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Digital-Filter-Based Compensation of Case Effect in Sound-Level Meters

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***Abstract*—The methodology for the design of a digital filter, which should compensate the effect of reflections and diffraction from the sound-level meter's casing (the so-called case effect), is presented. The coefficients of the family of the finite impulse response (FIR) filters, which were selected to fulfill the requirements of the compensation, were obtained in the MATLAB environment using the Remez algorithm. The frequency response of the selected designed filter are given. The chosen FIR filter was implemented in an on-chip Enhanced Filter Coprocessor of a fixed point 24-bit digital signal processor of a sound-level meter.**

***Keywords*—Digital compensation, DSP applications, FIR filter design, Remez algorithm.**

I. Introduction

A SOUND-LEVEL METER (SLM) is a handheld instrument used for measuring different parameters of sound, [3]. The current SLM American standard (ANSI S1.4-1983, [13]) and the former European, (IEC 60651, [12]) specify four types of SLMs: three grades of instruments – Type 0, Type 1 and Type 2 – and special-purpose Type S [15]. For each type accuracies applicable to a particular use are determined:

- Type 0 means laboratory standard instrument intended for use in the laboratory as a high-precision reference standard; it is not required to satisfy environmental requirements for a field instrument.
- Type 1 means precision instrument intended for measurements in the field and laboratory environment (roughly speaking, the errors of measurements performed by a Type 1 SLM will not exceed 1 dB).
- Type 2 means general-purpose instrument with more lenient design tolerances than Type 1, intended for general field use, particularly in applications where high-frequency (over 10 kHz) sound components do not dominate (estimated errors should not exceed 2 dB).
- Type S means special purpose instruments which have design tolerances associated with any of the three grades, but is not required to contain all of the functions stipulated for a numbered types.

The European standard (IEC 60672:2002, [4]) divides SLMs into two classes (so-called 'types'). The both classes have the same areas of application but differ in tolerances. Class 1 instruments have a wider frequency range and a tighter tolerance than a similar, lower cost, Class 2 instruments.

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Fig. 1. Different shapes of sound-level meters.

Almost all contemporary SLMs are based on a digital components and controlled by the digital signal processors where different noise weighting filters (simulating the frequency response of the human ear to unwanted sounds, [11]) required by the standards defining methods and sound parameters to be measured, are implemented.

SLMs, available on the world market, appear in various shapes and forms as it can be seen in Fig. 1.

However, they have some common features. In particular, they have a pointy bit at the top to stop the sound reflecting back at the microphone. Additionally, they have the microphone on an extension, containing usually preamplifier, to get it away from their casing, again to reduce reflection and diffraction effects (cf. Fig. 2).

One of the more difficult questions related to selecting an SLM is "How do you know if it complies with its claimed standard"? The standard IEC 61672 (part 2) tries to answer this question using the concept of "pattern approval". A manufacturer has to supply instruments to a national laboratory which tests one of them and if it meets its claims then the laboratory issues a formal Pattern Approval certificate. In EU the most common (and the most rigorous) approval is often considered to be that from the Physikalisch-Technische Bundesanstalt (PTB) in Braunschweig, Germany.

Fig. 2. Top of an SLM and a microphone placed on an extension.

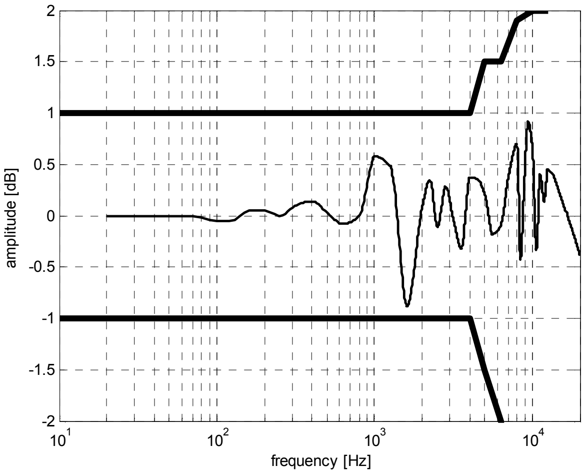

Fig. 3. Case effect in a sound-level meter: the characteristic of the compensated SLM (thin line) and the tolerance limits (thick lines).

As it was written above, for a Type 1 (Class 1) instrument all measurement results should be placed within the 1-dB band. There are plenty of sources of errors [2]. One of them is the SLM's casing which is the source of the reflections and diffractions of the measured signal. The exemplary case effect in a real-world SLM, taken under consideration, is presented in Fig. 3: the frequency characteristic is within the limits for Type 1 instrument, but it has to be taken into account that the case effect is not the only one which occurs. There are also effects coming from humidity, magnetic and radio-frequency fields, electrostatic discharge, ambient pressure, temperature, vibrations, etc. So, the case effect should be minimized and the characteristic should be as flat as possible in order to not exceed the Type 1 limits in the worst case with the mentioned-above influencing parameters.

II. Design Methodology

The idea for reducing the case effect is to design a compensation filter whose frequency characteristics are inverse to those produced by the case effect. The superposition of the filter characteristic and the characteristic produced by the case effect should result in a flat characteristic whose ripples are drastically reduced. The assumption was made that the characteristics obtained after the implementation of the designed filter should be compressed to the 0.1 dB band, i.e. the ripples of those characteristics should not exceed 0.1 dB.

The filter was designed in three steps:

first, the calculated coefficients of the selected FIR filter were implemented in the SLM,

next, the obtained frequency response was measured,

finally, the filter was redesigned.

As the starting point, the real-world measurement values were taken for 38 frequency values corresponding to the centre frequencies of the third-octave filters commonly used in the field of acoustics [16]. In Table I those values for the frequency range from 31.5 Hz to 20 kHz are presented.

TABLE I
Effect of Reflection and Diffraction from Meter's Case

Frequency [Hz]	Case effect [dB]	Frequency [Hz]	Case effect [dB]
31.5	0.00	63.0	0.00
100.0	-0.05	125.0	-0.05
160.0	0.05	200.0	0.05
250.0	0.00	315.0	0.10
400.0	0.14	500.0	0.03
630.0	-0.08	800.0	0.00
1000.0	0.58	1250.0	0.49
1600.0	**-0.88**	2000.0	0.08
2240.0	0.35	2500.0	-0.11
2820.0	0.29	3150.0	-0.03
3550.0	-0.32	4000.0	0.37
4470.0	0.36	5000.0	0.21
5600.0	-0.18	6300.0	-0.10
7100.0	0.40	8000.0	**0.70**
8400.0	-0.43	8900.0	0.34
9400.0	**0.92**	10000.0	0.67
10600.0	-0.33	11200.0	0.40
11900.0	0.14	12500.0	0.45
16000.0	0.08	20000.0	-0.38

The frequency values for which the measurements were made are defined by the IEC 61672 standard. In Table I the longest occurring values are bolded. They are inside the tolerance limits, but taking into account all possible effects which could affect the amplitude-frequency characteristic of a sound-level meter, they should be much smaller.

First, these values of the characteristic were interpolated by a cubic spline with the 10-Hz step. This trial was not successful because the number of points was not sufficient. So, some values resulting from local interpolation were added to the frequency characteristic as it is shown in Fig. 4.

With those additional points the spline interpolation was repeated, and its result did not contain undesirable oscillations.

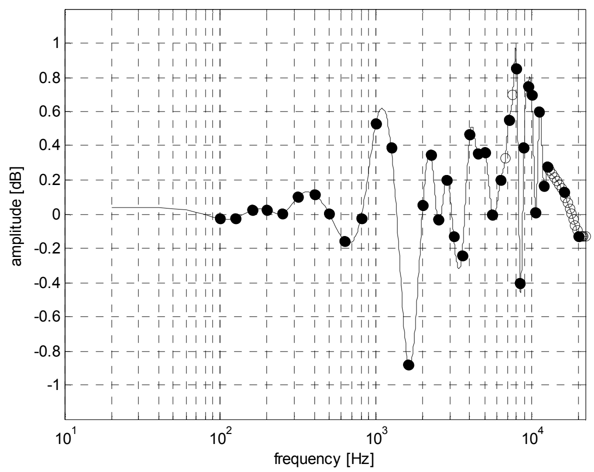

Fig. 4. Measured (black circles) and added points (white circles) of the frequency characteristic of the case effect and the spline interpolation (line).

The obtained cubic spline $A(f)$ was used as the basis for the calculation of parameters of the digital compensation filter whose characteristic has the form:

$$A_F(f) = \frac{1}{A(f)} \tag{1}$$

For computing the filter parameters, the Remez algorithm [10], also called the Remez exchange algorithm, was used. It is an application of the Chebyshev alternation theorem on the best polynomial approximation of certain functions under a number of conditions [1]. This algorithm in fact goes a step beyond the minimax approximation algorithm to give a slightly finer solution to an approximation problem.

The calculations were performed in the MATLAB environment using the **firpm** function which enables one to design a linear-phase FIR filter using the Parks-McClellan algorithm [8]. The latter algorithm uses the Remez exchange algorithm and Chebyshev approximation theory to design filters with an optimal fit between the desired and actual frequency responses. The filters are optimal in the sense that the maximum error between the desired frequency response and the actual frequency response is minimized. Filters designed in this way exhibit an equiripple behavior in their frequency responses and are sometimes called *equiripple* filters. The **firpm** function exhibits discontinuities at the head and tail of its impulse response due to this equiripple nature [6].

The characteristics were calculated for the filters of different orders from 64 to 128 and for each of them the maximum deviation from the 0 dB line was determined. In order to attain the assumed level of the ripples, the error in the linear scale should not be greater than 0.01. Such value was achieved for the FIR filter of order 117 as it is seen in Fig. 5.

After the implementation of the compensation 117-order filter in the SLM, the frequency response was measured once again. The experiment showed that for two frequencies the divergence between the expected and obtained results were excessive.

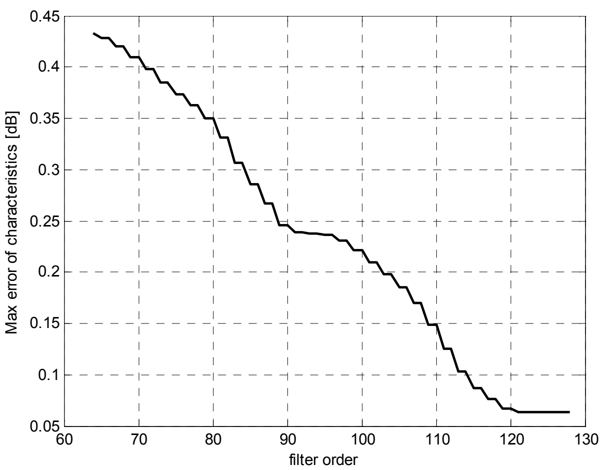

Fig. 5. Maximum error in the function of filter's order.

The probable reason of that was the local linear interpolation, performed in order to obtain the greater number of points necessary for spline interpolation. Thus the $A_F(f)$ characteristic was modified in such a way that for the frequencies where the ripples were too large the opposite values were added.

The filter was redesigned for such modified characteristics and after the implementation in the digital signal processor of the SLM the obtained results were satisfying

III. Obtained Results

The designed filter was implemented in the on-chip Enhanced Filter Coprocessor (EFCOP). The EFCOP [7] is a general-purpose, fully programmable filter coprocessor that operates concurrently with DSP core operations and requires minimal CPU intervention. The EFCOP has dedicated modes of operation for performing real and complex FIR filtering, adaptive filtering and multichannel FIR filtering. The usage of the EFCOP increased significantly the overall performance and efficiency of DSP56307 [5] which was placed in the compensated SLM.

The implementation of the FIR filter in this EFCOP is very simple since it requires only to place the calculated coefficients in the selected part of the coprocessor's memory. EFCOP performs filtering in the shade without any consumption of the calculation power of the main processor [10].

In Table II the values of the obtained compensated characteristic are given. It is visible that the largest obtained values are well inside the tolerance limits for Type 1 instrument, and the obtained frequency response is much flatter and does not contain undesirable ripples.

TABLE II
Characteristics after Digital Compensation

Frequency [Hz]	Case effect [dB]	Frequency [Hz]	Case effect [dB]
31.5	0.00	63.0	0.00
100.0	-0.11	125.0	-0.11
160.0	-0.02	200.0	-0.03
250.0	-0.08	315.0	0.03
400.0	0.11	500.0	0.07
630.0	0.04	800.0	0.00
1000.0	0.13	1250.0	0.03
1600.0	-0.08	2000.0	0.07
2240.0	-0.09	2500.0	-0.13
2820.0	0.18	3150.0	0.01
3550.0	0.00	4000.0	-0.14
4470.0	-0.02	5000.0	-0.06
5600.0	-0.21	6300.0	**-0.39**
7100.0	-0.25	8000.0	-0.21
8400.0	0.00	8900.0	-0.05
9400.0	0.16	10000.0	0.07
10600.0	**-0.38**	11200.0	-0.28
11900.0	-0.03	12500.0	0.27
16000.0	-0.01	20000.0	-0.18

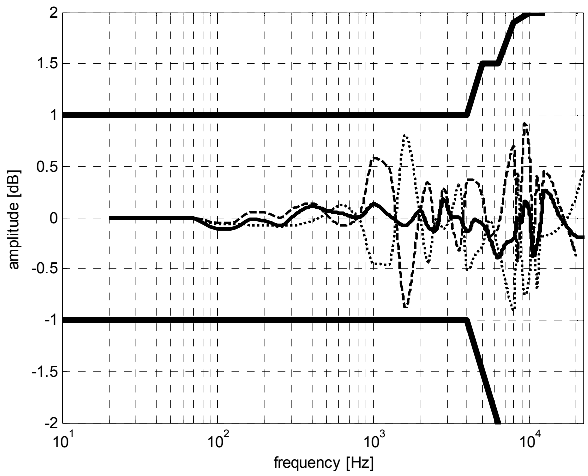

Fig. 6. The characteristics before the compensation (dashed line) and after the compensation (solid line); the characteristics of the compensation filter dotted line) and the tolerance limits (thick lines).

In Fig. 6, various frequency characteristics are compared: the SLM characteristic without compensation, the characteristic of the designed inverse FIR filter of order 117, and the final SLM characteristic which is within the 1-dB band and fulfills the requirements for Type 1 instrument.

IV. Implementation

The compensation filter was incorporated in the software of an SLM [17] presented in Fig. 7.

The compensation filter should be activated when the measurements are performed with the microphone placed on the preamplifier attached directly to the instrument's input. The filter should be switched off when the microphone with the preamplifier are attached by means of a cable and the measurements are performed on a tripod (cf. Fig. 8).

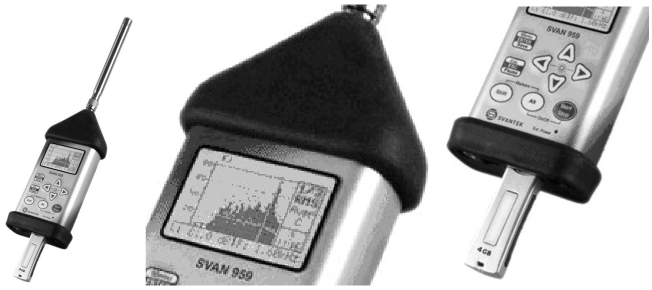

Fig. 7. SLM with the implemented designed compensation filter.

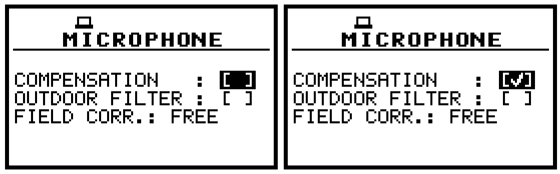

Fig. 8. Activation and deactivation of the compensation filter in the SLM.

V. Conclusion

The approach to the problem of the digitally-based filter compensation seems to be correct as the SLM in which the designed filter was implemented passed successfully the PTB approval (marked as 21.21/09.03).

The obtained positive results encouraged the authors to implement similar approach for the design of another compensation filters dedicated for a specially mounted microphones of the dosimeters.

References

[1] A. Antoniou, "Efficient Remez algorithms for the design of non-recursive filters", University of Victoria, 2003.
[2] L. Beranek, "Acoustical Measurements", Acoustical Society of America, 1988.
[3] D. A. Bies, C. H. Hansen, "Engineering Noise Control, Theory and Practice", 3rd ed., Spon Press, New York, 2003, pp. 1-472.
[4] *Electroacoustics, Sound level meters*, International Electrotechnical Commission, IEC 61672-1, Geneva, 2002.
[5] *DSP56307 User's Manua*l, Motorola, 1998.
[6] *MATLAB Help*, 1984-2009 The MathWorks, Inc.
[7] D. G. Minic, "Applied Matrix Multiplication With the DSP563xx Enhanced Filter Coprocessor (EFCOP)", Application Note AN2691, Freescale Semiconductor, 2005.
[8] *Programs for Digital Signal Processing*, IEEE Press, New York, 1979, Algorithm 5.1.
[9] D. R. Raichel, "The Science and Application of Acoustics", Springer, 2006, p. 53.
[10] T. M. Redheendran, "Programming the DSP56300 Enhanced Filter Coprocessor (EFCOP)", Application Note, APR39, Freescale Semiconductor, 2005.
[11] E. Ya. Remez, "Sur le calcul effectif des polynômes d'approximation de Tschebyscheff." *C. P. Paris*, 1934, pp. 337-340.
[12] *Sound Level Meters*, International Electrotechnical Commission, IEC 60651, Geneva, 1979.
[13] *Specification for Sound Level Meters and Supplement*, American National Standards Institute, ANSI S1.4-1983 (R2006)/ANSI S1.4a-1985 (R2006),.
[14] *SVAN 959 Users Manual*, App. C, SVANTEK, 2009.
[15] *The Noise Manual,* Rev. 5th ed., American Industrial Hygiene Association, Fairfax, 2003, pp. 41–100.
[16] K. Veggenberg, "Octave Analysis Explored: A Tutorial", EE-Evaluation Engineering, 2008.