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Turnkey free-field reciprocity system for primary microphone calibration

Erling Frederiksen[1]
Brüel & Kjær and
Danish Primary Laboratory of Acoustics
Skodsborgvej 307
2850 Nærum
Denmark

ABSTRACT

Practically all reference microphone calibrations that are performed by national metrology institutes are pressure response calibrations. This is the case even if most practical measurements are carried out under free- or diffuse-field conditions. Fortunately, for measurement microphones, there are essentially fixed ratios between the pressure-, free- and diffuse-field responses. Therefore, the free- and diffuse-field responses may be determined by adding corrections to the pressure response. However, some institutes must develop the free-field calibration technique, calibrate microphones and determine the necessary corrections.

As no commercial free-field calibration systems have ever been available, some institutes have built systems themselves, but only a few have succeeded in making free-field calibration a routine task and even fewer have established a calibration service. Looking at the calibration principles only, free-field calibration is simpler than pressure calibration, but it is technically more difficult, especially due to the very low sound pressure that is obtainable. However, after several years of thorough research, the Danish Technical University (DTU) has developed an elaborate system and obtained international respect for their free-field reciprocity calibration. Brüel & Kjær has supported this work with instruments and technical modifications. This paper describes the technical aspects of this system, which is now offered by Brüel & Kjær with software and technical support from the university staff.

1. INTRODUCTION

The free-field reciprocity calibration method has been applied for more than half a century. It is extensively described in the literature and the International Standard IEC 61094-3[1] describes in detail the principle and the influencing parameters. Other IEC standards belonging to the same microphone and calibration series, IEC 61094-1[2] and IEC 61094-4[3] describe Laboratory Standard Microphones and Working Standard Microphones respectively. The described microphones are all condenser microphones. They are thus reciprocal and suited for reciprocity calibration. The highest calibration accuracy may be obtained with the Laboratory Standard Microphones called LS1 and LS2, like Brüel & Kjær Types 4160 and 4180 (see Figure 1). They have in comparison with most other types of microphone been thoroughly analyzed


Figure 1: Laboratory Standard Microphones, Types 4160/80

[1] Email address: erlingfred@bksv.com

and described with respect to the position of their acoustic center[1], acoustic impedance, pressure-, temperature- and humidity coefficients[4].

A very detailed description of the method applied with the new turnkey system is given in the PhD thesis[5] of Salvador Barrera Figueroa, who performed his study under supervision of the Professors Knud Rasmussen (chairman of IEC/TC29) and Finn Jacobsen at the Danish Technical University, Kgs. Lyngby.

2. TECHNICAL CHALLENGES

The new system performs free-field reciprocity calibration by using three microphones (A, B, C), as is generally the case. For each possible microphone combination (AB, AC, BC) the transfer function is measured, while one microphone acts as a source and the other as a receiver; see Figure 2. This is all described in the IEC standard[1], but it is not described how this can accurately be performed in practice – without being disturbed by reflected sound, by inherent and ambient noise and by electrical cross-talk between the source- and receiver measurement channels. The quite severe technical difficulties that are related to the measurement process are caused by the fact that the microphones are very weak sound sources and that they need to be driven with a voltage that does not exceed 6V - 8V to ensure linear and non-distorted operation.

$$p_0 = \frac{\rho\ f}{2\ d}\ M_f\ i \qquad (1)$$

$$p_0 = \frac{\pi\ \rho\ f^2 C\ u}{d}\ M_f \qquad (2)$$

$$u_{out} = \frac{\pi\ \rho\ f^2 C\ u}{d}\ M_f(source)\ M_f(receiver) \qquad (3)$$

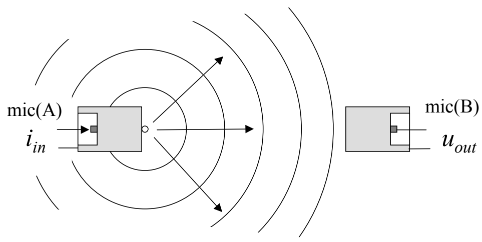

Figure 2: Principle of Free-field Transfer Function Measurement

Symbol	Parameter	Typical Value
p_0	Sound pressure module at a point distant from the source	-
ρ	Density of air	1.2 kg/m^3
f	Frequency	-
d	Distance from source to the point	LS1: 400 mm, LS2: 200 mm
M_f	Free-field sensitivity of microphone(s)	LS1: 50 mV/Pa, LS2: 12.5 mV/Pa (1000 Hz)
i	Current through the source microphone terminals	-
u_{in}	Voltage between the source microphone terminals	7 V
C	Capacitance of the source microphone	LS1: 55 pF, LS2: 20 pF
u_{out}	Output voltage of the receiver microphone	See table

The module of the sound pressure produced at a point, remote from a source microphone, may be calculated by equations (1) and (2), while the output voltage of a receiving microphone, placed at this point, is given by equation (3). The outputs of an LS1 (Type 4160) and an LS2 (Type 4180) microphone are calculated and shown in Table 1 for typical operation distances. As indicated by the numbers in the table, very low voltages have to be measured, when calibrating microphones at these frequencies. The voltages are, in fact so low at these frequencies that they are 'buried' in noise generated by the system itself and in noise occurring in the ambient. The requirement of high sensitivity calibration accuracy implies high measurement accuracy of the output voltages

Table 1. Typical output voltage (µV) of the receiver microphone during free-field reciprocity calibration

u_{out} (µV)	500 Hz	1000 Hz	2000 Hz
LS1	2.3	9.1	36.4
LS2	0.10	0.41	1.66

(0.01dB - 0.02 dB) and makes the measurements both difficult and quite time consuming. The system employs a Brüel & Kjær Multi-analyzer System Type 3560C and software for an adaptive measurement technique that is called Steady State Response (SSR). The SSR-measurement software records sine-signal packages that are sampled synchronously with the generator signal and averages them until a certain pre-selected accuracy of the averaged signal is achieved. The technique makes use of the fact that the synchronous signals add up directly, while the non-correlated noise adds up more slowly, following the law of the root of the squares. Even if this effective principle is applied, it may take several minutes to perform a complex (magnitude and phase) measurement at just one low frequency, say 800 Hz for LS1 or 3000 Hz for LS2 microphones.

Another technical challenge is to keep cross-talk from the transmitter channel to the receiver channel low. This is because the receiver channel works with output voltages that may be less than 1 micro-volt, while the transmitter channel works with 6 to 8 volts for driving the source. As the voltage ratio between the signals is thus about 10^7 times (140 dB) the cross-talk should ideally be down by 10^{10} times (200 dB), if the result is not to be disturbed by more than about 0.01 dB. However, the influence of this phenomenon and of sound reflections in the measurement room that is another, potentially very significant, source of error is minimized by the signal processing principle described in a following section.

3. PRINCIPLE OF CALIBRATION

According to the IEC standard[1] the parameters of equation (4) below must be determined to obtain the sensitivity product for each of the three microphone combinations, AB, AC and BC, where A, B and C designate the microphones. After having determined and inserted the parameters the sensitivities of all thee microphones are calculated by solving the three equations.

$$M_{f,1} \cdot M_{f,2} = -j \; \frac{2d_{12}}{\rho f} \cdot \frac{U_2}{i_1} \cdot e^{v \cdot d_{12}} \qquad (4)$$

where

Symbol	Parameter
$M_{f,1}$, $M_{f,2}$	Sensitivities of microphones ‘1’ and ‘2’
d_{12}	Distance between acoustic centers of microphones ‘1’ and ‘2’
ρ	Density of air
f	Frequency
v	Complex sound propagation coefficient
U_2	Output voltage of receiver microphone
i_1	Input current of source microphone

The system does not, in fact, measure the current of the source microphone directly. This is determined by measuring the voltage across a series capacitor that is placed close to the source microphone. The transfer impedance ($Z_{12} = U_2/i_1$) can thus, according to equation (5), be determined by a more simple voltage-ratio measurement and by an accurate calibration of the capacitance of the series capacitor.

$$Z_{12} = \frac{U_2}{i_1} = \frac{U_2}{U_1} \cdot (-j \; 2\pi \; f \, C) \qquad (5)$$

where

Symbol	Parameter
Z_{12}	Transfer impedance of microphones ‘1’ and ‘2’ valid for the parameters of equation (4)
U_1	Voltage across the series capacitor of source microphone ‘1’
C	Capacitance of series capacitor of the source microphones

The voltage ratio is a function of frequency and must be measured over the frequency range of interest for the three microphone combinations. In principle the sets of ratios only need to be measured for one distance between the microphones. This might be, for example, 400 mm for LS1 and 200 mm for LS2 microphones. But, at the primary level of calibration, national metrology institutes (NMIs) aim at an uncertainty that is as low as practically possible. Therefore, it is common to measure at more distances and obtain correspondingly more sensitivity results for comparison, for supporting the validity of each other and for averaging to a final result of higher precision. It is thus common to measure at two, three or four distances.

Often it is enough to determine the frequency response of a measurement microphone with a resolution of 1/12-octave or, in some cases, even with 1/3-octave. However, the processing of the voltage ratio results requires that they are measured with fixed frequency intervals that are typically between 100 Hz and 140 Hz. The fixed intervals lead to a relative resolution that is low at low frequencies and high at high frequencies. This fits well with the response of a microphone, whose slope is generally steeper at high than at low frequencies.

4. SYSTEM HARDWARE

The system for the measurement of the voltage ratios is shown in Figure 3. Its main instrument is the Brüel & Kjaer Multi-Analyzer Type 3560C, whose two parallel channels measure the voltage produced by the receiving microphone and the voltage across the series capacitor of the source microphone that represents its current. This is in principle, what needs to be measured, but there is typically a minor gain difference between the measurement channels that also needs to be measured and compensated for. This is done at each frequency by the insert voltage facility, which is implemented in the channels.

Several steps have been taken to minimize both the inherent noise of the system and cross-talk from the source channel to the receiver channel. For these reasons the gain of the preamplifier used with the receiver microphone is modified from its normal gain of ‘0 dB’ to ‘20 dB’.

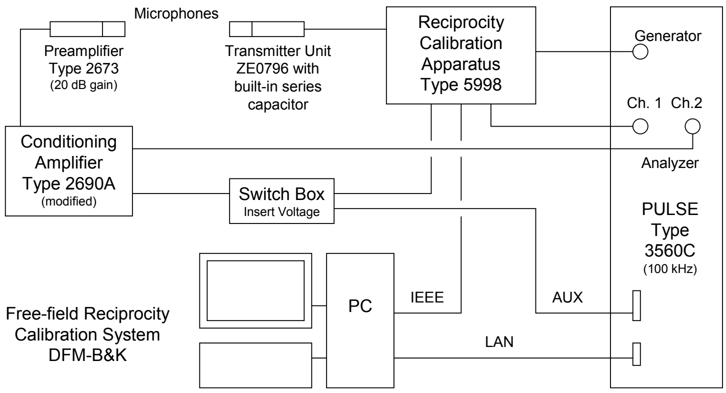

Figure 3: Block diagram of Free-field Reciprocity Calibration System with insert voltage facility for measurement of gain difference between source and receiver channels

The Brüel & Kjær Reciprocity Calibration Apparatus Type 5998 is, in fact, not designed for free-field calibration, but for the less demanding pressure reciprocity calibration. Therefore, even if this instrument has both a transmitter and a receiver channel, only the transmitter channel is applied. In order to ensure good separation between the system channels, i.e. the lowest possible cross-talk to the receiver channel, this is equipped with a separate conditioning amplifier, Brüel & Kjaer Type 2690A. Furthermore, in order to increase the immunity of the system to low-frequency noise of the ambient, an additional high-pass filter has been built into the conditioning amplifier.

5. SYSTEM SOFTWARE

The system is operated using three dedicated software programs. One is the measurement software that sets up the hardware according to pre-selected input parameters and automatically controls all measurements, until it finally stores the measurement results. A second program calculates the sensitivities in accordance with IEC 61094-3[1]. Knud Rasmussen and Salvador B. Figueroa developed both programs during their time with the Technical University of Denmark (DTU) and the Danish Institute of Fundamental Metrology (DFM). The output of the calculation program is the complex sensitivities valid at standard conditions. The results are stored in separate files for each measurement distance. A third program facilitates the presentation of the results by loading selected files into standard office programs for comparison and for preparation of reports.

6. PROCESSING OF MEASURED VOLTAGE RATIOS

The series of voltage ratio results that are to be processed are all measured at frequencies that are multiples of a pre-selected frequency step that is typically 120 Hz. The measurements themselves typically cover the ranges from about 1000 Hz to 31000 Hz for LS1 microphones and from 3000 Hz to 51000 Hz for LS2 microphones. There are always three files for each measurement distance and typically two to four distances. A sensitivity result will be obtained for each distance. All series of measurement results are to some degree ‘contaminated’ by disturbing acoustic reflections, whose influence is to be removed. To ‘clean’ the data, all measurement series are extended with additional data down to 0 Hz and up to about 51000 kHz and 70000 Hz for LS1 and LS2, respectively.

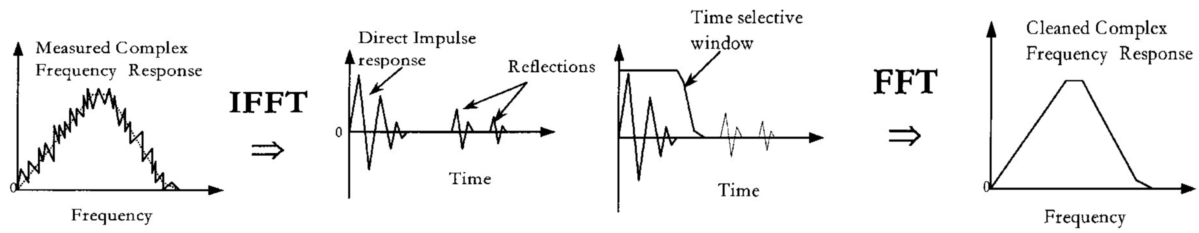

Figure 4: Principle of ‘cleaning’ voltage-ratio frequency responses from the influence of sound reflections

At the low frequencies, the calculation of the additional data for each series is based on the individual 250 Hz pressure sensitivities of the two applied microphones and on experience data describing the, very uniform, low-frequency responses of the LS1 and LS2 microphones, Types 4160 and 4180. At the high frequencies, the series are, in fact, already extended by measuring to frequencies above the normal operation ranges of the microphones. But further data is calculated by considering that the pressure responses roll off by 12 dB/octave.

The principle of the signal ‘cleaning’ process is illustrated in Figure 4. The first step of the procedure is to make inverse FFT of the extended voltage ratio frequency response. This leads to the impulse response of the two acoustically coupled microphones; see a true result in Figure 5. This response has a clear separation in time between the directly arriving sound and the delayed and disturbing sound reflections. The part of the impulse that belongs to the reflections is now removed by applying a time window. After this the results are converted back to the frequency domain by performing FFT. The influence of the reflections has, thereby, been removed from the voltage ratio transfer function.

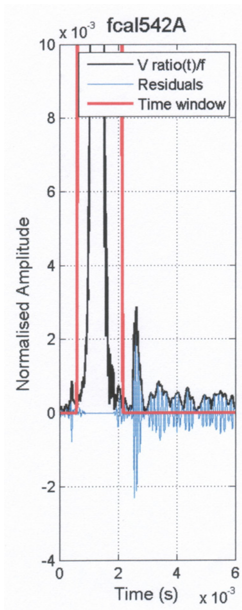

Figure 5: True transfer time function for a pair of LS1 microphones (d=250 mm)

After 'cleaning' the transfer functions, the complex sensitivities of the microphones are calculated for each measurement distance in accordance with equations (4) and (5). When measurements are made with more than one distance, the average responses can be calculated for both magnitude and phase and presented as the result of the calibration.

It should be added that the software used with the system accounts for the position of the acoustic center of the microphones. The center position is a function of frequency. For low frequencies and for the axial sound incidence, the center for LS1 microphones is about 9 mm in front of the diaphragm. For increasing frequency it moves closer to the diaphragm and is at 8 – 10 kHz at the diaphragm itself. It continues to move with frequency and is a few millimeters behind the diaphragm at higher frequencies. For LS2 microphones the corresponding numbers are 4.5 mm and 20 – 22 kHz. Data for the center position is implemented in the software, but the positions are also estimated by the system itself, if measurements are made at three or more distances. For reasons of transparency, the software stores measurement files, extended measurement files, acoustic center files and result files in linear and logarithmic frequency steps. The linear frequency steps are equal to the measurement steps. Results for logarithmic steps are determined by linear interpolation with a resolution of 1/12-octave.

7. SENSITIVITY CALIBRATION RESULTS

Free-field calibration results are in general obtained from 250 Hz to 31080 Hz for LS1 and from 250 Hz to 51120 Hz for LS2 . Figure 6 shows typical magnitude and phase response results for LS2 (Type 4180). It should be noted that these calibrations only comply with the international standard IEC 61094-3 within the frequency range covered by the true free-field measurements that for DPLA generally starts at about 1000 Hz for LS1 and at 3000 Hz for LS2. However, the determination of the low-frequency extensions to the measured voltage-ratio series are also based on some standards from the IEC 61094 series. DPLA measures the individual pressure sensitivities at 250 Hz in accordance with IEC 61094-2. Also the microphone responses from 250 Hz to the lowest measurement frequencies are based on IEC standards, partly 61094-2[6] and partly 60194-7[7], which specifies free-field corrections for perpendicular incidence on LS microphones. The file extensions and the low frequency responses do, therefore, also rest on standards and solid knowledge.

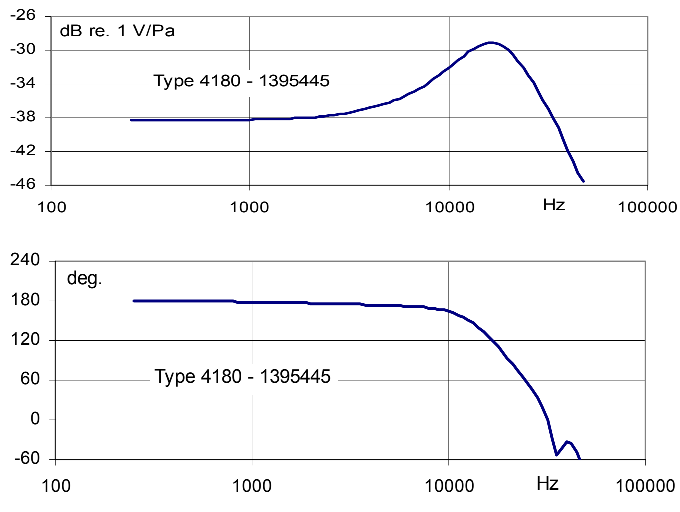

Figure 6: Measured magnitude and phase of the free-field response - Type 4180 No. 1395445.

It should, however, also be said that free-field measurements can be made at frequencies that are lower than the mentioned ones, but this would lead to a drastically extended measurement time. In fact the measurements at just few of the lowest frequencies may last longer than those at all the other frequencies of a series. As no significant differences and improvements have been observed by measuring down to lower frequencies, the above-mentioned start frequencies have been found to be a good compromise. At the high end of the frequency range measurements are made beyond the normal operation range of the microphones. This is done, because it slightly influences and improves the accuracy of the method at the highest frequencies that are within the range of interest. Valid results can be obtained to about 24 kHz for LS1 and to about 48 kHz for LS2. These limits are not set by the system, but rather by the microphones, whose frequency responses are not smooth beyond the mentioned frequencies.

A calibration made at one measurement distance will typically take 2 hours, while a thorough calibration made at four distances will take a full day. Both calibrations will, of course, result in free-field responses for each of the three microphones that took part in the process.

8. ESTIMATED CALIBRATION UNCERTAINTY

The described free-field calibration system is very similar to the system of Danish Fundamental Metrology (DFM) that together with Brüel & Kjær operates the Danish Primary Laboratory of Acoustics (DPLA). This laboratory has accreditation to perform free-field calibrations of LS1 and LS2 microphones in accordance with IEC 61094-3. The uncertainty of DFM-DPLA is specified for the two types of microphone in Table 2 and Table 3. For a new user of the free-field system it is assumed that the uncertainty will be, say 50%, higher until the staff have built up experience and detailed knowledge about the topic.

LS1 Type 4160	0.25 – 1.19 kHz	1.2 – 15 kHz	15.1 – 17.9 kHz	18 – 21 kHz	21.1 – 24 kHz
DFM-DPLA accredited	-	0.07 dB	0.08 dB	0.09 dB	-
New user of system	0.12	0.11 dB	0.12 dB	0.14 dB	0.25

Table 3: Estimated uncertainty of free-field calibration result -Type 4160 (LS1)

LS2 Type 4180	0.25 – 2.99 kHz	3 – 9.98 kHz	10 – 23.9 kHz	24 – 32 kHz	32.1 – 48 kHz
DFM-DPLA accredited	-	0.07 dB	0.09 dB	0.12 dB	-
New user of system	0.12	0.11 dB	0.14 dB	0.18 dB	0.30

Table 3: Estimated uncertainty of free-field calibration result -Type 4180 (LS2)

Generally DPLA-DFM performs calibrations with different distances between the microphones. As compensation is made for acoustic center positions and for propagation-loss in the air, there should ideally be no difference between the results obtained, if the distances are within the tested and verified range for the combination of system settings and acoustic measurement room. However, the purpose is partly to gain accuracy by a final averaging of typically four results and partly to verify that the microphones and the instruments have been stable and have worked properly, while the measurements were performed. Another advantage of performing more measurements at multiple distances is that this prevents mistakes that may be made during setting of the system and mounting of the microphones. Figure 7 shows typical deviations between four results for Type 4180 No. 1503934 that were obtained on the same day at the following distances: 170 mm, 200 mm, 240 mm and 300 mm.

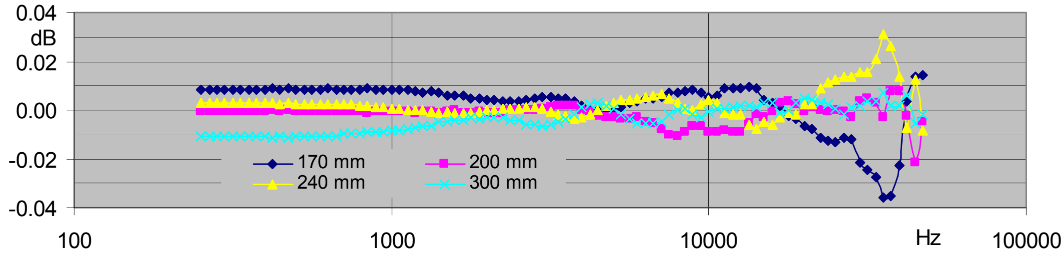

Figure 7: Deviations between the results of four free-field reciprocity calibrations performed at four different distances - Type 4180 No. 1503934. All results are shown relative to their common mean value.

9. ANECHOIC MEASUREMENT ROOM

Because of the time-selective measurement technique that the system employs, the voltage-ratio measurements do in principle not need to be performed in an anechoic room. The time window, applied with the post processing, will exclude the influence of reflections arriving from points that are more than 250 – 300 mm apart from each of the two microphones. But the much stronger reflections that would occur in a normal room, without damping material, would 'contaminate' the measurement results so much that the calculation of the low-frequency extensions and their linking to the measured data would become very difficult and far less precise. This makes one reason for using an anechoic room. Another reason is that a closed chamber with damping materials helps to keep the noise level of the background low. An an-echoic room is thus necessary, but it can be relatively small, say less than 2 m^3 and not necessarily of the highest class. The dimensions of the free space in the room used by DPLA-DFM are 0.8 m x 1.25 m x 1.7 m; see Figure 8. The wedges on the walls are 300 mm long. The microphone positioning system is automated.

Figure 8: The DPLA-DFM free-field calibration room. The microphone mounting bars are automatically moved between pre-selected microphone positions (In front: Knud Rasmussen)

10. SUMMARY

A description is given of an elaborated free-field reciprocity calibration system for laboratory standard and measurement microphones. Staff members of the Technical University of Denmark and Danish Fundamental Metrology have developed the methods and the signal processing software, while Brüel & Kjær has contributed with both dedicated and standard instruments and software. In addition to its low-noise amplifiers, the system uses the SSR-measurement technique for suppression of disturbing noise. It also uses a time selective signal processing method that minimizes the influence of sound reflections and cross-talk. Furthermore, the method reduces the requirements to the quality of the acoustic measurement room. The result is a system that works fast and gives highly repeatable results with low uncertainty. This is especially true for the Laboratory Standard Microphones Types 4160 (LS1) and 4180 (LS2), as the software contains correction data for acoustic center positions and ambient conditions for these microphone types.

ACKNOWLEDGEMENTS

Thanks to Knud Rasmussen and Salvador B. Figueroa of DFM, Kgs. Lyngby, Denmark for supporting me with details about the operation principle and about the results obtainable with the method and system described in this paper. The system is based on their research, method development and on their many years of experience with microphone calibration.

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