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Microphone and Sound Level Meter calibration in free field by a new method

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Abstract: A new method of calibration of microphones and sound level meter based on the sequential comparison method is presented. The proposed method allows to determine the sensibility in free-field without the use of the expensive anechoic chamber. The results here presented are in accordance with the IEC 61672-3.

Keywords: Microphone, sound level meter, free field, sweep sine, windowing.

1. INTRODUCTION

The International System of Units (SI) defines the derived quantity **Pa** as the derived unit of measurement to be used in the acoustical area. Also it is represented as sound pressure level as **dB**, where the reference level is 20 μPa, although the dB is still a off-system unit (SI). Its accomplishment is possible using the known absolute method as reciprocity. This method allows the determination of the microphone sensitivity, consequently the possibility to measure the sound pressure in a certain environment expressed in Pa or also of the sound pressure level expressed in dB. Commonly the reciprocity method is used for the obtaining of the greatness Pa with accuracy "Class LS".

When the objective is to accomplish calibrations of microphones and sound level meters (SLM) with an accuracy “Class 1” and/or “Class 2”, the used method is the comparison method or simply measuring the frequency response with the electrostatic actuator. The new IEC 61672-3 describe on the subject of the periodic verification of SLM and become compulsory the acoustic verification. It recommend the comparison method in free field or in pressure field or also with the electrostatic actuator to accomplish this verification. However, if one of the last two methods be used, a correction from the pressure field to free field should be applied in the result of the calibration. This correction factor, as well as its expanded uncertainty, is of the responsibility of the SLM manufacturer.

This work has for objective to present a solution for the problem described before. The accomplishment of a calibration in free field requests a anechoic chamber. The cost of an anechoic chamber to a Secondary Laboratory is very high. Also the option exists of working in pressure field or with electrostatic actuator. However, the correction from pressure field to free field and also its expanded uncertainty should be used. Until the moment, few SLM manufacturers make available the correction factors from pressure field to free field. The expanded uncertainty of these correction factors are still something more difficult of being found.

The presented solution for this work is the calibration by comparison in free field without the need of the use of the anechoic chamber. With this, the investment is small and also the need of application of correction factors is eliminated. The result of the calibration can be directly presented in the calibration certificate.

The idea is to use the technique of signal processing described in [2] and [3] to reach a result of free field but using a room without acoustic treatment. The technique consists of exciting the device under test (DUT) coaxial-room-microphone with a sweep. Thereby, the obtaining of the impulse response (IR) of DUT it is possible. Soon afterwards a window function is applied on the IR to suppress the undesirable reflections and reverberations. Immediately, the Fast Furrier Transform (FFT) it is applied, being possible to obtain the microphone sensitivity or SLM in free field.

2. MICROPHONE CALIBRATION

The method of microphone calibration proposed in this work is the sequential comparison where the sensitivity in free-field of the microphone under test ($M_{f,t}$) can be determined adding the sensitivity in free-field of the reference microphone ($M_{f,ref}$) to the difference of the sound pressure levels (SPL) L_t (measured SPL with the microphone under test) and L_{ref} (measured SPL with the reference microphone) as shown by the Equation 1.

$$M_{f,t} = M_{f,ref} + L_t - L_{ref} \qquad (1)$$

The technique used in this work differs of the classic technique, pure tones as excitation signal inside an anechoic chamber while the author's proposal is to use of the tool of signal processing proposed by Müller & Massarani [2]. The idea is to use a excitation signal of sweep sine, x(t), for the estimate of the impulse response, h(t), of DUT. The first step is to build an inverse filter, f(t), in such a way that the convolution of the excitation signal with this filter results in a Dirac function, as in the Equation 2.

$$x(t) \otimes f(t) \Rightarrow \delta(t) \qquad (2)$$

The impulse response that it would be obtained of the deconvolution of the response to the excitation signal, y(t), for excitation signal, x(t), it can be obtained through of the convolution of the response to the excitation signal, y(t), for inverse filter, f(t), as shown by the Equation 3.

$$h(t) = y(t) \otimes f(t) \qquad (3)$$

The FFT of f(t) and f(t) is, respectively, Y(f) and F(f). The product of Y(f) and F(f) allows the obtaining of the transfer function of the DUT, H(f). Applying IFFT on H(f) it is also obtained the impulse response of the DUT, h(t). Equation 4 represents this procedure mathematically.

$$Y(f).F(f) = H(f) \xrightarrow{IFFT} h(t) \qquad (4)$$

In this work the excitation signal, x(t), is the sweep sine built to be driven to 2 different channels (stereo). From the low frequencies until 800 Hz the sign was driven for the 16" Woofer . Starting from 1000 Hz the sign was driven for the 8" horn. The excitation signal has a duration of 2^{20} samples and it was played with a sample rate of 44,1 kHz. The figure 1 show the excitation signal used in the frequency domain (truly it represents the inverse frequency response of the coaxial load-speaker in free field). In the time domain the envelope of the amplitude is almost constant.

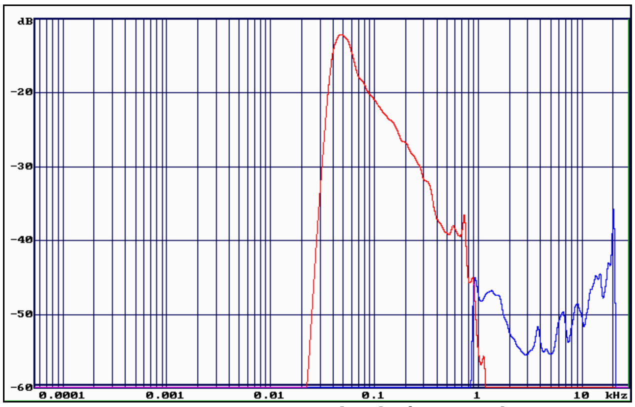

Figure 1 - Excitation signal in the frequency domain

The procedure for the obtaining of the impulse response of the DUT, when the reference microphone is measuring SPL inside the room, it counts with the immediate application of a function window, Blackman-Harris 4, to suppress the components of the reflections and also the reverberation in the case of the room be a reverberant chamber.

After this computing effort, the frequency response of the DUT can be obtained through FFT. This file can be saved in 1/3 octave band for subsequent application in the Equation 1. Of course that is possible to save this file as 1/12 octave band or also FFT. This depend on the certificate of calibration of the reference microphone. This procedure takes approximately 50 seconds. The same procedure is used for the microphone under test, allowing the complete solution of the Equation 1. Figure 2 presents the diagram of this technical procedure.

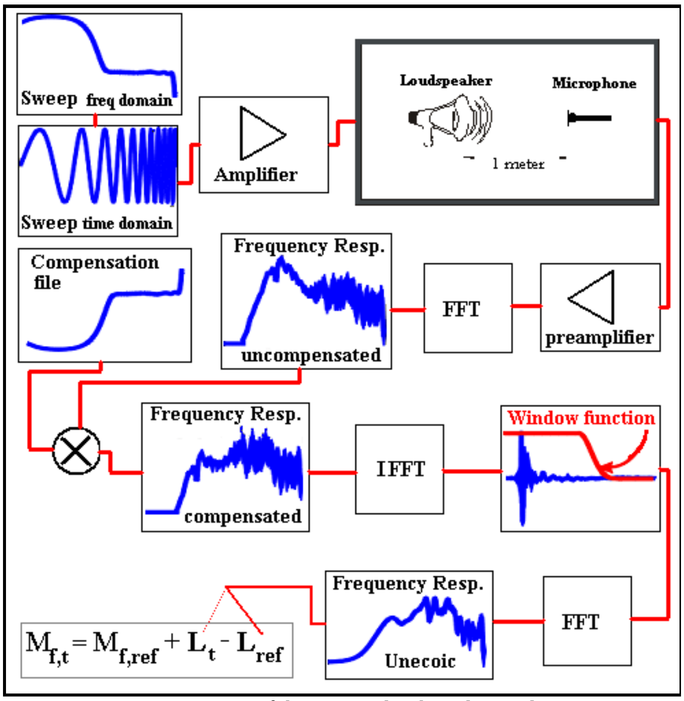

Figure 2: Diagram of the proposed technical procedure

2. FREE FIELD CONDITIONS

The condenser microphones calibration by sequential comparison in free-field doesn't still possess a Standard that can orientates the technical procedure for its accomplishment. Although IEC doesn't offer a Standard until this moment, IEC points this theme for a future project "PWI 29-2 Ed. 1.0: Calibration of working standard microphones by the comparison technique under free-field conditions". However, in September of 2000 the project 400 of the EUROMET was concluded with the publication of the Report [4] of the microphone intercomparison using the method of the sequential comparison. This document gives as guideline for Laboratories that want to accomplish the calibration by sequential comparison in free-field using a anechoic chamber.

The Report [4] In this work the authors don’t work with anechoic chamber. The report [4] uses the rmsd method to quantify the rms deviation of the free field into the anechoic chamber. The rmsd method is described in [5] and it is based on the inverse pressure/distance law and where the sound pressure level should fall 6 dB with the double of the distance.

150 points were measured and they are distant 2 mm from each point. The recommendation of 2 mm (λ/10) come from [5] and where the first point is 850 mm from the coaxial. The last point is 1150 mm from the coaxial. The idea is to quantify the deviation of the free field around 1000 mm. The proceeding of the calibration assumed here fix the

distance from the reference microphone and the coaxial in 1000mm. Then, a imaginary line linking the microphone and the coaxial gives the path to the points.

The proceeding to obtain the rmsd consist to measure the SPL (at each point) as a function of distance (for each frequency), to fit an inverse pressure/distance law (linear regression), to extract the values of the deviations of the experimental data covering a specified distance interval.

Obviously all measured SPL were using the proposed technique of IR, windowing and FFT. Figures 3, 4, 5, 6, 7 and 8 present all points measured for 125 Hz, 500 Hz, 1 kHz, 4 kHz, 8 kHz and 12,5 kHz and also the straight line of the linear regression that fit all experimental data and include also the straight line of the inverse pressure/distance law.

It is necessary to remember here that all the 1/3 octave band from 20 Hz up to 20 kHz were measured .

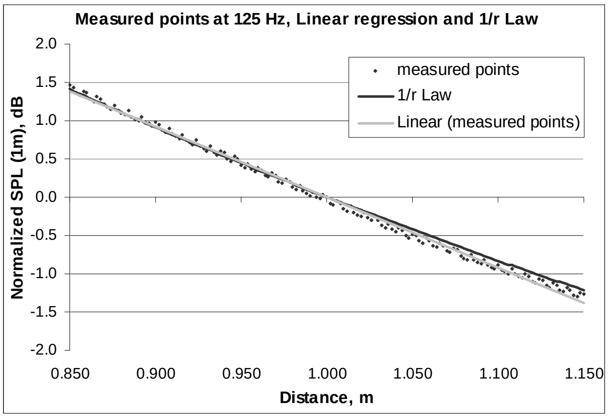

Figure 3 – all points measured for 125 Hz as a function of distance

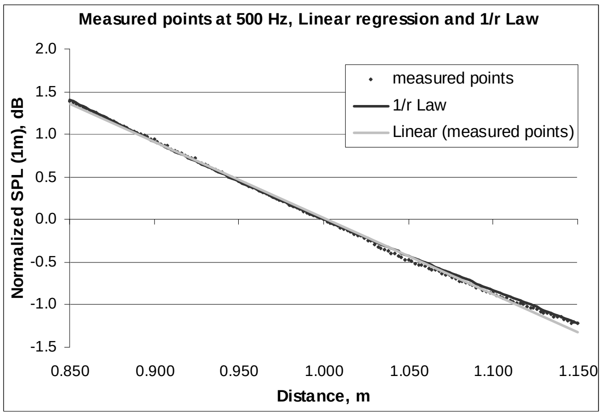

Figura4 – all points measured for 500 Hz as a function of distance

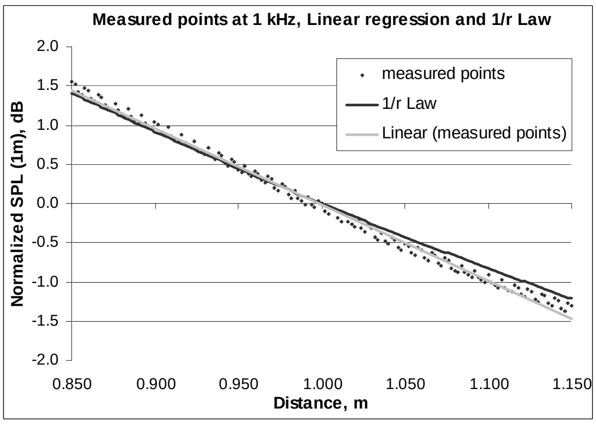

Figure 5 – all points measured for 1 kHz as a function of distance

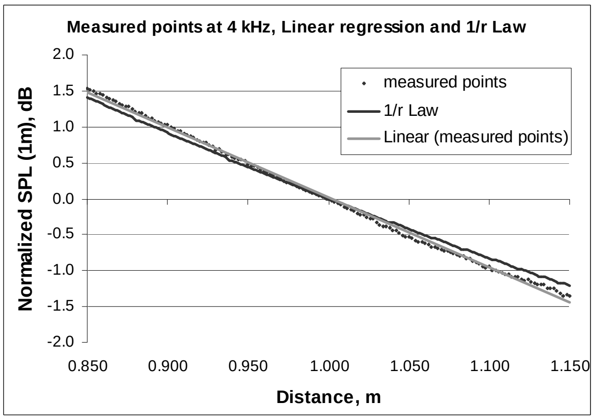

Figure 6 – all points measured for 4 kHz as a function of distance

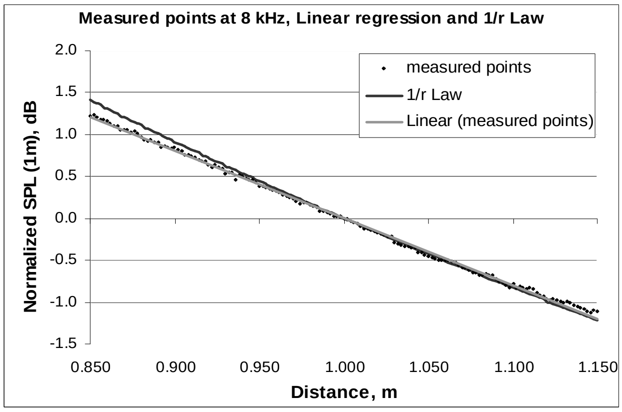

Figure 7 – all points measured for 8 kHz as a function of distance

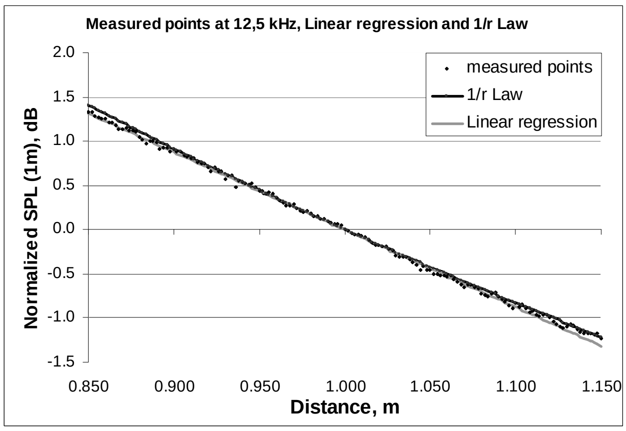

Figure 8 - all points measured for 12,5 kHz as a function of distance

After to calculate the root-mean-square of the square of the differences between the "Linear Regression" and the "1/r Law" is found the value of the rmsd for each frequency. Figure 9 presents the rmsd for the proposed method of microphone calibration in simulated free field conditions.

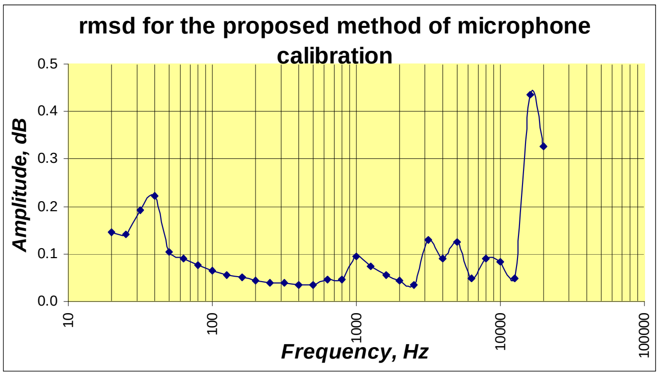

Figure 9 – rmsd for the proposed method of microphone calibration in simulated free field conditions

3. UNCERTAINTY BUDGET

As described in [6] the combined standard uncertainty $\boldsymbol{u_c}$ is the positive square root of the combined variance $\boldsymbol{u_c}^2(y)$, it is given for:

$$u_c^2(y) = \sum_{i=1}^{N}\left[\frac{\partial f}{\partial x_i}\right]u^2(x_i) \qquad (5)$$

Where f is $M_{f,t}$ in the equation 1 and x_i is each term of the right side of the equation 1. $u^2(x_i)$ is a standard uncertainty evaluated individually in the sub-items described below.

Notice that the terms with derivatives won't be them taken into account because the derivatives of the terms of the equation 1 are independent.

The uncertainty budget was estimated taking into account the effects of the uncertainty sources most significant to the proposed method. Following, it will be presented the uncertainty sources in full detail, of course it include the x_i terms of the equation 1.

3.1 Reference standard microphone calibration, $u_{mic,ref}$

The standard uncertainty of the reference microphone, $u_{mic,ref}$, is determined through the expanded uncertainty of measurement declared in the calibration certificate for the reciprocity method. As the coverage factor (k) declared in the calibration certificate is equal to 2, then it is enough to divide the value of the expanded uncertainty by 2 and to obtain the standard uncertainty of the reference microphone. The Table 1 gives the values calculated here.

3.2 Preamplifier gain - input capacitance, u_{cap}

As the technique of the insert voltage is not used and also as the used preamplifier doesn't have input impedance equal infinite the result of a calibration by comparison will be dependent of the reference microphone capacitance and of the under-test microphone capacitance. If the reference microphone is the same model of the under-test microphone then this effect can disappear. Using the Equation 6 and nominal values of the impedances is possible to estimate the gain variation of the preamplifier, ΔP_C.

$$\Delta P_C = 20\log[C_r/(C_r + C_i)] - 20\log[C_t/(C_t + C_i)] \qquad (6)$$

where:

C_r: Reference microphone capacitance in pF;
C_i: Nominal input capacitance of the preamplifier in pF;
C_t: Under-test microphone capacitance in pF.

For an input impedance of the preamplifier to be equal the 0,45 pF and assuming the reference microphone capacitance equal the 18 pF and the under-test microphone capacitance equal the 21 pF the value of ΔP_C gives -0,0303 dB. This value represent a systematic error and can be used like a correction to be applied to the final result of the calibration. But and if the values of nominal impedance has a variation of 10 %. This can give a value of the standard uncertainty $\boldsymbol{u_{cap}}$ equal the 0,0271 dB with rectangular probability distribution.

3.3 Non-linearity of the analyzer, $u_{lin,ana}$:

When the analyzer measures signals of different magnitudes is possible the result of these registers bring a error associated the non-linearity of the analyzer. The nominal sensitivity of the LS2P is -38 dB while the sensitivity of the under-test microphone can be , for instance, -26 dB. The linearity was determined through a reference generator. The standard uncertainty $\boldsymbol{u_{lin,ana}}$ with normal probability distribution is presented in Table 1.

3.4 Repeatability, u_{rep}

Ten replications were accomplished. This replications has allowed to calculate the repeatability starting from the standard deviation divided by square root of n, where n is the number of replications. The standard uncertainty $\boldsymbol{u_{rep}}$ with normal probability distribution is presented in Tables 1.

3.5 Polarization voltage, u_{pol}

If the under-test microphone is zero voltage, or pre-polarized, and the supply of 200 V diverge this value, then the sensitivity of the reference microphone will change. The polarization voltage of the microphone power supply used in this work is (200,0 ± 0,2) V, giving a semi-range of 20log(200,2/200) dB with rectangular probability distribution. The standard uncertainty $\boldsymbol{u_{round}}$ is 0,0050 dB.

3.6 Rounding error, u_{round}:

The results of the calibration declared in the calibration certificate will be written with a resolution of 0,01 dB, giving a semi-range of 0,005 dB with rectangular probability distribution. The standard uncertainty u_{round} with rectangular probability distribution is 0,005/√3 dB = 0,0029 dB.

3.7 rmsd, u_{rmsd}

The standard uncertainty u_{rmsd} is assumed with a rectangular distribution. As described in item 2, the values for each frequency were determined and now simply to divide them by $3^{1/2}$. Table 1 presents the values of u_{rmsd}. Physically this value (u_{rmsd}) represent quantitatively how much the non-uniformity of the free field can affect the final result of the microphone calibration. If the reference microphone has a different radiation impedance than of the under-test microphone, also is possible that this non-uniformity of the free field can introduce a error in the final result of the calibration.

Other comments can be done here, it is regarding of radiation impedance of microphones. In low frequencies all microphones work as a monopole radiation and this implicates not to compute the error derived of u_{rmsd}. But a conservative position will be assumed in this work, because it is included in the uncertainty budget the standard uncertainty u_{rmsd} in the low frequencies.

3.8 Expanded uncertainties , $U_{95\%}$

Adding the squares of each standard uncertainty and then taking the square root it gives the combined uncertainties u_c and, after calculating the effective freedom degree V_{eff} was possible to determine the coverage factor (k) to a level of confidence of 95 % , as presented in Tables 1.

Figure 10 presents the expanded uncertainty of the proposed method of microphone calibration in free field without the use of a anechoic chamber. Also is presented the maximum uncertainty permitted that it is given by [7].

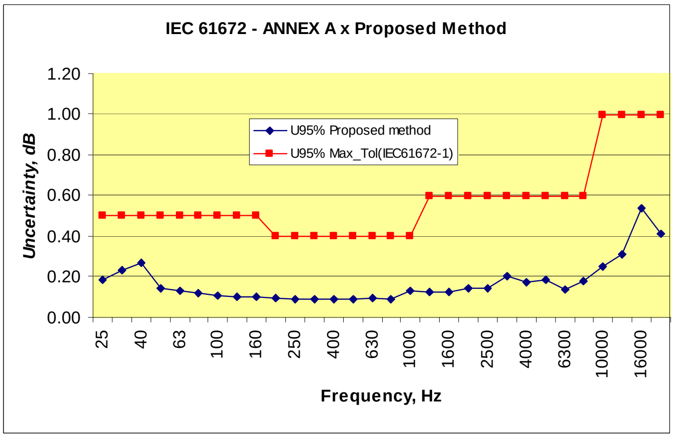

Figure 10 – Uncertainty of the proposed method and of the maximum permitted by IEC61672-1.

Table 1 – Uncertainty budget for the proposed method of microphone calibration in free field without the use of a anechoic chamber

	Frequency (Hz)																
Source	***25***	***31,5***	***40***	***50***	***63***	***80***	***100***	**125**	**160**	**200**	**250**	**315**	**400**	**500**	**630**	v_i	*distribuition*
Calibration of reference microphone	0.030	0.030	0.030	0.025	0.025	0.025	0.025	0.025	0.025	0.025	0.025	0.025	0.025	0.025	0.025	Infinite	normal
Preamplifier Gain	0.027	0.027	0.027	0.027	0.027	0.027	0.027	0.027	0.027	0.027	0.027	0.027	0.027	0.027	0.027	Infinite	retangular
Non-Linearity Analyser	0.016	0.016	0.016	0.016	0.016	0.016	0.016	0.016	0.016	0.016	0.016	0.016	0.016	0.016	0.016	Infinite	normal
Repeatability	0.016	0.015	0.013	0.011	0.009	0.007	0.007	0.007	0.007	0.006	0.006	0.006	0.006	0.006	0.007	9	normal
Voltage polarization	0.005	0.005	0.005	0.005	0.005	0.005	0.005	0.005	0.005	0.005	0.005	0.005	0.005	0.005	0.005	Infinite	retangular
Rounding	0.003	0.003	0.003	0.003	0.003	0.003	0.003	0.003	0.003	0.003	0.003	0.003	0.003	0.003	0.003	Infinite	retangular
rmsd	0.081	0.111	0.128	0.060	0.052	0.044	0.037	0.032	0.029	0.026	0.022	0.022	0.020	0.020	0.027	Infinite	retangular
Combined uncertainty	0.094	0.120	0.136	0.073	0.066	0.060	0.055	0.052	0.050	0.048	0.047	0.047	0.045	0.045	0.049	***	***
V_{eff}	11146	41965	120483	19049	28294	50384	36115	27536	30019	30972	32546	32299	29222	36031	27041	***	***
Coverage factor, k	1.96	1.96	1.96	1.96	1.96	1.96	1.96	1.96	1.96	1.96	1.96	1.96	1.96	1.96	1.96	***	***
Expanded Uncertainty, $U_{95\%}$ (dB)	**0.18**	**0.24**	**0.27**	**0.14**	**0.13**	**0.12**	**0.11**	**0.10**	**0.10**	**0.09**	**0.09**	**0.09**	**0.09**	**0.09**	**0.10**	***	***

	Frequency (Hz)																
Source	***800***	***1000***	**1250**	**1600**	**2000**	**2500**	**3150**	**4000**	**5000**	**6300**	**8000**	**10000**	**12500**	**16000**	**20000**	v_i	*distribuition*
Calibration of reference microphone	0.025	0.029	0.036	0.046	0.061	0.065	0.065	0.065	0.055	0.057	0.069	0.114	0.151	0.102	0.086	Infinite	normal
Preamplifier Gain	0.027	0.027	0.027	0.027	0.027	0.027	0.027	0.027	0.027	0.027	0.027	0.027	0.027	0.027	0.027	Infinite	retangular
Non-Linearity Analyser	0.016	0.016	0.016	0.016	0.016	0.016	0.016	0.016	0.016	0.016	0.016	0.016	0.016	0.016	0.016	Infinite	normal
Repeatability	0.009	0.007	0.005	0.007	0.006	0.007	0.004	0.009	0.009	0.009	0.012	0.016	0.016	0.016	0.015	9	normal
Voltage polarization	0.005	0.005	0.005	0.005	0.005	0.005	0.005	0.005	0.005	0.005	0.005	0.005	0.005	0.005	0.005	Infinite	retangular
Rounding	0.003	0.003	0.003	0.003	0.003	0.003	0.003	0.003	0.003	0.003	0.003	0.003	0.003	0.003	0.003	Infinite	retangular
rmsd	0.027	0.054	0.043	0.032	0.025	0.020	0.074	0.052	0.073	0.028	0.052	0.049	0.029	0.252	0.188	Infinite	retangular
Combined uncertainty	0.047	0.068	0.063	0.063	0.072	0.074	0.103	0.088	0.095	0.070	0.092	0.128	0.157	0.274	0.209	***	***
V_{eff}	6239	81788	168999	58918	181931	135770	2604505	87984	140693	40467	30730	33550	74570	686256	384256	***	***
Coverage factor, k	1.96	1.96	1.96	1.96	1.96	1.96	1.96	1.96	1.96	1.96	1.96	1.96	1.96	1.96	1.96	***	***
Expanded Uncertainty, $U_{95\%}$ (dB)	**0.09**	**0.13**	**0.12**	**0.12**	**0.14**	**0.14**	**0.20**	**0.17**	**0.19**	**0.14**	**0.18**	**0.25**	**0.31**	**0.54**	**0.41**	***	***

4. CONCLUSION

The proposed method here comply with the requirements of the allowed maximum expanded uncertainties given by the IEC61672-1. Its advantage of low cost is still intensified with the fast calibration, it takes around 20 minutes to the SLM calibration.

Other advantage is not to need to apply any correction factor on the final result. It means that the value determined during the calibration is the value to be written in the calibration certificate.

The costs for implantation of this method are also very advantageous if we compare the other methods described in IEC 61672-3. It is not necessary an anechoic chamber and the measuring system has cost equal to the of the other methods described in IEC 61672-3.

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