PRACTICAL APPLICATION OF A METHODOLOGY FOR DETERMINATION OF THE SOUND CONTRIBUTION OF INDUSTRIAL SOURCES OF NOISE IN THE ENVIRONMENT

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Abstract. *The principal challenge found in the control of environmental noise consists in identifying which sources of sound power will have to be treated so that the levels of noise measured in control points attend to the values demanded by the legislation in force. Being so, the professional of the area must be clever identifying different sources of sound power and its contribution at the receiver point. However, the precision of computational-numerical models for solution of these problems is in the range of +/-5 dB. Having in mind that errors of this magnitude are inadmissible for noise control agencies, this work takes as an objective the quantification of noisy sources coming from an industrial plant propagating to the exterior environment in a real situation without approximations or modeling adjustments. The used methodology consist of the Function Transfer (FT) determination between the noise source of interest (located inside the industry) and the receiver point (outside the plant industry), using adaptive filtering LMS (Least Mean Square) together with a reference noise source (sinusoidal sweep type signal). An experimental proceeding is presented in this work. Numerical simulations showed that the filtering capacity of LMS is compromised when the energy level of the reference source is reduced while compared with the energy produced by other sources at the receiver point. In order to go around this restriction, a methodology using least squares adjustment and time-frequency analysis (distribution of Choi-Williams) is presented. The obtained results proved the validity of the methodology, managing to quantify the sound level of a determined source with error below to 2 dB.*

Keywords*: Environmental noise, Noise source, Time-frequency analysis.*

1. INTRODUCTION

The population exposition to the surrounding noise, particularly in the urbane zones, has deserved a growing attention by a great part of the community. One of the main concerns is to analyze the industrial noise influence in the exterior environment quality where the people circulates and remains. This kind of noise is the biggest cause of claims in the competent agencies of the cities by the population (Handley, 1995).

Actually, problems with the internal and external noise surround the factory have been a constant. However, this works goal is the industry external noise propagation.

The principal challenge found in the control of environmental noise consists in identifying which noise source will have to be treated, so that the noise levels at measured points of control must attend to the values demanded by the legislation in force. The great facing problem resides in the fact that a middle size plant is composed by thousands of noise sources, each one contributing to the total noise level measured in the neighboring area. This type of problem is characterized like propagation of external noise (Gerges, 2000).

The methodology of reduction of the noise begins through effective methods of acoustic identification of the properties of noise source and the environment. For such methodology, the noise source must be firstly identified, followed by the identification of the transmission path. The spectrum of the acoustic sign is also very useful for identification of the noise source (Bell and Bell, 1993). This information is then analyzed in order to propose an effective solution for the noise reduction problem (Barber, 1993).

However, only the analysis of the spectrum is not sufficient, as it is common in plants and industrial environments the existence of coherent noise sources (different noise source have determined quantity of acoustic energy in the same frequency). Therefore there is a necessity of developing new techniques for identification and quantification of the noise in the receiver point that manages to remove this problem.

Another important characteristic of the industrial environments is that the noise sources need to operate simultaneously, not being possible to be turned off or moved for study of its acoustic contribution to the analyzed sound field.

In this context, researches have been made since the middle 70´s, what concerns the identification and quantification of the contribution of noise sources. Techniques, such as, Partial Coherence Function – PCF, developed by Koss and Alfredson (1974), sensors network (known as microphones array) applied to countless methodologies (Nishiura et al., 2002; Juhl et al., 2002); and beam forming has been gaining expressive importance in the acoustics field. However, the beam forming has been improving quickly (Chen and Hudson, 2002).

Keeping in mind the amount of new technologies in the market, some of them having an elevated cost, such as, microphone array, the goal of this work is to develop a methodology with acceptable cost, using a minimal number of sensors, but still being able to quantify noise sources in the external environment.

2. METHODOLOGY

This work intends to develop a methodology which quantifies the noise source contribution in external points of the industrial plant. These external points are used as noise environmental control. In this way it is possible to estimate the noise source influence on the total noise level measured at these specific control points. The significance of these estimate results leads to the necessity of identifying the noise sources of an industrial plant that will have to be acoustically treated, doing so the pressure levels measured externally will be below the allowed limits of the legislation.

In general, the industrial plants have these characteristics:

- Existence of countless noise sources, which contribute to the total SPL measured at an external receiver point;
- The noise sources can not be turned off due to continuous production process;
- Presence of coherent noise sources;
- The noise path way to the external receiver point is complex, i.e., with a presence of acoustic rays reflections.

Keeping the cited characteristics, the objective is to determine the transfer function between the noise source which we wants to quantify and receiver point located externally to the industrial plant.

In order to estimate the transfer function, an adaptive LMS – Least Mean Square is used in conjunction with a reference noise source which produces a sine sweep signal. The use of a sine sweep is justified once it is not correlated with any other noise source in the system, and the sweep covers the entire frequency range of interest.

2.1 – LMS filter application in the quantification of noise sources

The actual methodology proposed intends to use adaptive filters to estimate the transfer function between the noise source of interest and the receiver point.

For a better understanding of the used methodology Fig. 1 is presented with an adaptive LMS filter.

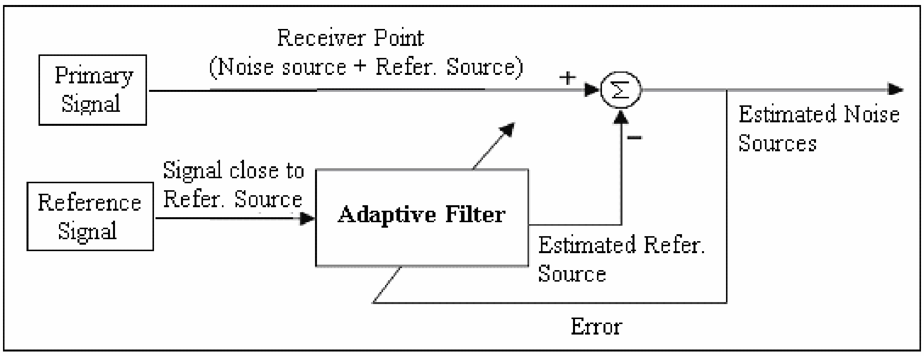

Figure 1 – Schematic of the adaptive LMS filter used in this work.

From Fig. 1 one can notice that the primary signal is constituted of signal at the receiver point (specific point) together with the contribution of all the other noise sources (the study area), including the noise originated from the reference noise source. The reference noise source is added to the system and positioned (approximately one meter) near to the noise source of interest, i.e., the measured signal has noise contribution from this source only.

Using the LMS filter we obtain the estimative of the will be the signal originated from noise reference source in the receiver point. See Fig. 1. With this signal and the estimated signal near to the noise reference source it is possible to estimate the transfer function between the noise source of interest and the receiver point external to the plant.

The reference noise source generates a sine sweep signal that is characterized by its frequency variation as time changes. Therefore the use of adaptive filter is justified since it has the capacity to adapt to the signal variations occurred along the time.

However, as the reference source has signal that changes in power with respect to time, a normalized LMS algorithm is used once it has a major ability of step adaptation if compared with the RLS – Recursive Least Square algorithm.

3. RESULTS AND DISCUSSION

Prior to this stage a sensibility analysis and numerical simulations were performed in order to identify the LMS filter applied to the adaptive noise cancellation.

After the sensibility analysis, where several configurations of primary signal and reference, were analyzed varying the energy levels, we conclude that as the reference noise source energy level reduces in the primary signal the quality of the filtered signal is compromised, being unsuitable for analysis purpose.

Based on the numerical simulations we noticed that for the presence of coherent sources in the model or in the simulated system, the obtained results using the filter has a good signal to noise relationship, about 17 dB. Thus we obtained at the receiver point the noise source signal only, once the reference noise source signal is not correlated to any other noise source present in the system.

The experimental procedure has the objective to validate the numerical simulations as cited before. Knowing the results of the sensibility analysis a methodology in the experimental procedure was developed to reduce the energy level originating from the noise sources, but keeping the energy level of the reference noise source.

To validate the simulations the experiments were carried out in an acoustic free field. The experimental apparatus was compose of a noise sources which were loudspeakers reproducing recorded noise source of an industrial plant. The sound was propagated in a path way that reaches a microphone located at the receiver point.

Three types of noise were recorded in a CD: 1) noise of a exhaust fan; 2) pink noise; 3) sine sweep in Matlab® environment through a chirp function that was converted to a WAV file.

3.1. Adaptive filter parameters adjusting

Before beginning the experimental procedure one have to determine the filter parameters for the sine sweep, in such a way that the filter has a good performance and convergence. The best parameters of the adaptive filter: L – Number of weights, μ – convergence factor and α – Forgetting factor; were determined trough numerical simulation in the Matlab®.

It is emphasized that during the numerical simulations was noticed that the adaptive filter parameters were dependents on the sweep characteristics, mainly sweep speed in the frequency range of interest.

To adjust the LMS filter parameters the simulated environment was a free field with one noise source being the sine sweep located near to the noise source that is to be identified.

A time-frequency diagram of the sweep used in the experimental proceeding can be seen in the Fig. 2.

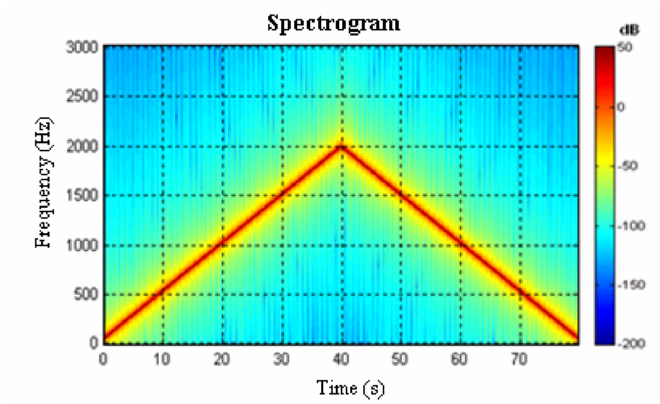

Figure 2 - Time-frequency diagram of the sweep used in the experimental proceeding.

A random optimization was used to determine the parameters μ, α and L used in the adaptive filter. The objective function was to minimize the error between estimated sweep (obtained via adaptive filter) and the theoretical (obtained numerically) one at the receiver point.

This way the optimized parameters found were: $L = 20$; $\mu = 0.05$ and $\alpha = 0.02$.

We can now test these optimized parameter in practical way in a free field propagation experiment.

3.2 - Free field

A schematic representation of the experimental setup is shown in Fig. 3.

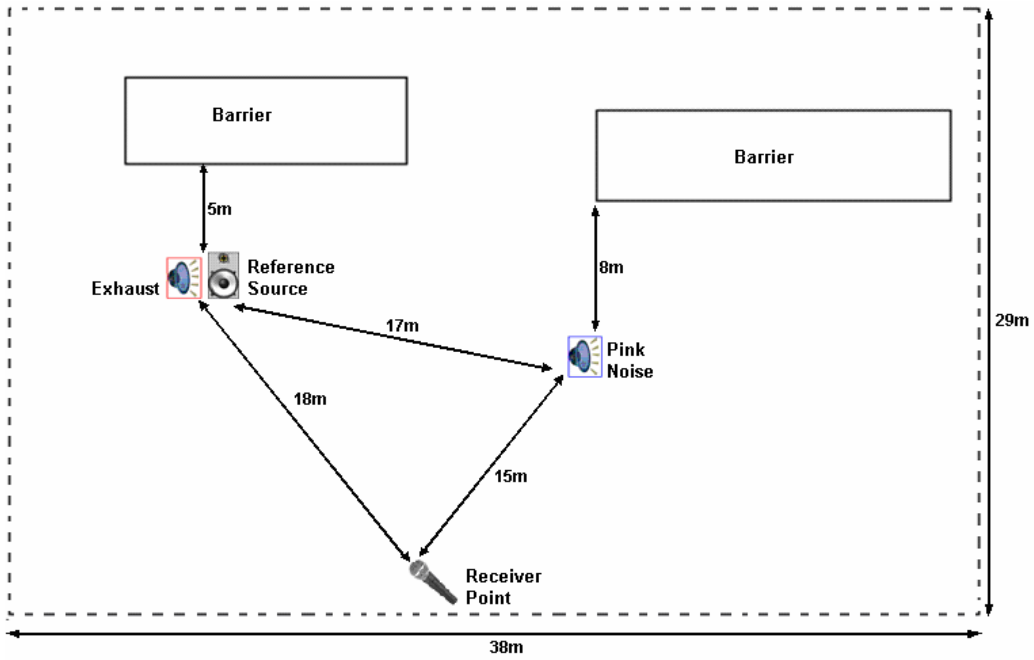

Fig. 3 – Schematic representation of the experimental plant.

Based on the previous results of sensibility analysis, the reference noise source signal measured at the receiver point should have a higher energy level when compared with the other noise sources that reaches the receiver point.

In practice, it is not always possible to obtain this configuration of noises sources. Another fact is that when the energy level of the sine sweep is much higher than the background noise (difference higher than 6 dB), the use of filter would be not necessary based on the acoustic fundaments. This way, a methodology was developed aiming to reduce the noise energy level in the surround area but keeping the same energy in the sine sweep.

To apply the methodology developed here, the first step is to collect several measurements at the receiver point (for our experiment 10 data acquisition were realized), when all the noise sources are turned on, and contributing to the total noise level. Subsequently, do a measured data average once the noise originated from the source will be diminished based on signal cancellation due to phase differences. The energy level of the sine sweep must be maintained, thus a second methodology should be applied.

The microphone was fixed in a tripod to avoid any type of disturbance in the path way between the transmitter and the receiver.

The result of the average with ten measurements carried out in the point receiver, with all the noise source turned on, is shown in the Fig. 4, where the spectrum of the signal averaged is shown.

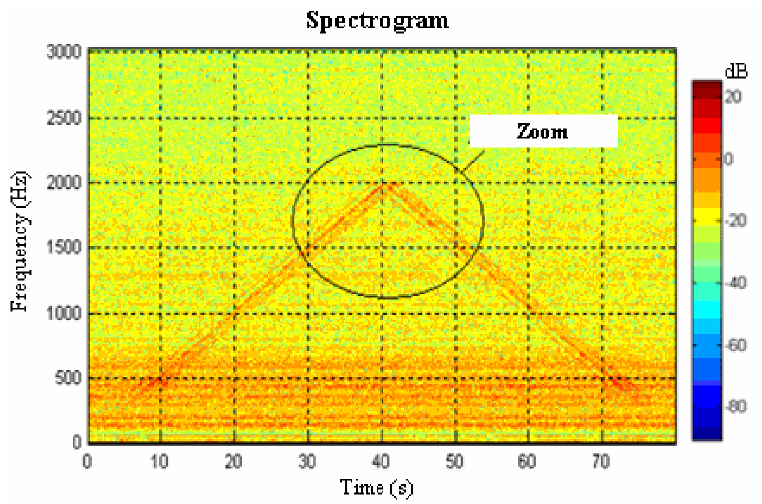

Fig. 4 – Averaged spectrum of ten acquisitions at receiver point.

Looking at Fig. 4 one notice that the sine sweep does not have the same phase in ten obtained data acquisitions, since there is a delay between the sweeps in the averaged signal. In other words, the averaged signal has the contribution of several sweeps as can be seen in Fig. 5. Such phase differences are due to handling the acquisition system at initial time of acquisition data.

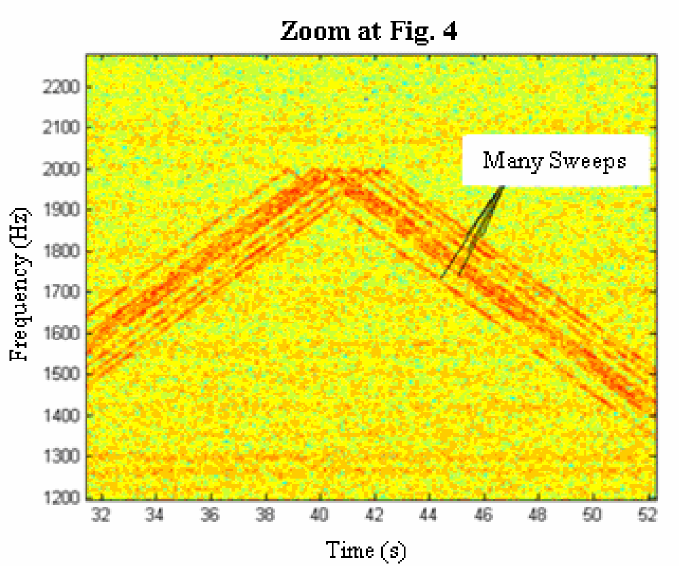

Fig. 5 – Zoom applied in the Fig. 4.

In order to the averaged measured signals at the receiver point has the energy level reduced maintaining the same sine sweep energy level, it is necessary that the sine sweep starts at the same instant of time. Thus is necessary to correct the time delays between the sweeps. The correction is obtained from the sweep that has a bigger starting measurement time as a reference. Hence the other sweeps are corrected to have the same time delay.

For this procedure a least minimum square is used in the time-frequency diagram to obtain a straight line in 50 Hz (initial frequency of the sweep) frequency that intercepts time axis at x time.

Applying this procedure the time-frequency diagram needs a very good time resolution, i.e., a raising time of 1/8192 seconds, so that one can obtain the exact initial sweep time. This time resolution can not be obtained using a spectrogram, but with a Choi-Williams time-frequency distribution.

To apply the Choi-Williams distribution in the entire signal has a very high computational cost, therefore

It can be applied to a determined part of the signal and the time-frequency diagram is obtained as can be observed in Fig. 6.

From a time-frequency diagram as the one showed in Fig 6, it is possible, using least minimum square to interpolate a straight line trough selected points. These selected points on the straight line, marked in red, in the 360 Hz up to 420 Hz frequency range (see the blue circle). Thus it is possible to estimate where the straight line intercepts the x-axis (time), correspondent to the initial sweep frequency, 50 Hz. Carrying out this procedure for each one of the ten measurements we can obtain the initial time for each sine sweep. Thus, with these delay times a new average is computed with the corrected signal, i.e., with the delay removed. The final result is shown in Fig. 7.

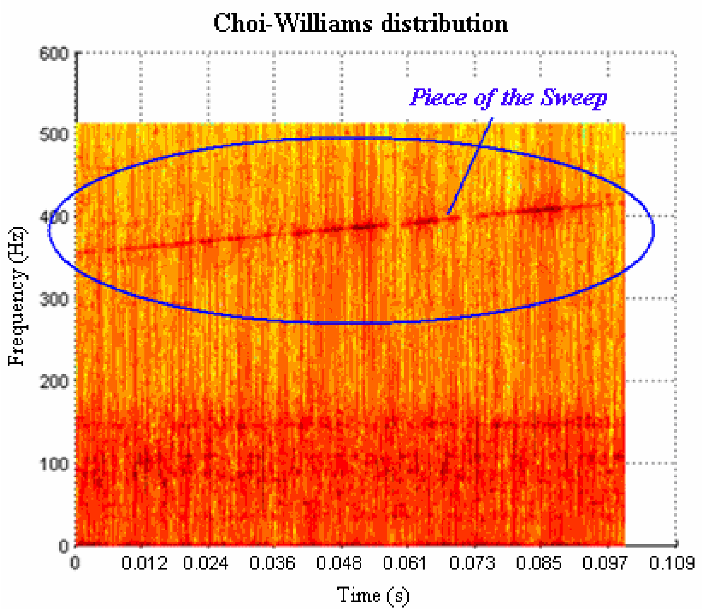

Fig. 6 - Time-frequency analysis using Choi-Williams distribution.

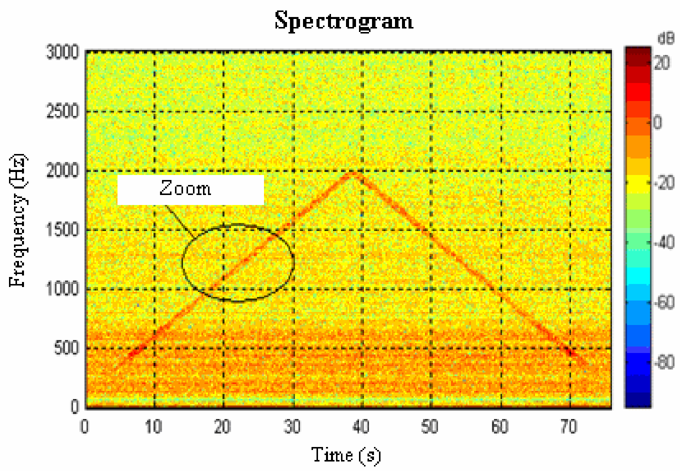

Fig. 7 – Averaged measurement after phase correction via Choi-Williams and least minimum square adjustment.

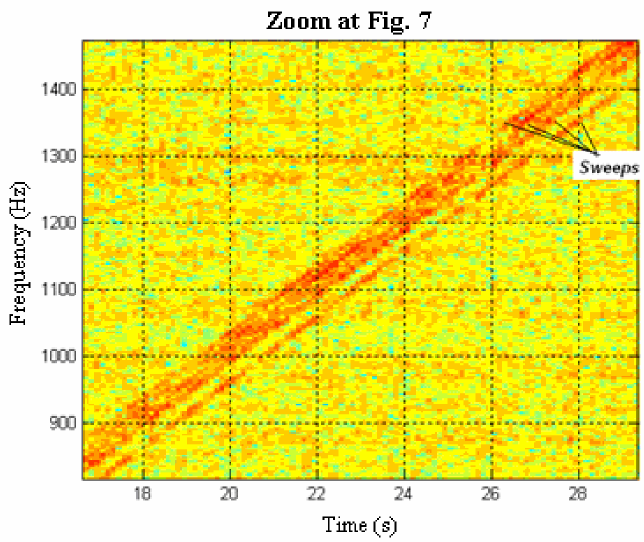

Fig. 8 – Zoom applied to Fig. 7.

The cepstrum technique which has as an objective the repeatability detection in the signal is used to eliminate phase differences between sweeps. In acoustics, cepstrum is mostly used to echoes detection. Using cepstrum it is possible to calculate the time delay between two signals with our desired resolution which is of 1/10 of milliseconds.

After the least minimum squared adjustment applied, one knows which one has the biggest initial time, i.e., which one has the biggest time delay among the measured signals. For cepstrum application on these measurements, the one which has the biggest delay was used as reference to calculate the other time delays.

Fig. 9-a shows an spectrum of the signal acquired at the receiver point, and Fig. 9-b an spectrum of the resultant averaged signal with the proposed methodology applied.

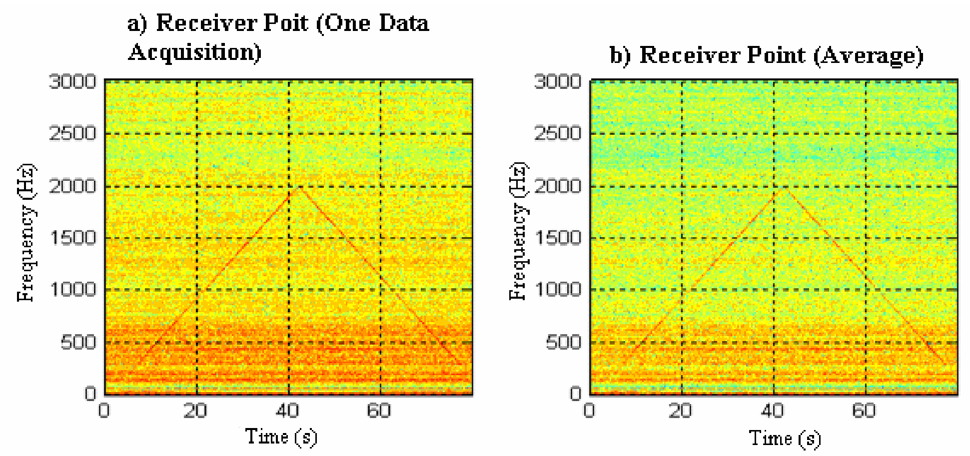

Fig. 9 –Time-frequency diagram: Spectrum a) From a data acquisition at receiver point; b) Average of ten acquisitions at the receiver point.

It is observed in Fig. 9-b that the sound pressure level at low and high frequencies are reduced for noise sources 1 and 3 (exhaust fan and pink noise, respectively), as can be seen by the red color reduced when compared with Fig. 9-a.

Using the averaged signal measured at the receiver point as the "primary signal" as LMS input, the obtained result after the adaptive filter is shown in Fig. 10-a, and Fig 10-b shows the absolute error between measured sweep and the estimate one.

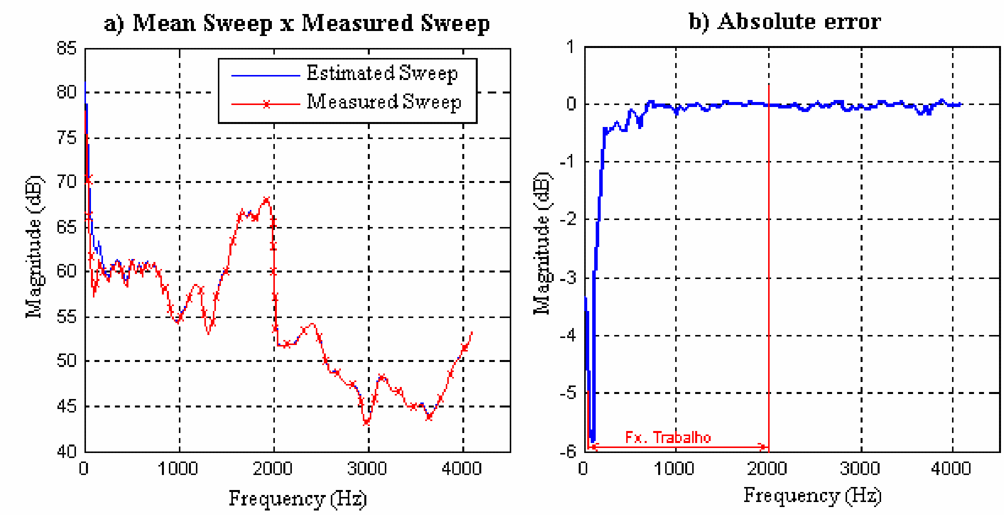

Fig. 10 – a) Spectrum of the mean estimated Sweep x measured Sweep. b) Absolute error between the signs of Figure 10-a.

It is observed in the Fig. 10 that the estimated sweep signal tends more towards the measured sweep in the low frequencies. However, the error still remains in the low frequencies, below 200 Hz, due to the energy level coming from other noise sources of the system when compared to the energy level in the sine sweep in the frequency range. To remove the energy level one option should be to increase the number of measured signals for average computation.

The transfer function between the receiver and transmitter points can be estimated using Eq. (1), once we already have the sweep filtered signal and the receiver point and the sweep signal near the generating noise source.

$$FT_{est} = 10\ \log_{10}(\bar{x}^2{}_{\mathrm{var}\,redura\_est}) - 10\ \log_{10}(\bar{x}^2{}_{Varredura\_prox}) \tag{1}$$

where: FTest is the estimated transfer function; $\bar{x}^2{}_{\mathrm{var}\,redura\_est}$ is the exponential averaged sweep at the receiver point with adaptive filter applied; $\bar{x}^2{}_{Varredura\_prox}$ is the exponential averaged sweep near the generating point. The exponential average is compute for a response time of 125 milliseconds.

The estimated function transfer with the Eq. (1) is used to compute the sound pressure level that reaches the receiver point due to the exhaust fan. Thus, with the obtained signal near to the exhaust fan and the estimated transfer function using the adaptive filter one can compute the exhaust fan noise contribution and the receiver point using Eq. (2).

$$Y(f)_{exaustor} = FT(f)_{est} + 10\ \log_{10}(X(f)_{prox\_exaustor}) \tag{2}$$

where Y(f) is the sound pressure level at the receiver point due to the exhaust fan contribution; FTest ist the estimated transfer function using Eq. (1); X(f) is the frequency domain signal measured near to the loudspeaker reproducing the noise of the exhaust fan.

The result obtained with Eq. (2), i.e., the exhaust fan noise contribution at the receiver point is shown in Fig. 11-a. The absolute error (measured SPL minus estimated SPL) of these signals is shown in Fig 11-b.

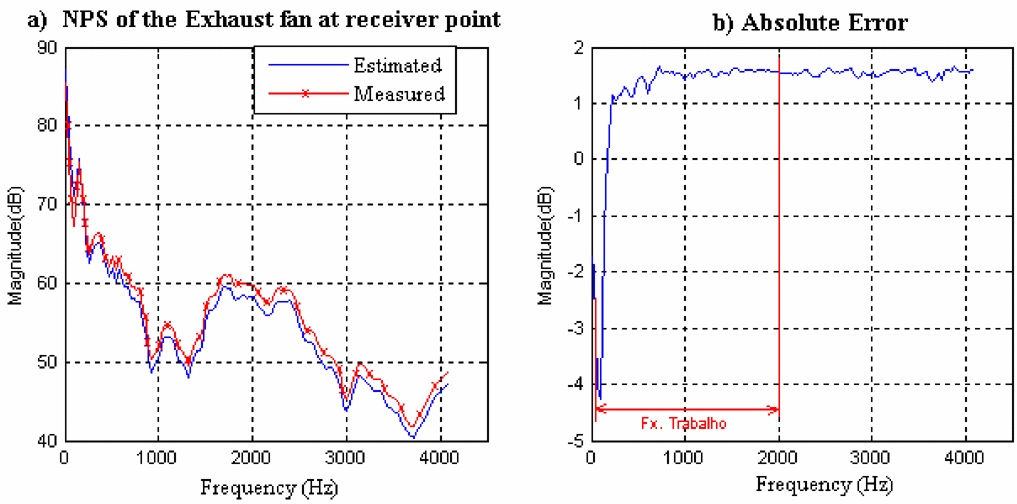

Fig. 11 - a) Noise contribution of the Exhaust fan (Source 1) at receiver point. b) Absolute Error.

Observing Fig. 11-b we noticed that the error obtained reaches values of 1.5 dB in the 200 Hz up to 2.000 Hz frequency range. The higher errors obtained below 200 Hz is explained in the sensitivity analysis. In this way, the obtained results are excellent for field measurements where acquiring errors in the order of ± 3 dB are satisfactory.

4. CONCLUSIONS

The main goal of this works is to develop a low cost methodology (using two sensors=microphones), with which it is possible to determine the noise contribution (quantify) of industrial noise sources at receiver points located in the circumambient community, trough the transfer function estimation between the noise source of interest and the receiver point using adaptive filter and a noise reference source (signal generator such as sine sweep).

The main conclusions of this work are:

1 – To use adaptive filter it is necessary to optimize the parameters such as: L - number of weights; μ – convergence factor and α – Forgetting factor; with a objective function that has a minimum error between estimated and theoretical signal. It is concluded that such parameters are function of the sweep speed used in the sine sweep.

2 – A methodology was developed to reduce the energy originated from the noise sources at the receiver point, but keeping the same energy level of the reference source. This methodology is constituted of averaging the measured signals at the receiver point, after a time-frequency analysis using the Choy-Williams distribution, LMS adjust techniques and Cepstrum calculations. Errors around 10 dB in the filtered estimated signals were obtained before applying this methodology were reduced to 2 dB.

3 - Experimental procedures are very well accomplished showing that the proposed methodology manages to quantify noise source level of determined sources with error of 2 dB, which are below the errors introduced by the instrumentations, i.e., ± 3 dB.

5. ACKNOWLEDGEMENTS

The authors would like to thank FAPEMIG (Research Support of Minas Gerais State Foundation), CNPq (National Counsel of Technological and Scientific Development) and FEMEC (School of Mechanical Engineering) at UFU (Federal University of Uberlândia) for their financial support.

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