Feedback Active Noise Control based on a Digital Signal Processor for Reducing Noise from a Blower Machine

Guillermo R. Friedrich
Dept. Ingeniería Electrónica
Universidad Tecnológica Nacional, Fac. Reg. B. Blanca
Bahía Blanca, Argentina
gfried@frbb.utn.edu.ar

Guillermo H. Reggiani
Dept. Ingeniería Electrónica
Universidad Tecnológica Nacional, Fac. Reg. B. Blanca
Bahía Blanca, Argentina
ghreggiani@frbb.utn.edu.ar

Pablo Girón
Dept. Ingeniería Mecánica
Universidad Tecnológica Nacional, Fac. Reg. B. Blanca
Bahía Blanca, Argentina
pgiron@frbb.utn.edu.ar

Adrián Azzurro
Dept. Ingeniería Mecánica
Universidad Tecnológica Nacional, Fac. Reg. B. Blanca
Bahía Blanca, Argentina
azzurro@frbb.utn.edu.ar

Abstract— **Rotary blowers often produce high levels of sound at discrete frequencies, related to their rotational speed. High levels of sound can cause more or less severe damage to the ear, depending on its level and the time of exposure. In some cases it is mandatory to use hearing protection and reduce the time of exposure to noise. In other cases the noise affects the comfort of the workers and their productivity. Therefore, there must be applied elements and techniques to reduce noise, which complement the protection elements. If the noise frequencies are at the low end of the spectrum they may be difficult to attenuate by mean of passive techniques. In the present work it is presented the development of an active noise control system to cancel or reduce the noise emitted by a high power blower in a petrochemical plant. The solution is implemented on a digital signal processor and based on feedback scheme.**

Keywords—active noise control, blower noise, feedback ANC.

I. Introduction

Noise in industrial workplaces is one of the most common health risk factors. In most cases is technically feasible to control its excessive level by using protective elements, but also applying technology to the noisy machines. High levels of sound can cause more or less severe damage to the ear, depending on its level and the time of exposure. In some cases it is mandatory to use hearing protection and reduce the time of exposure to noise. In other cases the noise affects the comfort of the workers and their productivity. Therefore, different techniques must be applied to reduce noise, which complement the hearing protection equipment.

In the industry, rotary blowers are a type of machines that typically produce high levels of sound at discrete frequencies, related to their rotational speed. Fig 1 shows a typical blower mounted in a cabinet.

Fig. 1. A typical blower mounted inside a cabinet

If the noise frequencies are at the low end of the spectrum they may be difficult to attenuate by mean of passive techniques.

The blower that motivates the present work is one of three lobes and 110 kW, which produces in front of the cabinet door a sound intensity of 105 dB(A). This noise has a main tonal component of 240 Hz and several harmonics of significant amplitudes, up to 960 and also 1200 Hz. Fig. 2 shows a FFT spectrum diagram obtained by mean of a Presonus PRM1 measurement microphone [1], an AudioBox USB device [2] and StudioOne software [3].

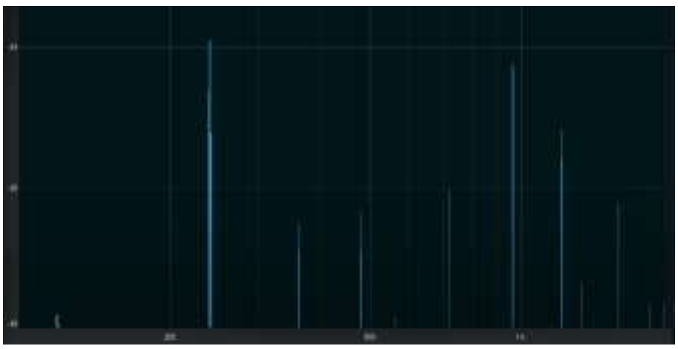

Fig. 2. Spectrum diagram of the noise produced by the blower, measured in front of the cabinet door (Obs.: the left scale is in dB but is not related to any acoustic magnitude).

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In front of the blower cabinet, about 30 m away, there is an office building through whose windows the noise coming from that equipment enters. Because the components of higher frequencies, 960 and 1200 Hz, could be attenuated by passive techniques and also more attenuated by distance, it was proposed to apply active noise control (ANC) techniques to reduce the 240 Hz component.

Although the theoretical foundations of ANC have become mature over the last decades ago, there is currently an important production in terms of its application to different environments and problems [4]. It is worth mentioning, among others, the applications of ANC to reduce the noise of power transformers in urban environments [5]; noise attenuation in natural ventilation windows [6]; the attenuation of sound propagation in an air-handling duct [7]; active control of noise in vehicle cabins [8]; active/passive silencers [9]; active noise systems for reducing outdoor noise [10].

The developed solution was based on an Analog Devices ADSP-21061 EZ-KIT Lite bundle [11], based on ADSP-21061 processor [12] running at 40 MHz, with an AD1847 16-bit stereo codec [13], configured at 8 kHz sample rate. The code was written in C.

Preliminary laboratory tests were carried out on a typical duct structure [14], in order to adjust details and parameters of the algorithms. Then, other tests were carried out on an open space structure, emulating the operating conditions of the blower together with the active solution. Fig. 3 shows one speaker used to reproduce the recorded blower noise and the other speaker used to emit the anti-noise generated by the ANC system. Also, the same figure shows the error microphone capsule and the microphone used to do measurements, located in front of the speakers.

Fig. 3. Speakers and microphones as was used in experimental tests.

The ANC scheme adopted was the feedback one because it is more simpler, since it uses a single microphone, and also avoids the acoustic feedback that appears in the feed-forward scheme. However, the results shown in [7], based on an almost similar test setup but applying feed-forward ANC, were used to compare.

II. Active Noise Control System

A. *Types of Active Noise Control Systems*

Active noise control systems are based on one of two methods: feed-forward and feedback. Feed-forward control is where a coherent reference noise input is sensed before it propagates to the canceling speaker. Feedback control is where the controller tries to cancel the noise without having an input noise reference (Fig. 6). Feed-forward ANC is, probably, the main technique used in today systems. They are further classified into two categories: adaptive broadband feed-forward control, using an acoustic input sensor (Fig. 4), and adaptive narrowband feed-forward control, which uses a non-acoustic input sensor (Fig. 5).

Figs. 4, 5 and 6 shows the three ANC schemes over a duct, which is the typical structure used to test ANC systems and techniques.

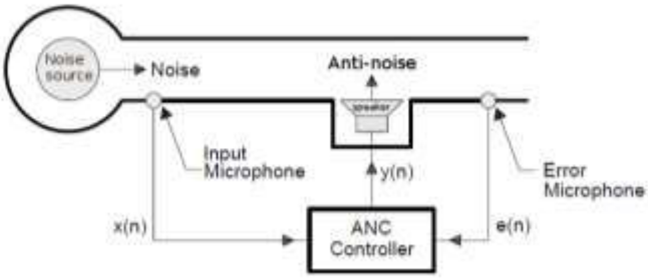

Fig. 4. Broadband feed-forward ANC system.

The broadband feed-forward ANC system has the disadvantage of having acoustic feedback, which needs to be cancelled. The anti-noise radiated by the loudspeaker not only cancels acoustic noise downstream, but also radiates upstream to the input microphone, contamining the reference input x(n). This acoustic feedback is a potential source of instability in this type of ANC system.

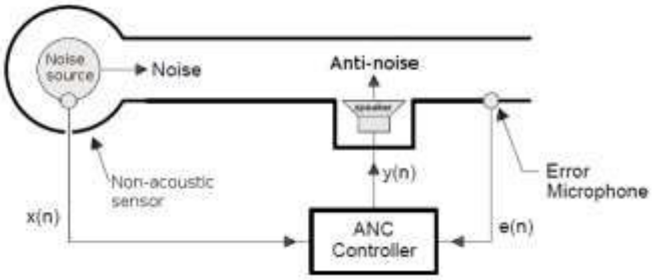

Fig. 5. Narrowband feed-forward ANC system.

When the primary noise is periodic (or nearly periodic), as produced by rotating machines, the input microphone can be replaced by a nonacoustic sensor (e.g.: a tachometer, an accelerometer, or some optical sensor), which is not affected by the acoustic feedback phenomenon.

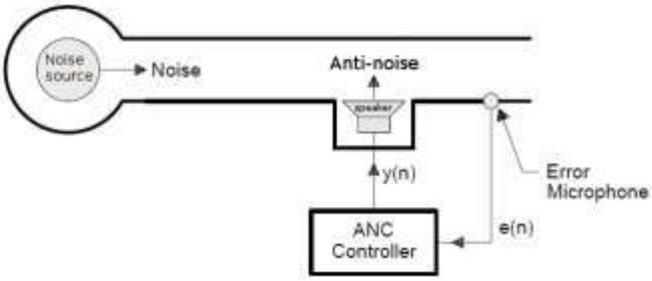

Fig. 6. Feedback ANC system.

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In feedback ANC scheme, a microphone is used as an error sensor to detect the undesired noise. From the error sensor signal is extracted an estimation of the original noise signal, which is then filtered and amplified to produce the magnitude and phase needed to cancel the noise.

This configuration provides less cancelation than feed-forward, and could suffer from instability due to the feedback loop. However, based on this scheme is possible to get a robust ANC system to cancel narrowband noise.

In the case of the machine under study, the feedback scheme has the advantage of being non-invasive, since it is not necessary to mount a sensor on the machine to obtain the noise signal.

In case of obtaining favorable results it will be possible to justify the need to work on the machine to improve the system, converting it into narrowband feed-forward. It must be kept in mind that this type of machine is part of a continuous process of very high cost, whose operation and maintenance is controlled by very rigorous and standardized procedures. Any experimental modification that one wishes to make must be very well justified.

B. *Feedback ANC System*

A feedback ANC system, as shown in Fig. 7, is based on an adaptive predictor. Because this system requires only one error microphone, it avoids the acoustic feedback problem inherent in the feed-forward ones.

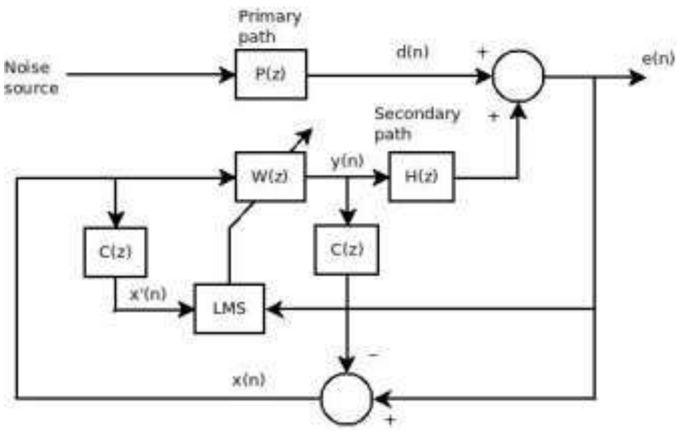

Fig. 7. Block diagram of a feedback ANC system.

The basic idea of this algorithm is to estimate the primary noise *d(n)* and to use this value as the reference input for the adaptive filter. *H(z)* represents the secondary path transfer function, while *C(z)* is an estimate of *H(z)* obtained by means of an offline FxLMS identification process. The primary noise is estimated as:

$$x(n)=e(n)-\sum_{i=0}^{M-1}(e(i)\quad c_i y(n\quad i)) \tag{1}$$

where c_i is the i-th coefficient of *C(z)*, the secondary-path estimation filter, and M is the filter order. From Fig. 7:

$$D(z) = E(z) - H(z)\,Y(z) \tag{2}$$

If C(z) is a model of H(z), then:

$$D(z) \cong X(z) = E(z) - C(z)\,Y(z) \tag{3}$$

And the error signal can be calculated as:

$$E(z) = D(z) + W(z)\,H(z)\,X(z)) \tag{4}$$

The error signal is 0 when:

$$-\,W(z)\,H(z)\,X(z) = D(z) \tag{5}$$

i.e., when *W(z) H(z)* is equal to a delay equivalent to a multiple of the signal period, and the minus sign indicates a 180° phase shift that is needed to cancel *D(z)*.

The coefficients of *W(z)* are adjusted online by mean of an adaptive algorithm, such as LMS, based on the value of error signal, *e(n)*.

The anti-noise value is computed from the value of *x(n)* obtained in (1):

$$y(n)=\sum_{i=0}^{N-1} w_i x(n-i) \tag{6}$$

where *N* is the filter order.

Also there is computed *x'(n)*, the filtered-X version of *x(n)*:

$$x'(n)=\sum_{i=0}^{M-1} c_i x(n-i) \tag{7}$$

Next, *x'(n)* is used by the LMS algorithm to update the w_i coefficients for the next iteration:

$$w_i(n+1)=w_i(n)-\beta e(n)x'(n-i), \qquad i=0,1,\ldots,N-1 \tag{8}$$

Where β is the step size of the LMS algorithm, from which depends the convergence speed and the stability of the system.

There is another version of FXLMS algorithm, called Leaky FXLMS, which introduces a leaky coefficient that avoids the numerical overflow which could appear in fixed point architectures, due to its limited precision. Although this project is based on a floating point 32 bit DSP, Leaky FXLMS improves the system stability at the expense of a little reduction in the noise cancellation. The update of coefficients using Leaky FXLMS is computed as follows:

$$w_i(n+1)=\lambda w_i(n)-\beta e(n)x'(n-i) \tag{9}$$

Where λ is the leaky coefficient, whose value is slightly less than 1.

Next, it is presented the section of C code that implements the adaptive adjustment of the filter coefficients W (z) through FXLMS:

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```
eb = error * beta;

for(i=N-1; i>=0; --i)
{
   w[i] = leaky * w[i] - eb * xw2[i];

   if(i)
      xw2[i] = xw2[i-1];
}
```

Where: *error* contains the value of *e(n)*, *beta* the value of β, *leaky* the value of λ; *xw2[]* is an array that contains the values of $x'(n-i)$.

C. Secondary-path Adaptive Identification.

As occurres in practical applications, *H(z)* is unknown and must be estimated by the filter *C(z)*. This estimation can be done by mean of an offline or online adaptive technique.

Assuming the characteristics of *H(z)* are unknown but time-invariant, an off-line modeling technique can be used to estimate *H(z)* during an identification stage. After this identification, the estimated model *C(z)* is fixed and used for active noise control. Fig. 8 shows the block diagram of the experimental setup for the direct off-line secondary-path identification. Uncorrelated white noise is internally generated by the DSP, and simultaneously sent to the secondary-path via the loudspeaker, and to the filter *C(z)*. The difference between *e(n)* and *r(n)*, called *e´(n)*, which should tend to zero during the identification stage, is used to feed the LMS algorithm, which adjust the *C(z)* filter coefficients.

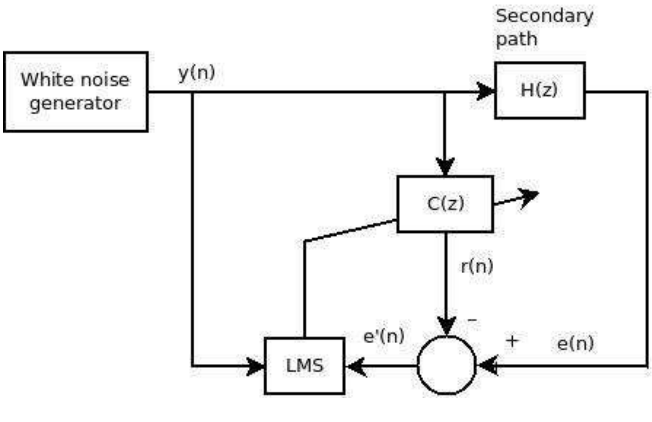

Fig. 8. Secondary-path identification procedure diagram block.

The response of the adaptive model is computed by:

$$r(n)=\sum_{i=0}^{M-1} c_i y(n-i) \tag{10}$$

Then, using the value $e'(n) = e(n) - r(n)$, the values of c_i are adaptively updated by means of the LMS algorithm as follows:

$$c_i(n+1)=c_i(n)+\mu e'(n) y(n), \qquad i=0,1,\ldots,M-1 \tag{11}$$

Where μ is the step size, whose value must fit between 0 and $(M P_y)^{-1}$, being P_y the power of the white noise.

The strategy used to find the best adjustment of the c_i coefficients as a function of the values of *M* and μ consisted of sending the samples of the residual error $e'(n)$ to the other output channel of the codec, and reading the voltage value with an AC voltmeter. The goal is to obtain the lowest reading on the instrument. In this way, it was determined that the values of M=384 and μ=5.10^{-3} were the ones that provided the best fit, both for the tests in the duct and in the open space.

Then, with the appropriate values of *M* and μ, it could be determined that for practical purposes it is only necessary to run the identification process until the RMS value of *e(n)* reaches 20% of the initial value (for practical purposes and according to the observations made during the tests, it would be sufficient to maintain the identification process for about ten seconds).

D. Experimental setup

In Fig. 9 is shown the block diagram of the set implemented to perform the tests.

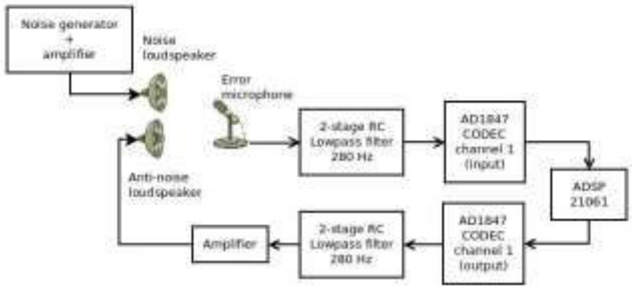

Fig. 9. Test setup block diagram

The input RC lowpass filter has been put to attenuate the higher order frequencies that are out of interest for the ANC system. The output RC lowpass filter has the objective of eliminate high frequency noise (ripple) caused by the adaptive signal processing, that corrupts the anti-noise signal. Both filters improve the stability and performance of ANC.

III. Tests and Results

In earlier stages, several tests were done over a duct installed in the laboratory, to check and tune the code, algorithms, hardware and another details. Then, it was the test setup (Fig. 3) mounted in multipurpose room without acoustic conditioning, to do the open space tests.

There were used two noise signals for the tests: a pure 240 Hz tone and a recorded audio from the blower, whose spectral diagram is shown in Fig. 2.

The noise measurements were realized using two devices / methods: (i) Studio One software feeded by a Presonus PRM1 measurement microphone through an AudioBox USB; (ii) Cesva sonometer. The attenuation of the 240 Hz tone was measured with the first method, while the global sound level attenuation was measured with the sonometer.

Table I shows the results obtained for a 240 Hz pure tone. It can be seen that when the error microphone was closer to the loudspeakers, and therefore gets more sound power, the attenuations are better. This is due to the signal to noise ratio is 1,5 dB better, and enables a more accurate adaptive noise control.

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Also, the global sound level gets a greater reduction than for 240 Hz single tone, because of the dB(A) compensation curve used by the sonometer (23,8 dB vs 19,4 dB).

Table II shows the results obtained using the recorded blower noise, whose spectral diagram is shown in Fig. 2. It can be seen that the 240 Hz tone attenuation is about 3 dB lower than for the case of a single tone noise (Table I). And, moreover, the global noise has almost no reduction, because the rest of the harmonics have a great value influence in the weighted sum which gives value expressed in dB(A).

TABLE I. ATTENUATION FOR 240 HZ SINGLE TONE SIGNAL

Attenuation for 240 Hz single tone signal **LMS: leaky coefficient = 0.99995 $\beta = 10^{-3}$**		
Distance from loudspeakers to error microphone	***40 cm***	***30 cm***
Original noise power (measured at error microphone with Cesva sound level meter using global scale)	82,6 dB(A)	84,3 dB(A)
Attenuation measured with mic. Presonus + AudioBox + Studio One software	15,2 dB	19,4 dB
Attenuation measured with Cesva sound level meter (dB(A))	18,1 dB	23,8 dB

TABLE II. ATTENUATION OF THE 240 HZ TONE OF BLOWER NOISE

Attenuation of 240 Hz tone component of blower noise **LMS: leaky coefficient = 0.99995 $\beta = 10^{-3}$**	
Distance from loudspeakers to error microphone	***30 cm***
Original noise power (measured at error microphone with Cesva sound level meter using global scale)	85,7 dB(A)
Attenuation measured with mic. Presonus + AudioBox + Studio One software (240 Hz)	16,2 dB
Attenuation measured with Cesva sound level meter in global scale (dB(A))	0,2 dB
Attenuation measured with Cesva sound level meter in spectral mode (250 Hz division)	16 dB

In order to compare these results with another system, Table III shows the results obtained with a leaky feed-forward FXLMS scheme, as presented in [7].

From the comparison between the values of Table I and Table II, it is possible to conclude that the presence of a combination of tones reduce the performance of ANC system: 19,4 dB vs 16,2 dB. This can also be seen in Table III for a feed-forward scheme: 28,3 dB for a single 200 Hz tone vs 19,8 dB for multiple tones. This is because, considering only the 240 Hz component, there is a worse SNR in the second case.

TABLE III. ATTENUATION USING LEAKY FEED-FORWARD FXLMS

Attenuation reported by Biaggini et al [7] using leaky feed-forward FXLMS ANC to reduce outdoor noise		
Signal characteristics	***Single tone 200 Hz***	***Multiple tones 200-300 Hz (every 20 Hz)***
Original sound power level	75 dB	70 dB
Attenuation	28,3 dB	19,8 dB

The order of attenuation achieved would improve the health conditions in the area near the blower, making it possible to reduce hearing protection, or increase the exposure time. Anyway, these modifications must be analyzed and authorized by health specialists.

IV. CONCLUSIONS

The noise produced by industrial equipment is a problem that affects health and comfort of workers and neighbors. For this reason it is interesting to find innovative solutions, as is the case of using ANC to reduce lower frequency noises.

In the present work it was presented the development and tests of a feedback ANC scheme, designed to be applied to reduce the noise emitted by a high power blower in a petrochemical plant.

The system was developed on an Analog Devices ADSP-21061 digital signal processor running at 40 MHz, using an AD-1847 codec who has two 16 bit channels, configured to 8 kHz sample rate.

This system was tested on laboratory, trying to emulate the real conditions of a typical blower.

The tests gave good results, achieving more than 19 dB of attenuation for a 240 Hz tonal noise (more than 23 dB when measured over a dB(A) scale). The comparison with solutions based on a feed-forward scheme enable to think that it is possible to try to enhance the current system, tuning certain parameters of it, such as: filters order, β, μ, leaky coefficient and sample rate.

In addition, from the same comparison it can be seeing of interest to evaluate a feed-forward type solution due to its higher performance, although it has some more complexity.

However, the possibility of implementing a feed-forward solution in a next stage, using either a microphone or a non-acoustic sensor attached to the machine will be also considered.

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