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NEW PRINCIPLE OF ACOUSTIC CALIBRATORS DESIGN

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Abstract: A new principle of acoustic calibrators design, based on the use of a microprocessor in the control loop of the calibrator, is presented. The mechanical and functional features of the calibrator using this principle are briefly described, and its technical specification is given. The applied algorithms of digital regulation loop and the coherent sampling of the reference signal are described in details. The carried-out experiments are presented as well as the results of the tests. The strong and the weak points of the selected solution in the relation to the traditional analogue techniques are pointed out. The conclusions concerning the practical usefulness of the developed calibrator are drawn.

Keywords: acoustic calibrator, microprocessor-based acoustic signal source, highly stable acoustic generator.

1. INTRODUCTION

An acoustic calibrator is an instrument generating a stable acoustic signal of known frequency and level. It is widely used in the acoustic metrology for calibration of measurement channels in sound-level meters and sound analyzers. The parameters of a generated signal should be stable in time, under different atmospheric conditions (temperature, pressure). Various algorithms, methods and constructions are used in the acoustic calibrators to attain this objective [1], [2], [3]. In the proposed approach, a microprocessor is used for controlling the process of signal generation. Together with the set of digital potentiometers, it ensures the possibility of the temperature and pressure compensation of the generated signal parameters. In addition, the coherent sampling is used to eliminate the necessity of analogue filters and to enable the designer to construct in the future a multi-frequency calibrator. According to our knowledge, such a solution has not yet been applied up to now in the acoustic calibrators available on the market. The proposed construction has also some new, not applied yet, interesting features described in more details in the paper, such as the automatic switch-on after inputting the tested microphone to the input of the instrument.

The aim of this work is to develop of the digital acoustic calibrator and to compare it with the well-known analogue solutions. The research is also focused on the algorithms of the digital stabilisation of the acoustic pressure level and the coherent sampling of the reference signal and their usefulness in the presented calibrator.

2. MECHANICAL DESCRIPTION

2.1. External view

The designed acoustic calibrator is a small, handheld two-range sound source meeting Type 1 (according to the standard PN-EN 60942:2003) accuracy requirements (cf. Fig. 1). The external cover of the instrument consists of specially designed aluminium profile closed by plates covered with adhesive plastic film and two rubber rings used for protection. The investigated microphone is put into the hole at the upper side of the calibrator with the rubber sealing ring. On the other side of the calibrator, there is its user interface: one push-button and two diodes which indicate the selected range.

Fig. 1. Microphone input and bottom cover of the acoustic calibrator.

2.2. Acoustic chamber

The calibrator has a built-in loudspeaker for generation of acoustic pressure and a reference microphone for checking the level of the produced signal. Both converters are placed in an acoustic chamber. The second end of the chamber is open; it is covered with plastic and a rubber seal, and is used as the input for the tested microphones. In the chamber, there is also made a special, very small hole in order to equalize the static pressure after the insertion of the microphone and to speed up the return to the initial conditions allowing the beginning of the calibration procedure.

2.3. User interface

The user interface consists only of one push-button and two diodes which causes that each element of this interface supports a few functions.

The push-button is used for switching on, switching off, changing the working range, and for hardware resetting of the calibrator. The diodes are used for the indication of the current range (94 dB or 114 dB), of the transient state of the calibrator (if the required level of the acoustic pressure in the chamber is not available), as well as of the low power and the calibration mode.

The optical sensor of the microphone's presence in the acoustic chamber can be also accounted to the user interface. It enables the automatic switch on and switch off of the calibrator after the microphone's insertion to the chamber without the usage of the push-button. In this case, the push-button can be used for changing the range of the generated signal.

3. FUNCTIONAL DESCRIPTION

3.1. Principle of operation

From the functional point of view, the calibrator is composed of two main parts: the first one is used for the generation of the required signal controlled by the feedback loop; the second is responsible for the setting of the stabilization level in that loop, the temperature and the pressure measurements, user interface handling, *etc.*

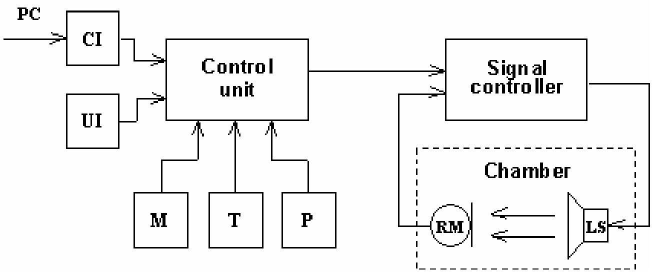

Fig. 2. Functional block diagram of the acoustic calibrator.

The detailed block diagram of the calibrator is shown in Fig. 2. It comprises a signal controller, a control unit, a loudspeaker (LS), a reference microphone (RM), a memory (M), a temperature sensor (T), a pressure sensor (P), a user interface (UI) (push-button and two diodes) and a communication interface (CI) for receiving from a PC correction coefficients and other parameters of the calibrator.

The so-called signal controller is used for the stabilization of the acoustic signal generated in the chamber. By performing digital signal processing, it works in the feedback loop together with the acoustic chamber and other necessary elements of the calibrator. The reference microphone and the loudspeaker are placed directly in the chamber and they constitute the parts of the acoustic feedback loop.

The main control unit performs the measurement of the temperature and the atmospheric pressure, the calculation of the correction to the generated signal and writing it to the signal controller. The feedback loop is automatically tuned by the signal controller to the new value of the generated signal causing its high stability in the wide range of ambient temperature and pressure. Additionally, the control unit has the access to the non-volatile EEPROM memory where the parameters of the calibrator are stored. It controls also the user interface (push-button and two diodes) and the communication interface, which is used to presetting of the calibrator. The control unit governs the work of the whole calibrator.

The acoustic calibrator has three operating modes, controlled by the control unit: the switched off mode, the normal mode and the service mode.

The control unit is working all the time, even when the calibrator is switched off. In this state, the push-button and the optoelectronic circuit are monitored permanently. It switches on the calibrator after pressing the push-button or after receiving proper signal from the optoelectronic circuit indicating that something was put into the unit's input. Additionally, one of the power saving modes of the controller is used to minimize calibrator's power consumption during its switch-off mode.

In the normal mode, the calibrator is switched on. It means that the range diodes are active, the temperature, atmospheric pressure and the battery voltage are measured and the required signal (94 dB or 114 dB) is generated.

The service mode is not available for the user of the calibrator. Using the connector, placed on the user interface board, it is possible to plug in the special cable from a PC, reload the control program and read/write the contents of the EEPROM memory. The self-calibration of the calibrator is also made in the same mode.

3.2. Digital feedback

The calibrator's acoustic chamber is placed in the feedback loop controlled by the signal controller (*cf.* Fig. 3).

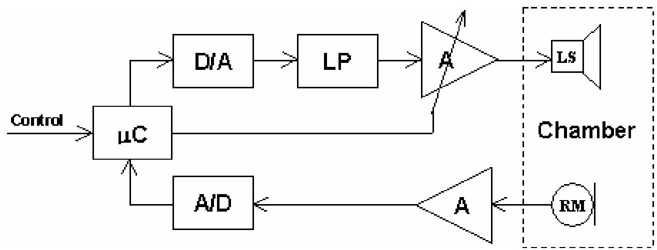

Fig. 3. Functional block diagram of the signal controller.

The output channel of the signal controller consists of the D/A converter, low-pass filter, the amplifier with the regulated gain and the micro-loudspeaker in the calibrator's chamber. The 1 kHz sinusoidal signal with the constant value of the amplitude is generated digitally and converted into the analogue form by the D/A converter [5], [6]. Next the signal is filtered in LP, amplified in A and sent to LS where the acoustic pressure is generated in the calibrator's chamber. By regulating the gain in the output channel, the signal controller influences the level of the acoustic pressure in the chamber. The resolution of the setting of the acoustic pressure level depends on the properties of the signal in the output channel of the signal controller. The accuracy of the setting is related to the resolution at which the signal controller is able to set the required pressure value in the chamber as well as of the accuracy of the measured signal from the RM.

The acoustic wave, generated by LS, is subject to the interferences, diffractions and multiple reflections from the casing, RM and other elements placed in the calibrator's chamber. The energy of the emitted acoustic wave is diminishing with the square of the distance. That is why the level of the acoustic pressure, measured by RM, differs from the level generated by LS. Additionally, the levels measured by RM and the investigated one are also different. For this reason, an assumption is made that the internal geometry of the chamber is not changed during the calibrator's work and the so-called effective volume of the microphone's load[1] of the tested microphone in a given calibrator is predefined. The tested microphone with another effective volume can cause the change of the acoustic pressure distribution in the chamber and influence the measurements performed by the reference and tested microphones.

3.3. Coherent sampling

RM is used for measuring of the RMS value of the signal generated in the acoustic chamber and transmitting this result to the signal controller. In the case of the stationary signal, the RMS value can be easily determined using the synchronic detector [4], [7].

The electric signal from the reference microphone is amplified in the amplifier A and coherently sampled by the A/D converter. During the *N* periods, $4*N$ samples of the input signal are measured on the output of the signal controller with the 90° phase-shift between the consecutive samples (*cf.* Fig.4). The values of the samples are calculated according to following equations:

$$\begin{aligned} u_1(n) &= \sin(2\pi n + \varphi) \\ u_2(n) &= \sin\left(2\pi n + \frac{\pi}{2} + \varphi\right) = \cos(2\pi n + \varphi) \\ u_1'(n) &= \sin(2\pi n + \pi + \varphi) = -\sin(2\pi n + \varphi) = -u_1(n) \\ u_2'(n) &= \sin\left(2\pi n + \frac{3\pi}{2} + \varphi\right) = -\cos(2\pi n + \varphi) = -u_2(n) \end{aligned} \tag{1}$$

where φ is the phase shift between the first value of the sinusoid and the beginning of its period (*cf.* Fig. 4).

On the basis of the collected data, the average values of the samples u_1 and u_2, are calculated from the equation:

$$\overline{u_k} = \frac{1}{2N}\sum_{n=0}^{N-1}\left[u_k(n) - u_k'(n)\right] \tag{2}$$

where k=1, 2.

Finally, the signal RMS value is calculated on the basis of two sinusoidal samples as shown below:

$$u_{RMS} = \sqrt{\frac{1}{2}\cdot\left[\left(\overline{u_1}\right)^2 + \left(\overline{u_2}\right)^2\right]} \tag{3}$$

The phase shift φ does not influence the calculated RMS value since:

$$u_{RMS} = \sqrt{\frac{A^2}{2}\left[\sin^2(2\pi n + \varphi) + \cos^2(2\pi n + \varphi)\right]} = \frac{A}{\sqrt{2}} \tag{4}$$

The phase of the input signal can be changed during the block collection of $4*N$ samples under the condition that the total number of the sampled period will be equal to N and the phase shift will be in the range ±90º. The essential is the constant, 90° shift between the consecutive samples u_1, u_2, u_1' and u_2'. Theoretically, the required shift is ensured by the microcontroller working in the set of the signal controller which sampling the input signal at the moments strictly related to the generated values of the sinusoid. The sinusoid sampling is performed every *k* samples of the output signal with the constant, strictly defined phase shift between the generated and measured sample.

In practice, the values of the acquired are subject to the jitter (the limited accuracy of time counting in a processor) and noise. The additional averaging of the samples decreases the jitter error and improves the S/N ratio of the measured RMS value of the input signal [5], [6].

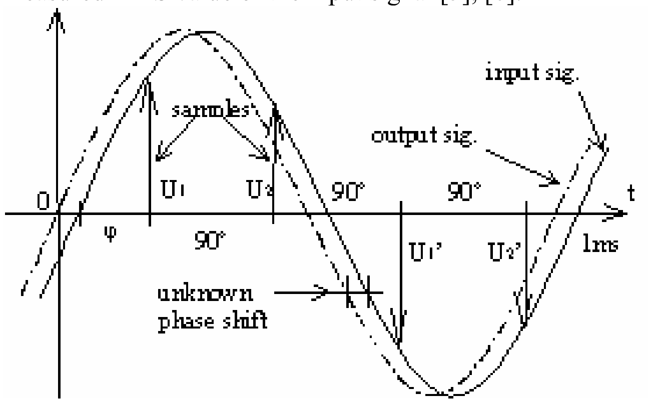

Fig. 4. The input signal, output signal and the input signal sampling.

3.4. Temperature and pressure compensations

The temperature and pressure compensation of the presented calibrator with the digital feedback loop is necessary. The main sources of the sound pressure-level errors in the chamber are the changes of the sound converters (LS and RM) properties changing with the ambient temperature and pressure. With the increase of the atmospheric pressure, the efficiency of the RM is decreasing. In lower temperatures, the RM membrane becomes more rigid and its efficiency decreases.

The application of the feedback loop in the presented solution reduces the need for compensation to the input channel of the signal controller (mainly the RM). The value of the sinusoidal signal, transmitted to the LS, is decreased/increased until the required level of the signal from the RM is attained. The output channel is always set in

[1] the effective volume of the microphone's load is the air volume which has the same acoustic susceptibility as the chamber constraint by the contact face between the microphone and the calibrator, the microphone's membrane and the external cylindrical surface of the microphone in the contact face.

relation to the input and thus does not require any thermal or pressure compensation of the level. The value of the signal gain in the output channel is irrelevant from the point of view of the acoustic pressure-stabilisation algorithm.

Due to the negative feedback loop, the calibrator has the characteristic with the opposite sign and approximately equal value in relation to the RM characteristics.

The measure of the atmospheric pressure and temperature in the calibrator's chamber is performed in the circuit of the main controller. The value of the current pressure and temperature is taken every few seconds. If the measured values differ from the previous ones, then the main controller sends to the signal controller the proper values of the correction coefficients.

4. RESULTS OF TESTS

A calibrator, developed according to the proposed principle, has been tested in different pressure and temperature conditions provided by special chambers.

The thermal drift of the calibrator was investigated using the thermostatic chamber in the temperature range from 0° to +50°. Separately, RM and its preamplifier were verified. The exemplary thermal characteristics of the calibrator before and after compensation, as well as the characteristics of its RM and the preamplifier, are shown in Fig.5.

The linearity nature of the obtained characteristic enables one to extrapolate it on the temperature ranges $-20^{\circ} \div 0^{\circ}$ and $50^{\circ} \div 60^{\circ}$ as stated in the standard PN-EN 60942:2003.

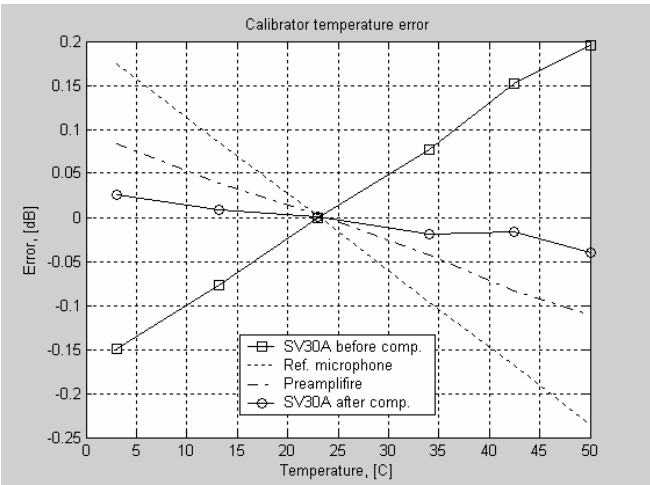

Fig. 5. The exemplary thermal characteristics of the calibrator before and after the compensation of RM and its preamplifier.

The pressure drift of the calibrator was investigated for the whole range of the pressures required by the standard PN-EN 60942:2003: 650 hPa - 1150 hPa. The exemplary pressure characteristics of the calibrator and corresponding RM are presented in Fig. 6.

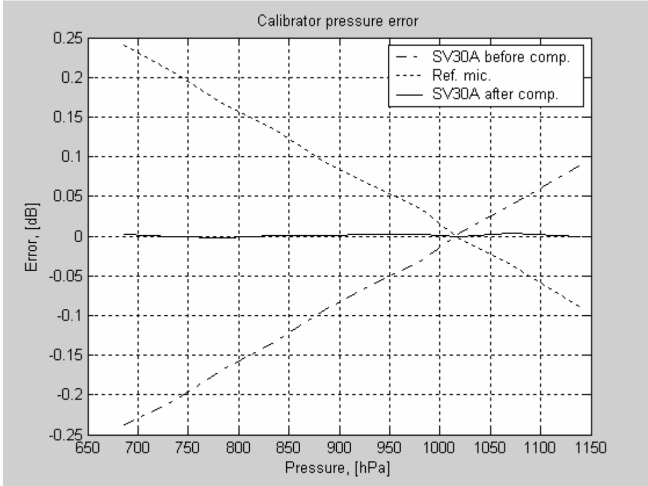

Fig. 6. The exemplary pressure characteristics of the calibrator before and after the compensation and the RM pressure characteristics.

The obtained results of the experiments prove the statement formulated in Section 3.4 that the characteristics of the calibrator are more-or-less opposite to the RM characteristics. The slight differences in the calibrator's thermal characteristics can be explained by the influence of the ambient temperature on the electronic circuits of the calibrator's input channel. This influence is relatively small if compared to the temperature influence on RM, thus the differences between the characteristics are not very significant (exact to a sign). In the case of the pressure characteristics, there are not any differences between the calibrator and RM. The atmospheric pressure does not influence the electronic circuits in the signal controller's input channel.

Additionally, the accuracy of the generated frequency and the level of the total harmonic distortions (THD) of the acoustic wave, generated in the chamber, were investigated. The accuracy of the frequency is related with the accuracy of the clock used in the circuits of the signal controller. The relative error of the generated frequency is not less than 0.02 % while the accuracy of the used oscillator is at the level of 100 ppm. The investigations of the THD were performed using the Type 1 sound analyser working in the FFT mode.

The following results have been obtained: the level of the harmonic distortions (THD) for 114 dB of the acoustic pressure in the calibrator's chamber is less than 0.75%, and for 94 dB is less than 0.25%.

5. DISCUSSION

The construction of the acoustic calibrator presented in the paper has some strong and some weak points. The advantage of the calibrator with the digital loop of the acoustic pressure level stabilisation in comparison to the classic analogue solutions (or any other without the digital loop [1], [2], [3]) is based on the possibility of the introduction to the calibrator's memory any thermal and pressure characteristics by means of the correction coefficients. In such a way any type of the microphone can be compensated if it is necessary.

It is possible to easily design a multi-range and multi-frequency acoustic calibrator without changing the hardware of the unit. The development of such a new unit is possible now by replacing the control programmes of the controllers. The application of the condensed reference microphone enables one to precisely copy the acoustic pressure levels in the calibrator's chamber in the very wide range of the atmospheric conditions. The calibrator with the digital stabilisation loop has very good stability of the frequency of generated signals. The long- and short-term stability, as well as the stability in the wide temperature and pressure range, depends on the parameters of quartz oscillator used in the unit. In addition, the strong point of the presented solution is the possibility of the production automation, including the calibration for different temperature and pressure values.

The weak point of the described calibrator is a rather complex construction which leads to the complicated production process. Additionally, the application of the condensed reference microphone in the feedback loop increases significantly the costs of the calibrator. But, on the other hand, it is worth that the calibrator is the generator of the standard test signals whose most important feature is the precision and the stability; thus the cost is not the most important factor.

6. CONCLUSION

The presented new principle of the acoustic calibrator design has been verified in practice. The received results of testing and achieved parameters of the device have confirmed that the proposed approach is correct and very promising. The application of the microprocessor for controlling the calibrator and the use of the feedback loop for that purpose enables the wide-range compensation of the temperature and pressure influence on the stability of that device. The coherent sampling eliminates the necessity of analogue filters and will facilitate in the future the development of the multi-frequency calibrators which up to now do not exist on the market.

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