REVISTA FCEFYN

Development of a narrowband multichannel active noise control system for enclosures

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Resumen— Este trabajo presenta el desarrollo de un sistema multicanal de control activo de ruido para el interior de recintos. El control se aplica al ruido de baja frecuencia producido por el motor de un automóvil, en donde las técnicas de control pasivo resultan ineficientes. El trabajo se enfoca en la optimización del sistema para obtener resultados óptimos al implementarlo en un procesador digital de señal comercial.

Palabras clave—control activo de ruido, procesamiento digital de señales.

Abstract—This paper presents the development of a multichannel active noise control system for enclosures. It is applied on reducing low frequency car engine noise, where passive noise control techniques are inefficient. This work focuses on system optimization for optimal results when implementing on a commercial digital signal processor.

Keywords —active noise control, digital signal processing.

INTRODUCTION

Currently, in the area of automotive noise, vibration and harshness (NVH), studies focuses on modifying noise and vibration characteristics of vehicles, particularly in cars and trucks. While noise and vibration can be objectively measured, the "hardness" is a subjective quality that is measured through juries evaluations, or by the use of analytical tools that provide results correlated with the human subjective impressions studied in psychoacoustics (Ahrens et al, 2013). NVH is basically divided into two areas: indoor NVH, which deals with noise and vibration experienced by individuals inside the cabin, and outside NVH which focuses on how noise is radiated from the vehicle to the environment (Beranek and Ver, 2006). The firsts active noise control systems in cars were frontloaded tonal control engine noise arrangements (Elliot et al, 1988), developed as part of a research program which also focused on active noise control inside propeller aircraft (Nelson and Elliot, 2001). In recent years, low frequency sound in a number of planes and cars is attenuated using commercial active noise control systems (Elliot, 2009).

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Acoustic noises can be classified in narrowband and broadband. The former presents discrete frequency components where most of the energy is concentrated. These components can be harmonics of a fundamental frequency, as in car engine noise. Broadband noise spectrum includes frequency bands with uniformly distributed energy (Hansen and Snyder, 1997).

Regarding cancelling strategy it is possible to apply passive noise control (PNC) or active noise control (ANC). PNC operates with insulating and absorbing materials aiming to reduce unwanted noise amplitude, nevertheless it exhibits low frequency limitations due to wavelength size. ANC arises as a complement of PNC, generating an antinoise signal similar to noise to be canceled, but with opposite phase. Both signal act in the desired quiet zone cancelling each other by destructive interference. The difference between both signals is called the system error signal. It is used in the learning process of an adaptive system. ANC efficiency has an upper frequency limit of 500 Hz (Hansen, 2001). At higher frequencies the quiet zone size is very much reduced.

From the design architecture point of view, ANC systems may be classified as feed forward or feedback. In the first case a noise correlated reference is taken before it propagates to the quiet zone. On the other hand a feedback system cancels noise without this reference (Narahari, 2003), performing estimations through the error signal itself. It assumes the incoming noise cycle is similar to previous ones, cancelling only periodic components and those located around a central frequency. In this work a feedback system was chosen for cancelling only periodic car engine noise. Decision was based on the fact that placing a reference sensor outside the vehicle could lead to cancel any kind of sound from outside, including warning signals, which represents a threat in a moving vehicle.

According to the desired quiet zone size, single cannel or multichannel ANC systems are implemented. When aiming to cancel noise in small places like ear cavity, a single sensor and secondary source may achieve the objective. In enclosures multichannel systems become necessary, mainly because of sound field’s complexity.

In multichannel ANC systems we aim to calculate several transfer function´s output and its adaptation process in a single sample period. For that reason we use a commercially available Digital Signal Processor (DSP) which has to be optimized to achieve this challenge. This means transferring acoustic signals to electric environment through microphones (with relevant signal conditioning), and then converting them to the digital domain with analog to digital converters (ADC). Once the cancelling signals have been generated, they are transferred to the analog domain by digital to analog converters (DAC). Then DAC outputs are amplified and turned into acoustic signals through speakers. These transfer functions are grouped in a single function called secondary path for each source – sensor pair of the multichannel system.

The present work was carried out as an extension of the final degree project developed by (González Vergara, 2011) based on active noise control in headphones.

In acoustic analysis section the basic model is developed, describing sources and sensors locations, secondary paths details and the first optimization of the system. Then, the selected DSP is described, and algorithm implementations and further system optimizations are analyzed. Finally results obtained are presented along with main conclusions.

ACOUSTIC ANALYSIS

When sound field is excited inside an enclosure stationary waves are generated by means of constructive and destructive interference, called normal modes. By increasing the secondary source number, more normal modes can be controlled. However, the number of modes which significantly contribute to enclosure’s sound field grows with frequency in an approximate proportion of its third power. Then many secondary sources may be necessary to achieve global cancelling inside the complete room (Elliot and Nelson, 1993) (Haykin, 1996). On the contrary, local ANC systems allow noise cancelling in smaller areas with reduced amount of sources and error sensors. In car interiors quiet zones are generally chosen to be in the vicinity the driver’s and passenger’s heads, whereby local ANC was chosen for the application.

Enclosure design and frequency response

Enclosure design relied on two main aspects. On one side four different sedan cars dimensions were studied, on the other hand Bolt criteria (Bolt, 1946) was applied for achieving uniform normal modes distribution. The cabin was built with medium density fiber and phenolic, placing acoustic absorbing material inside aiming to reduce reverberation times. This allowed operating with high direct to reverberant signal ratio secondary paths, which improved system performance by means of energy coupling between secondary sources and sensors (Kuo and Morgan, 1996).

The frequency response of the directive sources was 40 Hz to 2 kHz ± 3 dB. Error sensors selected for the application were electret measurement microphones (Behringer ECM 8000) with omnidirectional directivity pattern and frequency response from 20Hz to 20 kHz.

Sources number and location

For determining secondary sources number it was considered that the more sources there are, the higher attenuation the system can achieve (Budde et al, 2013). According to this, and to the fact that sedan cars usually present four speakers, the ANC system was implemented with four secondary sources.

Kuo and Morgan (Kuo and Morgan, 1996) state that car speakers location are usually acceptable in noise control. Besides they suggest that most ANC systems rely too much in signal processing, but without any physical array optimization the system could fail to achieve good results. Taking these guidelines normal modes were studied inside the enclosure and speakers were located in sedans usual position.

Acoustic model simulation

The ANC transfer function was made with adaptive transversal filters. Simulations were carried out for determining filter length and to estimate required computational load. A four stroke and four cylinder engine operating at 2500 rpm noise signal was synthesized to simulate the car´s engine sound to be cancelled.

In simulations it becomes necessary to consider not only the sound field inside the enclosure and transducer's response, but also the signal processing to be performed. Once the cabin was built and transducers were located, the impulse response was measured for each source – sensor pair. Impulse responses of primary noise source and error sensors were also measured. Convolving noise signal with the later responses allow us to obtain noise sensed by each microphone.

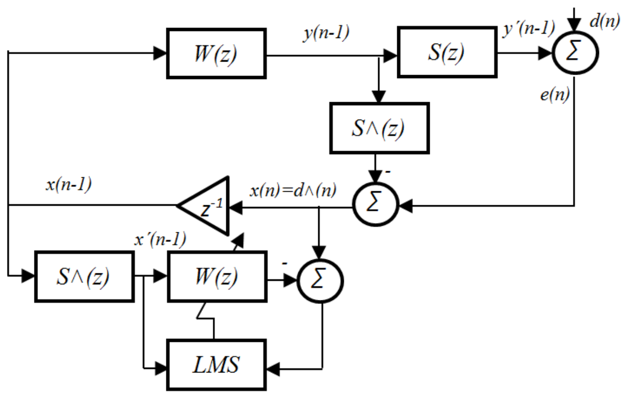

Fig. 1. Single channel MFxLMS algorithm diagram.

Then convolving control generated signals with secondary paths responses results in the acoustic antinoise signals (Budde et al, 2013). By adding both, residual noise and attenuations were computed. The presence of secondary paths prevents the direct use of fast and robust adaptive algorithm as Least Means Squares (LMS) (Kuo and Morgan, 1996). This can be compensated with an improvement of LMS called the Filtered-x Least Mean Square (FxLMS) algorithm (Widrow and Stearns, 1985). In this work a further modification, the modified FxLMS algorithm (MfxLMS) was selected (Fig. 1), mainly because it exhibit higher stability than traditional FxLMS without any performance degradation (González et al, 2012).

In Fig. 1 S(z) represents secondary path and S^ (z) the corresponding estimation, while W(z) is the control filter. Signal d(n) is the noise to be canceled, and e(n) is the residual noise. Simulations were carried out with double precision floating point arithmetic, which causes finite precision errors to be negligible. Fig 1 represents just one source –sensor pair. In a system with four sources and two error sensors, control filters are updated by means of both error signals, which bring a considerable raise in the computational load required. In Equation 1 updating of $W_1(z)$, one of the control filters, is carried out in (n+1)th sample, where μ is the adaptation step, $x_1(n)$ one of the estimated references, and $e_i(n)$ the error signals. $S_1(z)$ and $S_5(z)$ represents secondary path associated to number 1 source and both error sensors :

$$W_1(n+1) = W_1(n) + \mu\{[S_1(z) * x_1(n)]e_1(n) + [S_5(z) * x_1(n)]e_2(n)\} \quad (1)$$

Operations begin by learning secondary paths with white noise input signals. Process were carried out according to plant identification schemes, since secondary paths transfer functions are unknown by the controller. Learning stage finished when mean squared errors reached stable values. In simulations only the first 100ms from each secondary path were taken into account. Adaptations were carried out with finite impulse response (FIR) filters of 600 taps, which means that only the initial 75ms were considered. Even though path estimations are not perfect because of the portion being ignored, the selected algorithm presents high error tolerance (Kuo and Morgan, 1996). Once analog and acoustic characteristics were defined, control filter's learning processes were enabled through MFxLMS algorithms.

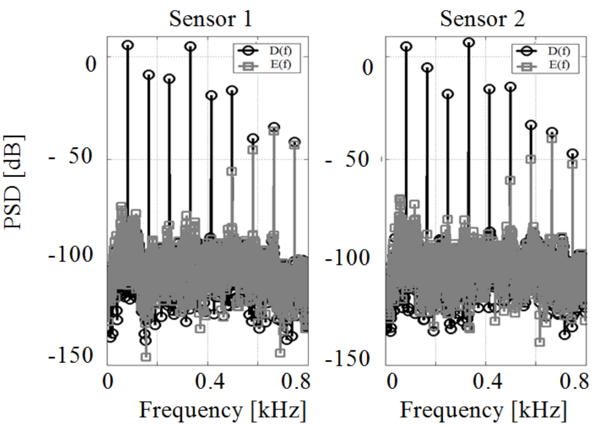

Fig. 2. PSD of error signals in simulations for 4x2 ANC system off and on (D(f) and E(f)).

TABLE 1. Average attenuations obtained in error sensors in 4x2 ANC system simulations.

Harmonic number	Frequency [Hz]	Attenuation [dB]
1	83	97
2	166	85
3	249	67
4	332	96
5	415	72
6	498	42
7	581	11

ANC system completely cancelled the five first harmonic components of the synthesized noise. At higher frequencies attenuations decreased. In Fig. 2 Power Spectral Density (PSD) of error signals are shown for system's stable operation (grey), and without ANC (black). Attenuation values are described in Table 1. Optimum results were obtained with 200 coefficients for each control filter. We observed no significant improvement when raising such number. However, it must be taken into account that double precision was used. Working with a single precision DSP may require a considerable increment in filters length to achieve similar results. After defining W(z) and S^ (z) filters length, multiplications number per sample period for real time processing were computed (Table 2).

TABLE 2. Multiplications per sample period in 4x2 ANC system, for $L_{s\wedge} = 600$ and $L_w = 200$.

Operation	Filter number	Coefficients	Multipli-cations
W(z) filtering	8	200	1600
W(z) copy filtering	16	200	3200
S^(z) filtering	8	600	4800
S^(z) copy filtering	16	600	9600
W(z) update	8	200	3200
Multiplications per sample period			22400

Right from the first simulations it was clear that working with four secondary sources would require a considerable computational power. Not only the filter outputs and their

Fig. 3. Secondary sources and error sensors inside the enclosure for 2x2 ANC system.

adaptation, but also all the cross-channel effects needed to be computed in a single sample period, To lower operations required per sample period, we reduced the ANC architecture to two error sensors and two secondary sources (Fig. 3). By means of reducing the distance between sources and error sensors, improvements in direct/reverberant signal ratios were achieved. This allowed decreasing S^(z) filters from 600 to 500 coefficients, and W(z) filters from 200 to 150 coefficients. Results are shown in Fig. 4 and Table 3. Multiplications numbers per sample period are computed in Table 4.

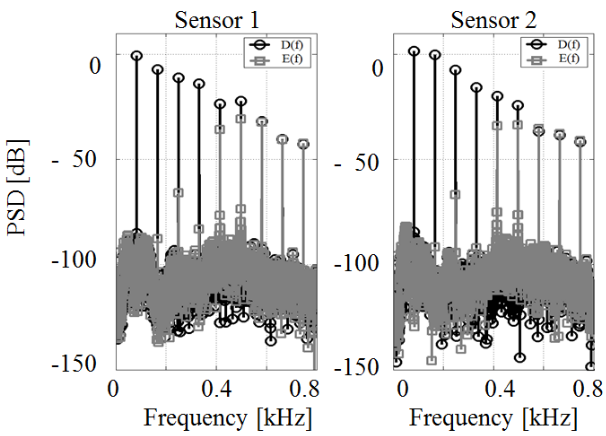

Fig. 4. PSD of error signals in simulations for 2x2 ANC system off and on (D(f) and E(f)).

TABLE 3. Average attenuations obtained in error sensors in 2x2 ANC system simulations.

Harmonic number	Frequency [Hz]	Attenuation [dB]
1	83	92
2	166	70
3	249	65
4	332	52
5	415	19
6	498	12
7	581	1

Conclusions of the Acoustic Study

By lowering secondary sources number from 4 to 2 no significant results impairment were observed. Improved sources location allowed filter's length reduction, which together with filter number decrement lead to a considerable computational load reduction in a proportion of 3.

TABLE 4. Multiplications per sample period in 2x2 ANC system, for $L_{S^\wedge}$ = 500 and L_w = 150.

Operation	Filter number	Coefficients	Multipli-cations
W(z) filtering	4	150	600
W(z) copy filtering	8	150	1200
S^(z) filtering	4	500	200
S^(z) copy filtering	8	500	4000
W(z) update	4	150	600
Multiplications per sample period			8400

DIGITAL SIGNAL PROCESSOR IMPLEMENTATION

Once simulations were completed, DSP was selected and the control system was implemented.

Processor Selection and Description

The digital signal processor selected for the application was StarCore MSC7116 from Freescale Semiconductor Inc., with the MSC7116EVM evaluation module. Module integrates AK4554 CODEC from AKM Semiconductor Inc., which has ADC and DAC together with anti-alias filters. StarCore MSC7116 is a fixed point 16 bit Word length DSP. SC1400 core has four Arithmetic and Logic Units (ALU), capable of carrying out 1000 million multiplication and accumulation per second at 266 MHz. Each ALU has a multiplication and accumulation unit (MAC) for arithmetic operations, which can perform multiplication between two 16 bit data obtaining 32 bit results.

ANC implementation

The application program was developed in SmartDSP real time operating system. In the first stage secondary paths learning process is carried out (Fig. 5) with white noise as input signals. The filter coefficients of S^(z) were updated during a few seconds until the f error reached a stable value.

Once learning process has finished for each secondary path, ANC system began operation with the MFxLMS algorithm. FIR Filtering and LMS updating algorithms were developed in assembler language for maximum processing speed. Modulo addressing was used to produce efficient circular buffers in the filter output calculations).

Process optimization

Signal precision selected for secondary path learning process was 16 bits. To compute Ec. 1, the coefficient μ x e product was computed, using 16 bits for both variables. Only the 16 higher order bits out of the 32 bit result were used for the following calculations. Then that 16 bit result was multiplied with the 16 bit input signal filtered by the secondary path. For the result, once more the higher part from the 32 bit result was taken to update the coefficients.

In order to improve performance, several optimizations took place. The first system optimization assigned 32 bit precision for W(z) filter's coefficients. This lead to an alternative approach for coefficient updating. If only the

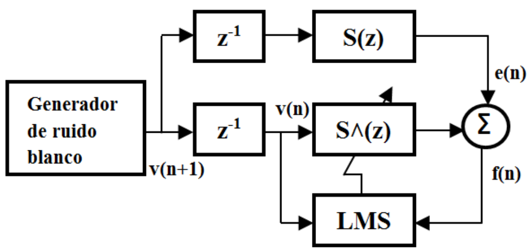

Fig. 5. Secondary paths identification scheme

higher 16 bit portion of the 32 bit result were considered from μ x e multiplication, adaptation would finish as soon as the updating signal starts lying on the lower 16 bits, loosing essential information. To solve this problem signed e(n) algorithm was implemented, which assigns the value "μ" to error if it is positive, and "-μ"otherwise (Kuo and Morgan, 1996).

Although convergence speed decreased, desired stability was obtained and the system perfectly canceled the initial harmonic components of noise signal. W(z) filters were implemented with 160 taps, while S(z) filters lengths were 400.

Next step was the four ALU parallel working. This reduced filtering and updating operations time. As a result, higher filter lengths could be accommodated in a sample period, improving noise attenuation. Even though DSP includes four parallel data movement instructions between the four ALU and memory, these are not prepared for modulo addressing. To enable the desired circular buffer scheme, filters and signals were defined in an unconventional but efficient way for the application. Filter dimensions were inverted, so in each quadruple multiplication and accumulation cycle four filters were involved in the four available ALUs, instead of performing 4 operations of a single filter.

Finally partial updating of W(z) filters coefficients was developed. This means only a fraction of the filter's coefficients are updated in a simple period. This results in lowering convergence speed (Kuo and Morgan, 1996), but enables the use of longer filters (W(z) filters with 500 taps), achieving a better performance.

RESULTS

In Fig. 6 mean squared error signals are shown. The peaks at t = 2s are due to stationary waves taking place when system operation begins. They are cancelled once adaptive process converges.

In Fig. 7 spectral analysis of error signal are shown when the system has already converged (grey) and without active noise cancelling (black). Attenuations obtained are listed in Table 5.

Finally C weighted noise levels were measured at several points inside the enclosure with TES 1352-A sound level meter. Comparing noise levels with and without active noise control allowed quiet zone characterization (Fig. 8).

CONCLUSIONS AND FUTURE WORK

To conclude, computational load required by multichannel ANC systems is considerably higher than in single channel systems, due to secondary paths involved. By reducing the number of secondary sources from 4 to 2, and improving direct/reverberant signal ratios, multiplications per sample period were reduced by a factor of 2.67.

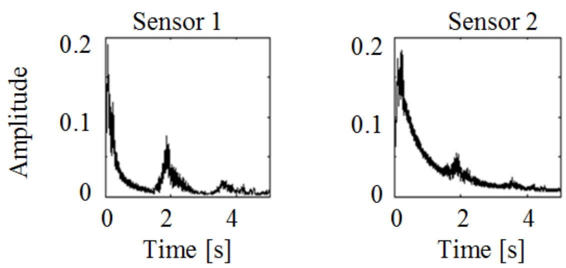

Fig. 6. MSE signals for the initial 5 seconds.

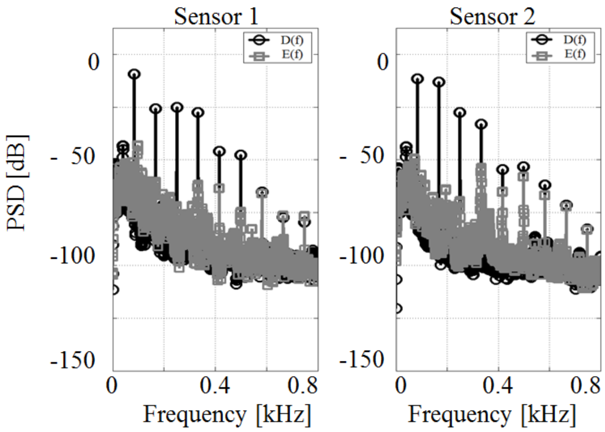

Fig. 7. Error signals PSD with ANC system off and on (D(f) and E(f)).

TABLE 5. Average attenuations obtained in error sensors with 2x2 ANC system.

Harmonic number	Frequency [Hz]	Attenuation [dB]
1	83	47
2	166	50
3	249	40
4	332	44
5	415	11
6	498	14
7	581	2

The selected DSP was acceptable for the application, but satisfactory results were obtained under certain limitations: size of the quiet zone is relatively small, main frequency of primary noise did not present any variation, and only engine noise was canceled. Even though main functions were developed in assembler language for maximum speed, partial updating became necessary. This allowed working with sufficient filter length for strongly cancelling initial noise harmonics, obtaining almost 50dB attenuations in first and second harmonic, and 40dB in third and fourth harmonic.

Finally we must emphasize that operating with two error sensors instead of four leads to smaller quiet zones, as expected. If we aim to obtain a wider quiet zone, more error sensors are required, which makes it necessary to work with more secondary sources as well. This means a considerably computational load increment, which impacts on DSP selection. Alternative solutions can be provided by the implementations of hard wired systems, where FPGA can increase the time efficiency and also the reliability of the system (Gaikwad et al, 2013).

As future work development of a feed forward system is proposed, so any signal noise correlated to a previously sensed reference can be canceled. Comparisons of feed forward and feedback techniques were carried out by (Gulyas et al, 2002) in agricultural machines. Another investigation path could include beam forming technique to allow noise cancelling in a particular spatial point without the need of placing a sensor in such position (Dmochowski and Goubran, 2004).

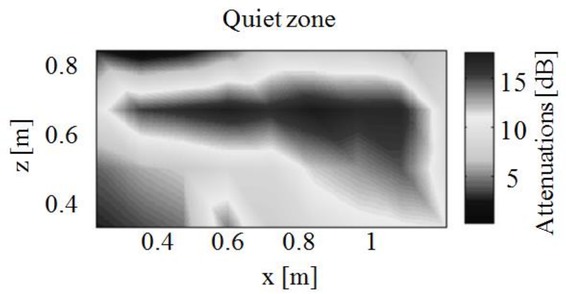

Fig. 8. Quiet zone in the area defined by error sensors and secondary sources.

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