APPLICATION OF LMS FILTER AND SYNCHRONIZED MEAN FOR NOISE SOURCE QUANTIFICATION IN ENVIRONMENTAL ACOUSTICS

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Abstract. *In the last twenty years, studies about the noise effects on hearing and the quality of life has received a great impulse, considering that noise can cause disturbances during sleep and on the health of human beings. The major parte of these studies analyzes the influence of industrial noise on the quality of the external environment where people circulate or live. This type of noise is one of the biggest causes of protest by communities at the competent agency of the cities. The methodology consists of the determination of Transfer Function (TF) between the noise source of interest and the receiving point using an adaptive filter, LMS (Least Mean Square), and a reference source generating sine sweep. It is possible to estimate the level of noise proceeding from the source that we want to quantify based on the estimated TF and the generated signal (acoustic pressure) near the noise source. This estimation can be made at a receiving point which can be internal or external to the industrial plant. Excellent results were obtained with numerical simulations. A numerical-computational simulation developed for control of environmental noise generates errors in the range of 5 dB. However, the method's efficiency is compromised when the level of energy of the reference source is reduced if compared with the energy generated by other sources at the receiving point. In this study, a methodology based on Synchronized Time Domain Mean STDM is proposed to minimize the influence of secondary sources on community received sounds. The proposed methodology was validated with numerical simulations and experimental results.*

Keywords: *noise sources quantification, environmental noise, LMS adaptive filter, Synchronized Time Domain Mean.*

1. INTRODUCTION

In the last few years, the concern with the quality of life increased significantly. It's known that this quality is affected by noise, which makes the search for acoustic comfort grow quickly. The most serious effects caused by noise will occur over time, such as deafness, which will be followed by mental imbalance and physical degenerative diseases. It's observed that in moderated levels of noise the human body tend to present stress, physical, mental and psychological disturbances, insomnia and hearing problems (Pimentel-Souza, 2000).

One of the main concerns is to study the influence of industrial noise on quality of the external surrounding where people circulate or live. This type of noise is one of the major causes of protest by the communities to competent organs (Handley, 1995).

With the invasion of industrial areas by residences, the concern of the security engineer, which previously was only with the worker in the factory, now is with the neighboring communities. Nowadays, it is very common for an industry to receive a notification asking to evaluate the noise around the factory once certain cities have laws that limit the permitted noise level (Nunes at al., 2007).

According to Coelho (1999), during the study of the project of a Noise Control Program, which should always be created by professionals or experts in this area of research, it is essential to predict the noise levels in work areas, due to noise produced by machinery and other equipments. The noise prediction in external surrounding permits the planning of the control of the industrial noise sources to be treated.

The Predicted Sound Propagation is a group of techniques used to predict the noise level in a given environment. The most popular techniques are based on classical propagation equations, ray tracing, boundary elements and finite elements methods.

Although the simulations are accurate for occupational noise control, the same doesn't happen for environmental noise (Nunes, 2006), once the distances are considerable (in order of kilometers), which implies: complex path of wave propagation, soil influence, weather effects, presence of vegetation, etc. Errors of until 5 dB are expected when using classical techniques to predict noise in communities. On the other hand, there are situations in which a reduction of 2 dB at a control point may be sufficient legally.

In her study, Nunes (2006) developed a low cost methodology to determine the sound contribution of industrial noise source in an external community control point, through Transfer Function (TF) estimation between the noise source to be quantified and the external receiver points, using an adaptive filter and a reference source (signal generation such as a sine sweep).

In her study, Nunes, observed that when the signal-noise ratio is reduced, the accuracy of the filter is strongly reduced. Her studies have show that the contribution of the reference source in noise level measured at control point needs to be, at least, 3 dBA higher than the background noise (contribution of all other noise sources).

This limitation of 3 dBA disables the use of the proposed methodology, once this condition is not achievable in real situations, either due to the overall Sound Power Levels (SWL) of the industrial plant or due to the distances between the reference source and the control point, where it is needed to estimate the contribution of the source studied.

In this paper a methodology based on fuzzy logic and pos-trigger signals which makes possible the use of Synchronized Time Domain Mean (STDM) to minimize the effect of background noise on the reference sound is shown. Therefore, the aim of this study is to permit the utilization of the Nunes's methodology (Nunes, 2006) in real applications.

2. METHODOLOGY

The methodology proposed by Nunes (2007) consists in estimating the Sound Transfer Function (TF) between a point near to the sound source to be identified (fan) and an external receiver control point (microphone) as show in Fig. 2. After the TF estimation, the contribution of the source being identified on the overall Sound Pressure Level (SPL) measured at the control point can be easily estimated after measuring the SPL generated by the source being identified at the near point.

In order to estimate the TF, a LMS (Least Mean Square) adaptive filter is used in conjunction with a reference noise source like 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. The reference source is placed close to the source being studied and the noise (at the receiver point) due to all sources present including the sine sweep is measured.

Finally, the TF between the two points (emitter and receiver) can be calculated using the following signals: sine sweep signal estimated close to the source being identified (at one meter of distance) and the sine sweep signal obtained at the receiver point (estimated through adaptive filtering).

The TF obtained by adaptive filtering can be used in predictive acoustics to quantify the source contribution at the receiver point. However, the source has to be located close to the reference source used to estimate the TF.

The adaptive filter uses the LMS algorithm to obtain the best estimation. Figure 1 shows the schematic of the adaptive filtering technique used (Widrow and Stearns, 1985).

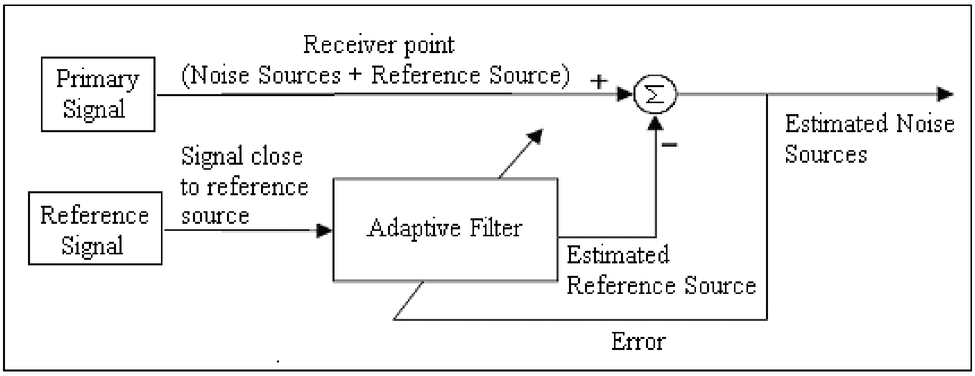

Figure 1. Scheme of adaptive filter

The primary signal which is obtained at the receiver point (previously located) is made of contributions from all sources of the system (or study area) including the reference source. The reference source added to the system should be located close to the source being identified. The reference signal is the reference source signal at a distance of 1 meter.

Initially, a sensitivity analysis to evaluate the capability of the filter and the quality of the filtered signal was carried out.

The analysis was based on Fig. 1. The primary signal, obtained at the receiver point, is composed of an arbitrary noise source (constructed by the model) and a sine sweep signal. The reference signal is obtained close to the reference source and is correlated with the sine sweep which is obtained at the receiver point. The reference signal is also a sine sweep signal. The filter output signal is an estimation of the reference signal (sine sweep) and the resulting signal of the process is an estimation of the level of the noise source of interest.

The filter parameters were defined after a random optimization using initial values of the bands specified by the theory of LMS filters.

The estimated transfer function (*TF* between estimated sine sweep with filter - at the receiver point - and the sine sweep measured near the generator – at the reference source location) is expressed in Eq. (2).

$$TF_{est} = 10\log_{10}(\overline{x}^2_{sweep\_filt}) - 10\log_{10}(\overline{x}^2_{sweep\_near}) \quad (2)$$

where $\overline{x}^2_{sweep\_filt}$ is the exponential average of the system input: the sweep estimated at receiver point with the adaptive filter, and $\overline{x}^2_{sweep\_near}$ is the exponential average of the system output: the estimated sweep near to the reference source. The time response for the exponential average was 125×10^{-3} seconds.

The theoretical transfer function, TF_{teor} (TF between the theoretical sine sweep at the receiver point and the sine sweep obtained near to the generator – at the reference source location) is expressed as in Eq. (3).

$$TF_{teor} = 10\log_{10}(\overline{x}^2_{sweep\_teor}) - 10\log_{10}(\overline{x}^2_{sweep\_near}) \quad (3)$$

where $\overline{x}^2_{sweep\_teor}$ is the exponential average of the theoretical sweep (obtained numerically) at the receiver point. The time response for the exponential average was also 125×10^{-3} seconds.

The exponential average ($\overline{x}^2$) used in Eq. (2) and (3) can be expressed as in Eq. (4).

$$\overline{x}_i^2 = \overline{x}_{i+1}^2 + (\overline{x}_i^2 + \overline{x}_{i-1}^2)/k \quad (4)$$

where $\overline{x}_i^2$ is the i[th] term of vector $\overline{x}^2$ and k can be expressed as in Eq. (5).

$$k = f_{aquis}T + 1 \quad (5)$$

where f_{aquis} is the acquisition frequency (Hz) and T is the integration time or response time (125×10^{-3} s). This fast integration time permits that the influence of past events in the signal be minimized.

The error between the theoretical TF and the estimated TF is given by Eq. (6).

$$Error_{TF} = TF_{teor} - TF_{est} \quad (6)$$

Now it is possible to quantify the noise source of interest at the receiver point multiplying the signal obtained close to the noise source by the obtained TF with de filter.

Synchronized time domain mean can be used before LMS filter to minimize the signal-to-noise ratio between the reference sound and the background noise. The main problem of this approach resides in that any problem of synchronism between the various acquisitions results in the degradation of the reference signal (Nunes, 2006).

To avoid this, Nunes used a time-frequency transform (Choi-Williams) to obtain the initial point in each signal to be used for synchronization purpose. Despite the good results, this procedure has an expensive computational cost, in addition to the fact that this is a manual technique, which limits its use in real time procedure. In this study a methodology based on the use of two walk-talks for trigger purpose is presented. With a walk-talk turned on near the reference source, a sound level trigger generated by this source will be received clearly by the acquiring sound signal system (located at the control point) using the second walk-talk turned on close of the microphone. After the trigger process the second walk-talk can be turned off. To avoid the additive effects of background noise on trigger signal, a post-trigger signal is generated for synchronization purpose. This post-trigger signal is compound by several harmonics sounds that (after acquiring, filtering and processing) will be used as input values of a fuzzy network which is used to point the beginning of synchronization. The steps are:

- First, the background noise is measured at the point where the first walk-talk will be placed (near the reference source). A spectral analysis is carried out to define the best harmonic composition to be used as pos-trigger signal;
- After the creation of the pos-trigger signal, the experimental apparatus is placed in a noiseless place and the post-trigger sound is measured by the receiver microphone with the walk-talks turned on. After the preprocessor signal step (of the acquired sounds signals) the resultant filtered levels will be used to adjust the fuzzy parameters;
- After the calibration of the post-trigger procedure, the reference sound source is placed near the noise source to be identified and the acquisition system is placed at the receiver point. In the field measurements the walk-talks are turned on during the period of pos-trigger signal. After words the walk-talk of the receiver point is turned off;
- Finally, the LMS filter is used and the resultant signal is passed through an exponential average algorithm and the Sound Transfer Function between the two points of interest can be estimated.

3. EXPERIMENTAL PROCEDURE

An experiment was developed to validate the adapted methodology. The Fig. 1 shows the experimental configuration – the position of the equipments and the instrumentation used. Three noise sources, a microphone and two walk-talks were used. The acquisition system is composed of an A/D board, BNC cable and a notebook.

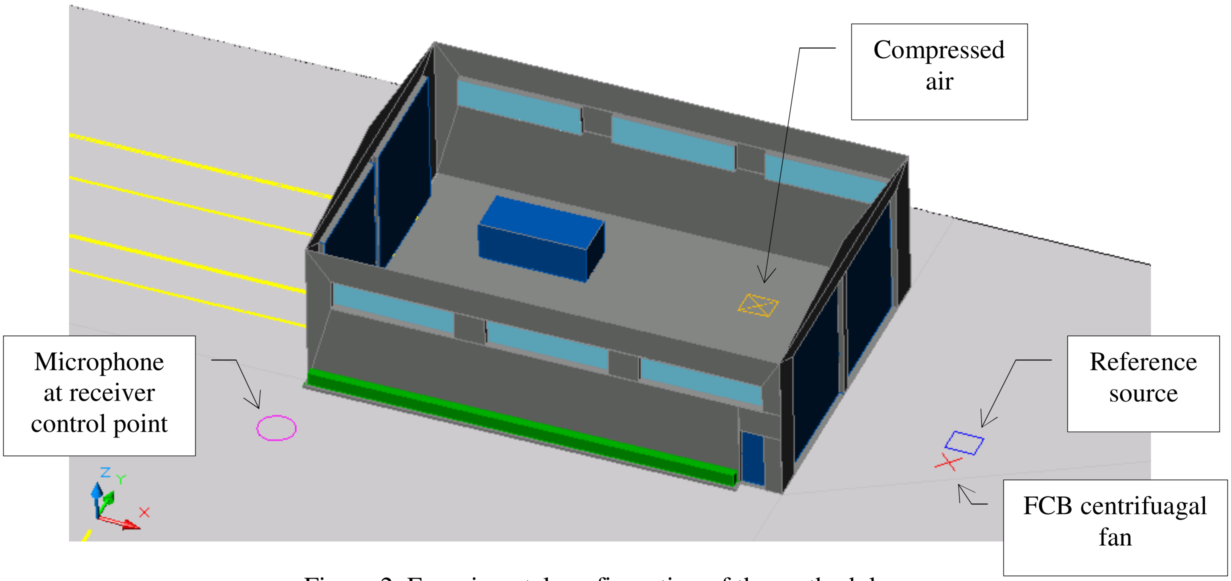

Figure 2. Experimental configuration of the methodology

The reference source was near to a centrifugal fan, which is the source whose noise contribution at control point is to be identified.

The sine sweep used in this study has the following characteristics: seventy (70) seconds duration, of which:

- 0 to 35 seconds, the frequency range increases linearly from 0 Hz to 4096 Hz;
- 35 to 70 seconds, the frequency range decreases linearly back to 0 Hz.

In this procedure it is possible to obtain all the signals emitted by each source separately, which is not possible in an industrial area. This is used to analyze the efficiency of the methodology.

First, the signal at one meter of the sine sweep without any other source was measured. This signal will be used as input of the LMS filter to estimate the TF. After which, all sources were switched on to acquire the signals. One walk-talk was placed in front of the reference source and the other was placed near the microphone.

Thirty samples were acquired at the receiver point, each of 85 (eighty five) seconds length. In the first ten seconds, the walk-talks were turned on. After which, they were turned off, leaving only the noise contribution of the involved sources. These signals were saved in the notebook for posterior processing and analysis.

In the processing stage, each measured signal was passed through the fuzzy logic function to determine the exact moment of the beginning of the sine sweep. The frequency range of the sine-sweep used for synchronization purpose is between 1500 Hz to 4096 Hz because the background noise measured interferes in lower frequencies. The LMS parameters were optimized using a genetic algorithm.

Several approaches were studied to obtain the sound TF between the two chosen points:

- With or without the LMS filter;
- The order of the application of the mean procedure and the LMS filter, that is:
  a) calculate the mean of the synchronized signal and apply the LMS filter on the result;
  b) apply the LMS filter on each synchronized signal and calculate the mean;
- The use of the theoretical sine sweep as target signal of the LMS filter;
- The use of the measured sound (at 1 meter of the reference source) due to the sine-sweep only as target signal of the LMS filter;
- Turn off the other sound sources and use the reference source to estimate the TF;
- The use of a ray tracing algorithm to calculate the theoretical values.

4. RESULTS AND DISCUSSION

Fig. 3 shows the spectrogram (Short Fourier Transform – SFT) of one measured sound at the receiver point. It can be observed that the first four seconds is the signal used to synchronize the acquired signals (walk-talks turned on). In the next eighty one seconds the walk-talks were turned off and resulting signals of the reference source and the other two sound sources with background noise can be observed.

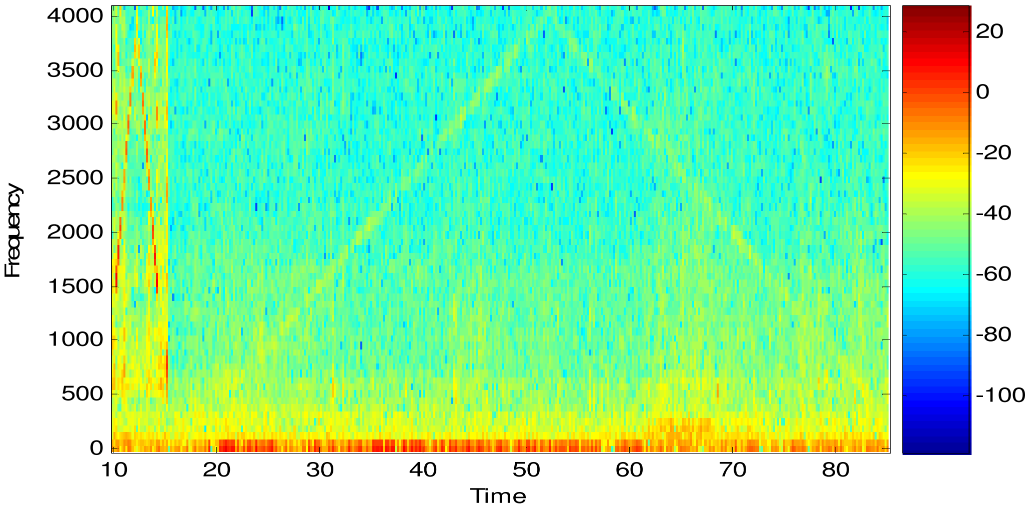

Figure 3. Example of a spectrogram obtained at receiver.

Using fuzzy logic, it was possible to synchronize the thirty samples. Fig. 4 shows a zoom of the pos-trigger region of the spectrograms. In Fig. 4a. the result of the triggered signal mean is shown and in Fig. 4b. the synchronized mean is shown. The parameter used in fuzzy logic was the filtered band-pass (1500 to 4096 Hz) squared root value (exponential mean with T_0=1.0 s.) of the measured signal and the correlation coefficient value between the measured pos-trigger signal. One of the signals is used as reference.

Analyzing Fig. 4b, a little raise in the amplitude of the region of signal compared to Fig. 4a is noted. It occurs because of the superposition of the signals. Also is observed a reduction of the background noise indicated by the increased percentage of yellow and green areas in Fig. 4b as compared to Fig. 4a.

However, the most important characteristic observed in Fig.4 is the perfect superposition of the signals, that is, the use of fuzzy logic obtained excellent results in the synchronization process.

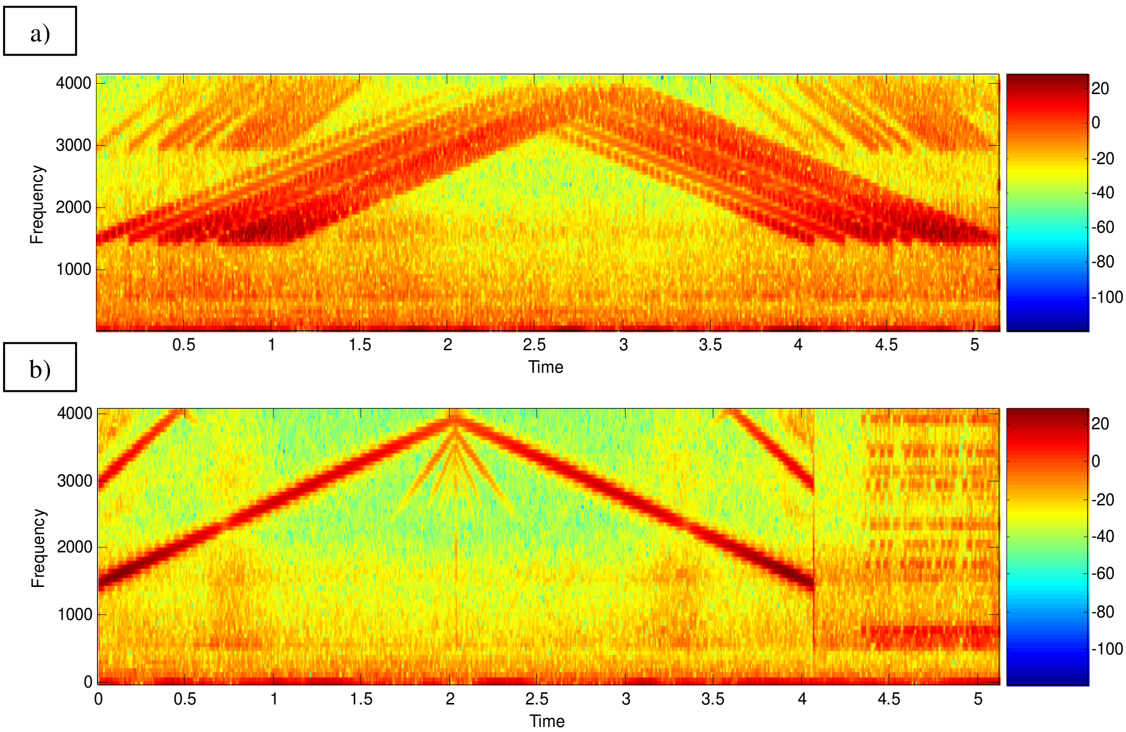

Figure 4. Spectrograms of the signals: (a) non-synchronized; (b) synchronized

After synchronization, the mean of the signals and filtering using the LMS adaptive filter is realized. Fig.5a shows the spectrograms obtained after the synchronized mean and Fig. 5b after filtering. As the signals are already synchronized, Fig. 5 shows only the reference source (sine sweep) used to obtain the TF.

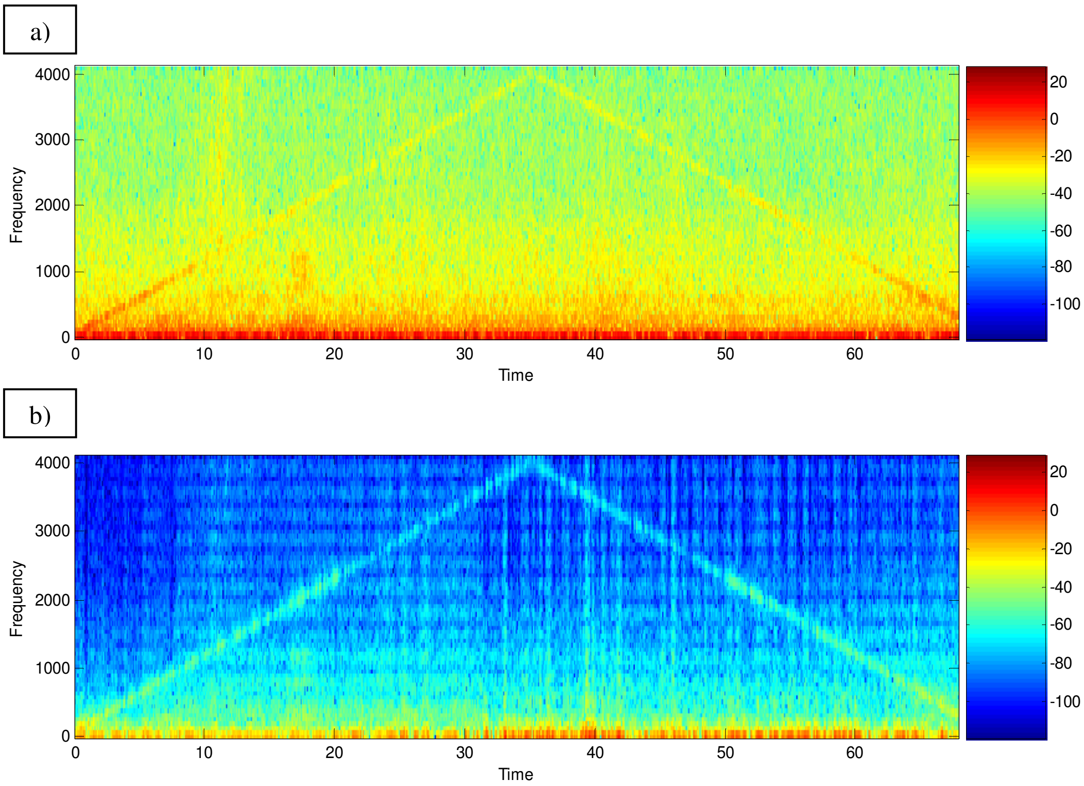

Figure 5. Spectrogram of: (a) the synchronizing signals mean; (b) synchronizing signals filtered.

Comparing Fig. 5a with Fig. 3, a great improvement in the signal-to-noise rate result of the use of Synchronized Time Domain Mean can be observed.

Analyzing Fig. 5b, the good performance of the LMS filter can be observed. This is evidenced by energy reduction of the background noise. Numerically, the signal-to-noise ratio of the signals shown in Fig. 5.b is nearly 42 dB.

The transfer function between the fan and receiver point was estimated using seven different experimental procedures, where the reference signal at the receiver point was obtained by:

TF1: The use of the LMS filter procedure on each synchronized signal followed by calculation of the mean of the 30 filtered signals. The target signal of the LMS filter was the measured signal at 1 meter of the reference source.

TF2: The Synchronized Time Domain Mean was applied to 30 measured sound signals followed by use of the LMS filter to improve the signal-to-noise ratio. The target signal of the LMS filter was the theoretical one.

TF3: TF2 however the target signal of the LMS filter was the measured signal at 1 meter of the reference source.

TF4: The Synchronized Time Domain Mean was applied to 30 measured sound signals;

TF5: The signal due to only the reference source turned on;

TF6: TF5 followed by use of the LMS filter to improve the signal-to-noise ratio. The target signal of the LMS filter was the theoretical one.

TF7: TF6 however the target signal of the LMS filter was the measured signal at 1 meter of the reference source.

Since the level of background noise was very high, none of the results of the experimental procedures can be used as reference. In order to establish a transfer function of reference a numerical procedure based on ray tracing model to calculate the theoretical transfer function TFref was used. Despite of being obtained by a numerical procedure, the TFref results have a precision of ±2.0 dB which was estimated using a large number of simulations and measurements made in the test area.

Analyzing Tab. 1, it can be observed that the improvement of the TF precision is directly related with the accuracy of the tools used to remove the effect of background noise. It is good to remember that this reference could not be the real transfer function of the analyzed source.

Table 1. Estimated Transfer Functions

Frequency Range [Hz]	TFref[dB]	TF1[dB]	TF2[dB]	TF3[dB]	TF4[dB]	TF5[dB]	TF6[dB]	TF7[dB]
63	-20	-24	-20	-22	-28	-16	-18	-22
125	-28	-27	-27	-26	-35	-25	-22	-25
250	-28	-30	-29	-27	-29	-22	-30	-26
500	-35	-37	-37	-37	-32	-25	-37	-36
1000	-40	-43	-42	-43	-39	-29	-42	-44
2000	-46	-49	-48	-49	-45	-32	-47	-48
4000	-47	-51	-53	-51	-47	-32	-52	-50
Overall	-50	-54	-55	-54	-50	-37	-54	-53

It can be observed that the function TF4, which corresponds to the synchronized mean of the signals without the LMS filter, doesn't presents good results in it's response in frequency bands, specially in the low ones.

The best result by frequency band is observed in TF2, which presents values rather close in relation of TFref. On the other hand, TF5 presents the worst results. It could be justified by the non-utilization of the synchronized mean and the LMS filter.

It could be observed that, all the functions presented equivalent levels higher than TFref, except TF5. This occurred because the numerical simulation, through which the reference TF, was obtained did not considered possible interferences that could occur in the measured signal at receiver point, that is, the measured experimental signals were contaminated with various other noise sources, resulting in a Sound Pressure Level higher than TFref.

5. CONCLUSIONS

The objective of this study is to determine the TF using adaptive filtering and synchronized mean.

Therefore, the main conclusions were:

- The proposed methodology can be used with success to estimate the source contribution to environmental noise even with a high background noise;
- The use of walk-talks and fuzzy logic were effective to synchronize all acquired signals;
- The best result by frequency band is observed in TF2 which is the Synchronized Time Domain Mean applied to all measured signals followed by use of the LMS filter (the target signal of the LMS filter was the theoretical one).

6. ACKNOWLEDGEMENTS

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7. REFERENCES

Coelho, K.V., 1999. "Saúde Ocupacional na Mineração- MTE", Monograph of specialization course in Security Engineering, Federal University of Brasilia.

Handley, J.M., 1995, "Controle de Ruído Industrial", Revista Acústica e Vibrações. 15th Ed.

ISO 9613-2:1996. "Acoustics – Attenuation of Sound During Propagation Outdoors. Part 2: General Method of Calculation". 1st Ed.

Nunes, M.A.A., 2006. "Uma Metodologia para Quantificação da Contribuição Sonora de Fontes de Ruído Industrial no Meio Ambiente". Dissertação de Mestrado, Federal University of Uberlândia, Brazil, 153 p.

Nunes, M.A.A., Duarte, M.A.V., Teodoro, E.B., 2007, "Application of LMS Filter for Noise Source Quantification in Environment Acoustics – A Computational Contribution", Proceedings of the XII International Symposium on Dynamic Problems of Mechanics, Ilhabela, Brazil.

Pimentel-Souza, F., 2000. "Efeito do Ruído no Homem Dormindo e Acordado". Proceedings of XIX ESBA, SOBRAC-2000, Belo Horizonte, Brazil.

Widrow, B., Stearns, S. D., 1985, "Adaptive Signal Processing", Prentice Hall, Englewood Cliffs, New Jersey, 1st ed, 528p.

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