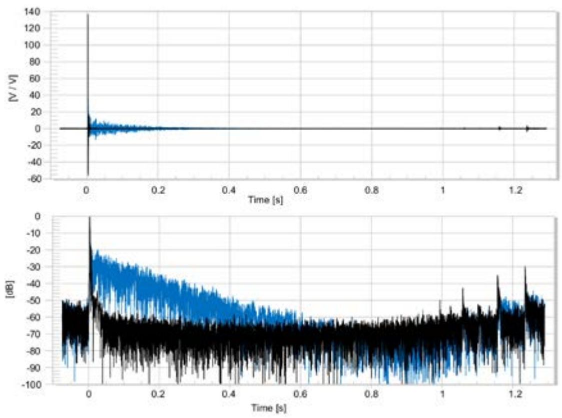

Figure 12: Impulse response (top) and the energy time curve (bottom) based on the original microphone signal (blue line) without compensation filter and based on filtered microphone signal (black line) using the compensation function $H_c(f)$ determined by the LFR method.

5.2. Harmonic Distortion Measurement

As discussed in section 3.1.2, the separation of harmonic distortion in time domain introduced by Farina requires a fast decay of the impulse responses representing the fundamental and the harmonic components. Figure 12 shows the impulse response on the upper diagram and the energy time curve on the lower part calculated with and without compensation function $H_c(f)$ applied to the microphone signal $p_{test}(f,r_t)$. This filter removes the long ringing and shortens the impulse responses. The energy time curve illustrates that the reverberant part from the fundamental, the 2nd, 3rd and higher harmonic distortion is compensated. The results are short impulses that can be separated by a time window.

The relative total harmonic distortion (THD) based on the original microphone signal with the THD curve from the filtered microphone signal where the room influence is compensated is compared in Figure 13. The compensation function $H_c(f)$ removes the high peak values at distinct frequencies and generates a much smoother curve shape. The high variation in the original THD measurement is caused by variations of the fundamental components (20 dB) and by variations of the nonlinear harmonics (6 dB), which are depicted as absolute signal components in the lower diagram. That means the nonlinear distortion measurement performed in the far field of the loudspeaker in a non-anechoic requirement needs a compensation filter in order to remove the room influence before spectral analysis.

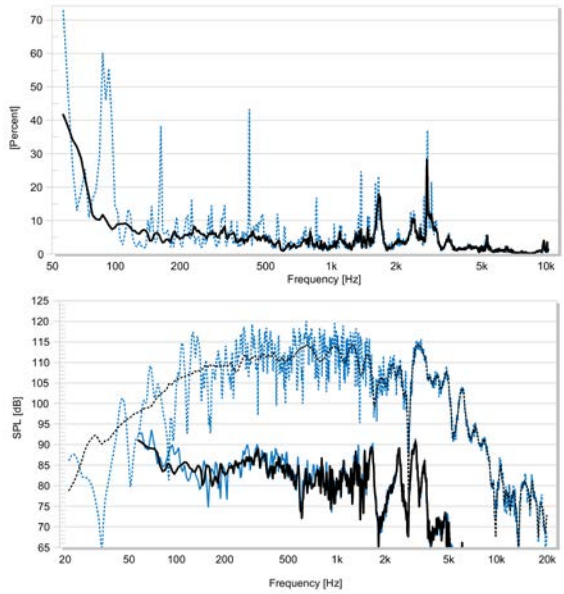

Figure 13: Relative THD (top) and fundamental with absolute THD (bottom) of the measured (blue line) and filtered (black line) sound pressure signal by using the LFR compensation method

5.3. Maximum sound pressure output

The CEA 2010 standard specifies the measurement of the maximum peak sound pressure level SPL_{max} for subwoofers. The sound pressure time signal generated by the loudspeaker at a point in the far field (r=1m) needs to be analyzed to determine the value $SPL_{max}(f)$ for a frequency f from 20 Hz to 60 Hz.

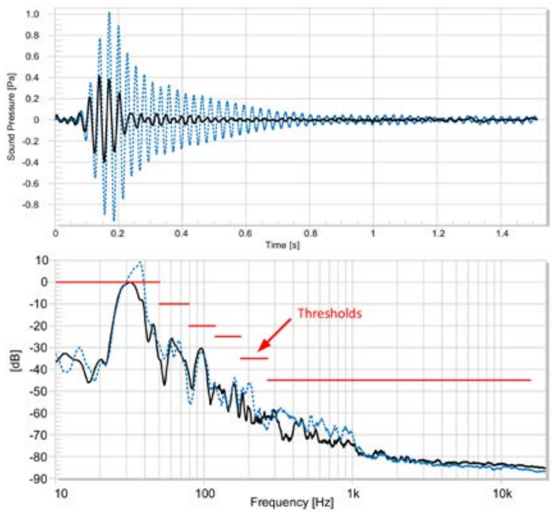

Figure 14: Waveform (upper diagram) and spectrum (lower diagram) of the original burst signal reproduced by a loudspeaker in non-anechoic environment (dashed line) compared with the filtered signal compensating the room influence (solid line).

As shown in Figure 14, the 32 Hz tone burst excites a room resonance at 40 Hz, which causes a distinct peak in the magnitude response at 40 Hz. The SPL at 40 Hz is approximately 10 dB above the SPL of the burst center frequency. The room resonance produces a long ringing in the original waveform. By filtering microphone signal with the complex compensation function $H_c(f)$, the long ringing caused by the room mode can be removed. In addition, the spectrum shows the expected 1/3-octave bandwidth of the burst.

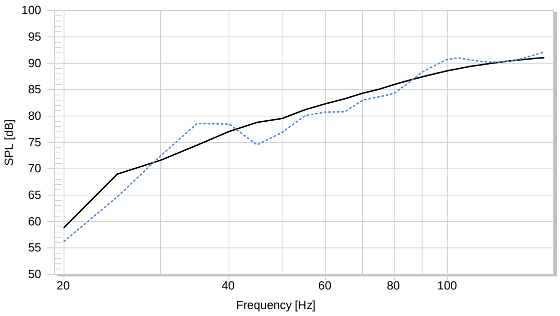

Figure 15: Maximum peak sound pressure level versus frequency of the burst signal according to CEA 2010 with (solid line) and without (dashed line) compensation of the room influence.

Figure 15 reveals about ±6dB differences in the maximum peak sound pressure levels determined with and without compensation function $H_c(f)$. The original data without room compensation shows irregular frequency variation which can not be explained by the loudspeaker behavior, but which is an artifact generated by the room modes boosting or attenuating the fundamental and harmonic components at particular frequencies. Since standard CEA 2010 reads the peak sound pressure in the time domain, the room influence can only be reduced by using a complex compensation function $H_c(f)$ that considers the amplitude and phase information to reconstruct the free field time signal correctly.

6. Applications

Table 1 gives an overview of the applications of the proposed compensation methods depending on the properties of the room and the measurement distance.

All compensation methods (FBR, LFR, LFC) are applicable to loudspeaker measurement performed in the far field in an anechoic room where early reflections on walls and measurement gear are sufficiently attenuated. The compensation method limited to low frequencies (LFC) is recommended for this acoustical environment because the compensation function $H_c(f)$ mostly compensates the standing waves in the room that correspond to the room geometry and insufficient sound absorption on the walls at low frequencies. The cone geometry and position of the loudspeaker in the room determines the excitation of each mode. However, it is strongly recommended to measure the $H_c(f, \mathrm{DUT})$ on all relevant kinds of loudspeaker types after defining the speaker and microphone position for further testing. The calculated mean value and variance of $H_c(f, \mathrm{DUT})$ over the DUTs and the requirements on the accuracy of the measurements decide whether the mean value can be used as a constant room compensation curve for further testing on similar loudspeaker.

Position and Room	**Method**		
	FBR	**LFR**	**LFC**
Far field measurement in an anechoic room	A	A	R
Far field measurement in large workshop	A	R	-
Far field measurement in a small office (reverberant)	R	-	-
Near field measurement in any environment	R	-	-

Table 1 : Recommended (R) and applicable (A) compensation methods in various acoustical environments considering the complete compensation method with low frequency reference (LFR), full-band reference (FBR) and low frequency compensation method (LFC)

In proper anechoic rooms, the variance is relatively small because the positioning error is much smaller than the wave length below 1 kHz and devices under test have a similar (omnidirectional) radiation behavior.

A large workshop room is an interesting place for performing all kinds of electrical, mechanical and acoustical measurements during everyday loudspeaker development. Here the complete compensation function $H_c(f)$ calculated based on a low frequency reference (LFR) is the preferred method. The coefficients $C_{mn}(f)$ of the wave expansion Eq. (4) identified with a minimum of scanning points allow the generation of the required reference curve $H_{ref}(f,\mathbf{r}_t)$ at any test point $\mathbf{r}_t$ determined in the final testing. Thus, the compensation function $H_c(f,\mathbf{r}_t)$ can be automatically determined after defining the test point $\mathbf{r}_t$ and measuring the in-situ response $H_{test}(f,\mathbf{r}_t)$. The windowing performed in the LFR makes the compensation function $H_c(f)$ robust against positioning errors. However, the windowing in LFR method requires that the distance between the loudspeaker and microphone is shorter than the distance between loudspeaker and other reflective boundaries (e.g. ground floor). The mean value of $H_c(f, \mathrm{DUT})$ over different devices under test reveals the excitation of the room modes and the interaction between the loudspeaker directivity and the reflective boundaries. If the variance is sufficiently small in the particular application, it is possible to use the compensation function for other types of loudspeakers and to skip a reference measurement for other loudspeaker types.

If early reflections cannot be removed by windowing in the LFR, it is recommended to use the complete compensation function with full-band reference (FBR) for all kinds of nonlinear distortion measurements. The statistical analysis of the mean value and variance of the $H_c(f, \text{DUT})$ over multiple DUTs will only provide meaningful data at low frequencies because the positioning error may generate significant deviations in magnitude and phase at higher frequencies.

The complete compensation method based on a full band reference (FBR) allows accurate distortion measurements to be performed in an office or any other room of arbitrary size and reverberant properties. This method requires a complete reference response $H_{\text{ref}}(f,\mathbf{r}_t)$ over all frequencies of interest. The compensation function $H_c(f, \text{DUT})$ is only valid for the particular device under test and the selected test point $\mathbf{r}_{t.}$ After measuring the in-situ response $H_{\text{test}}(f,\mathbf{r}_t)$, the position of the loudspeaker and microphone can not be changed during the following in-situ tests.

The measurement in the near field of the transducer by placing the microphone as close as possible to the diaphragm provides the highest signal-to-noise ratio (SNR) under the influence of ambient noise. Only the complete compensation function based on a full-band reference (FBR) can cope with the near field and provides accurate results. To provide the reference response $H_{\text{test}}(f,\mathbf{r}_t)$ at a test point $\mathbf{r}_t$ in the near field, a holographic near field scan is required. The approach is very convenient for woofers and other transducer measured in a standardized R&D test box, providing a relatively smooth response for the limited size and weight of the test box. The compensation function $H_c(f)$ determined for a transducer type can be easily applied to other units of the same type operated in the same standardized box.

End-of-line testing of loudspeakers uses a test box where loudspeaker and microphone position are clearly defined and the geometrical variation of the diaphragm and other radiating elements are negligible. The compensation function $H_c(f)$ determined from a selected “Golden reference unit” based on the FBR method can be applied to other DUTs in end-of-line testing to generate a free field response $H_{\text{free}}(f,\mathbf{r}_r)$ which is direct comparable with R&D measurements performed under standardized free field conditions.

7. Conclusions

The paper presentes a new way to generate simulated free and far field condition based on a single point measurement by filtering the microphone signal $p_{\text{test}}(f,r_t)$ with a compensation function $H_c(f)$ prior to applying windowing and other analysis. The complex compensation function $H_c(f)$ needs accurate amplitude and phase information of the room response $H_{\text{roo}}(f)$ to reduce the ringing in the linear and nonlinear impulse responses, which is required for the accurate measurement of the harmonic distortion based on the Farina technique and other transient analysis. This filter also transforms linear and nonlinear distortion that is measured at a higher SNR in the near field to a point in the far field.

The compensation function $H_c(f)$ can easily be calculated based on a valid reference response $H_{\text{ref}}(f,\mathbf{r}_t)$ and frequency response measured $H_{\text{test}}(f,\mathbf{r}_t)$ at the in-situ test point $\mathbf{r}_t$. The reference response $H_{\text{ref}}(f,\mathbf{r}_t)$ can be calculated prior to the in-situ testing based on the coefficients $C_{mn}(f)$ of the spherical wave expansion determined by holographic near field scanning. This scan is only performed once for a particular unit, type or class of loudspeaker designs with similar geometries, and it gives full flexibility in choosing the optimum distance and angle of the single test point in the in-situ measurement. The compensation function $H_c(f,\text{DUT})$ is only valid for a clearly defined microphone and loudspeaker position in the in-situ testing. The compensation functions $H_c(f)$ can be determined by different methods. The FBR method generates a compensation function at all frequencies based on a full band reference curve. This method can be used in almost any acoustical environment, but the compensation function $H_C(f, \text{DUT})$ depends on the particular device (DUT) and is prone to positioning errors. The alternative methods LFR and LFC reduce the dependency on the particular DUT and the on positioning errors and provide a compensation function for the room influence which can be applied to other loudspeakers with similar properties. The accuracy of the simulated free and far field data generated by a single point measurement performed in small anechoic rooms, normal offices or workshop can be validated at any time by performing a holographic measurement.

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