# Acoustic Mapping for Active Transformer Quieting Applications Based on a Modular DSP Platform

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***Abstract*— The *Laboratoire d'Electronique Industrielle* has realized an active noise controller. The system is designed to efficiently attenuate low frequency transformer noise where passive methods are either ineffective or tend to be very expensive. This paper describes an acoustic mapping experiment: two loudspeakers simulate a transformer, the noise is measured by a 15 microphones antenna and the DSP calculates an acoustic map with the beamforming method. Simulation and experimental results are presented.**

## I. INTRODUCTION

Transformer noise has always been a concern for electrical companies. However, the growing public sensitivity to noise disturbances, the strengthening of the regulation due to these environmental issues and the growing urbanization around the transport and distribution (T&D) substations have recently raised more attention to this old problem [1].

In the past, three possibilities were offered to reduce the noise levels of these installations:

- Install the substation on larger premises.
- Replace old and noisy transformers with new generations ones.
- Build anti-noise screens around the transformer.

The theory of active noise control (ANC) has been developed in the 20's [2]. Its principle is very simple: noise is an acoustical wave that can be cancelled by an opposite wave. For decades, it has been very difficult to have a working system due to the required processing time between the measurement of the incident noise and the antinoise emission by the electroacoustic actuators.

Recent technological progress has overcome these difficulties: making the development of ANC systems possible. The noise is measured by microphones installed around the transformer. A digital signal processor (DSP) calculates the opposite wave applied by the radiating acoustic devices (RAD).

The *Laboratoire d'Electronique Industrielle* (LEI) has developed an optimized modular controller (in collaboration with IAV). The controller is built around a 40 Mhz DSP platform [3]. One module receives inputs from 16 microphones and controls 8 RAD outputs. The modularity gives a high adaptability to more or less complexe problems at a low cost.

This paper presents an acoustic mapping experiment using the beamforming method. Two loudspeakers simulate a transformer and an antenna composed of 15 microphones measures the noise. This method easily identifies the noise source and is thereby very helpful to place efficiently the RAD.

## II. MODULE DESCRIPTION

To actively reduce the noise, microphones and actuators are needed. Their number depends on the transformer characteristics. The modularity of the controller gives an economical solution from the simpliest to the most complexe case. In a complexe case, the modules communicate together to share their measurements and to coordinate their actions.

All modules (Fig. 1) are exactly the same. The heart of the system is the DSP performing the ANC or beamforming algorithm. A module has 16 input and 8 output channels. It is composed of four boards:

- The DSP platform with the A/D converters and the communication interface.
- Two signal conditionning boards. See § II-A.
- The power supply and the RAD drivers are on the last board. See § II-B.

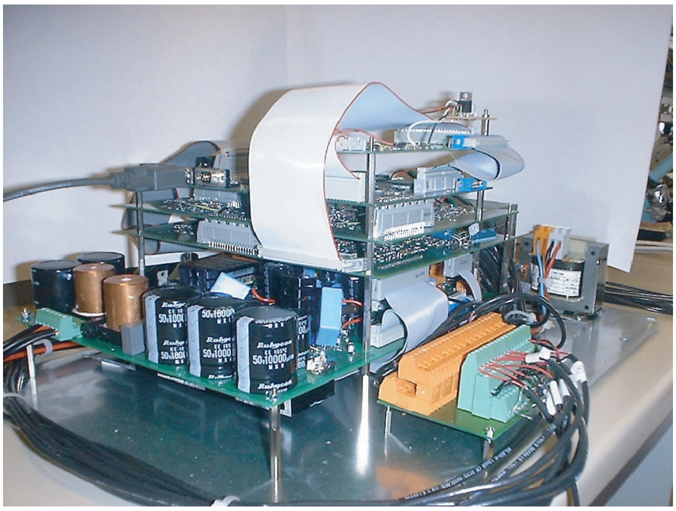

Fig. 1. The module hardware.

The transformer noise frequencies are mainly composed of harmonics 1 to 6 of the double network frequency (100 Hz to 600 Hz in Europe). The dominant harmonic is at 200 Hz. This defines the bandwidth of the signal conditionning and the frequencies of the antisound.

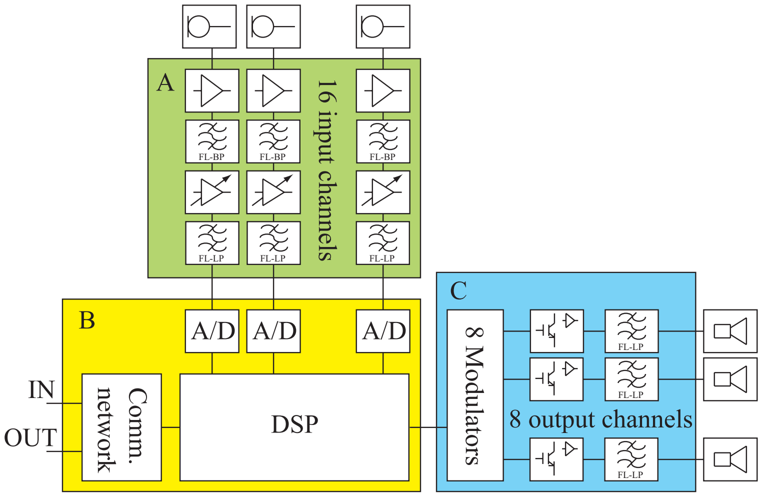


Fig. 2. Structural diagram of the controller. A: Conditionning of the 16 microphone inputs, B: DSP platform and C: Class D amplifier structure

### A. *Signal conditioning and acquisiton*

Microphone signals are small and noisy. A conditionning stage is used to amplify and filter the input signal.

It is first amplified and then filtered by a $4^{th}$ order passband filter whose passband corresponds to the bandpass is the transformer noise frequencies (100-600 Hz).

The amplitude of the input signal amplitude varies with the noise level and depends on the microphone location. Therefore, a programmable amplifier whose gain is automatically adapted by the DSP, adapts the signal level to the A/D dynamics.

Finally, a lowpass filter (1 kHz) based on a Sallen & Key cell removes the high frequency noise induced by the previous stage before the signal is digitized by A/D converter to be processed by the DSP. The conditioning steps are illustrated in Fig. 2 A.

### B. *Control of the Radiating Acoustic Devices*

The ANC system cancels the unwanted noise based on the principle of superposition. An anti-noise is generated and combined with the primary noise. The phase, frequency and amplitude of this wave must be controlled independantly for each channel. The RADs (Fig. 5) have been specially developped to work in the noise frequency range (100-600 Hz).

A compact module requires high efficiency amplifiers. Class D amplifiers have been selected for their low power dissipation. Their output transistors operate as a switch: no current flows when it is off and the voltage drop is very low when on.

The amplifier is divided in three main parts (Fig. 2 C):

- A digital Pulse-Width-Modulated (PWM) modulator drives the transistors. A triangular wave is compared to the DSP set point. The DSP controls the anti-noise amplitude, frequency and phase by controlling the set point. The wave quality is maximal when a new set point is given at the pulse frequency.

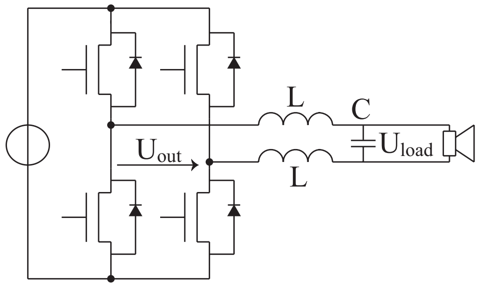


Fig. 3. The power stage and the output filter

- The power stage is made of an H-bridge structure (Fig. 3). It allows a balanced output stage with a single power supply. High speed transistors are used to reduce distorsion.
- A filter demodulates the output waveform of the power stage. This filter has to be carefully designed to increase the quality of the anti-noise. The transfer function is defined by (1).

$$\frac{U_{out}}{U_{load}}(\omega) = \frac{1}{1 - 2LC\omega^2 + j\dfrac{L}{R_{load}}\omega} \tag{1}$$

In the case of a Butterworth filter, we obtain the values of the inductances L (2) and the capacity C (3):

$$L = \frac{\sqrt{2}R_{load}}{4\pi f_c} \tag{2}$$

$$C = \frac{1}{2\sqrt{2}R_{load}f_c} \tag{3}$$

where $f_c$ is the cutting frequency, $U_{out}$ and $U_{load}$ are the power stage output voltage and the load voltage respectively, $R_{load}$ is the load resistance.

### C. Communication and synchronization

Communication is needed in large transformer quieting applications. Indeed, the controllers will share their measurements and coordinate their actions to cancel the noise. In a multi-controller system, the measurements must be synchronied, otherwise the ANC algorithm will be inefficient. Data exchange and synchronization is done through a special communication network.

The modules are connected in a ring where a module sends data to another module and receives from another one. A master supervises the network and initiates the transmission by sending its data package and then giving a token to its neighbor. When a slave receives a message, it immediately transfer it to the next module. If it receives the token, it begins to send its own package. Once the master receives the token, the frame transmission is over and the master is ready to start a new frame. A frame is composed of a package from every active module. Inactive modules only transfer the message and the token.

Every package contains the identification of the sender, some control bits, N bytes of data (N is configurable but is the same for every module) and an error detection code (Cyclic Redundancy Code). If necessary, the token is added at the end. The 8 Mbit/s transmission is asynchronous to limit the wiring to one twisted pair. The network logic is implemented in a Field Programmable Gate Array (FPGA).

This is a real time and completly deterministic network. The synchronization is done at the end of each frame, which is possible because each frame has exactly the same length. The network configuration and the frame format are illustrated in Fig. 4.

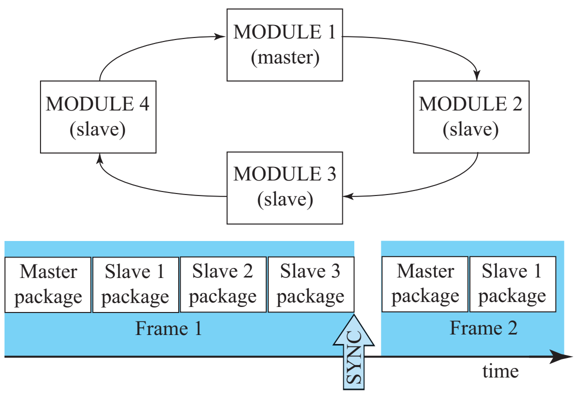


Fig. 4. network configuration and frame format

Data and commands are transmitted through this network. The commands have a higher priority than the data. Because the package size is limited and the data messages are larger, they are cut into several parts and send separetely. Commands are small enough so that several of them can fit into one package.

## III. Acoustic Mapping

In a first approach, the DSP platform has been used to perform an acoustic mapping experiment. A beamforming algorithm was developed using Matlab and then translated in DSP language.

The beamformer is composed of a rectangular array of 15 microphones (5 X 3) equally distanced at 75cm. The antenna is placed 5 m away from two loudspeaker sources simulating a transformer (Fig. 5). According to the Nyquist anti-aliasing criteria, the array is designed to handle the 200 Hz harmonic.

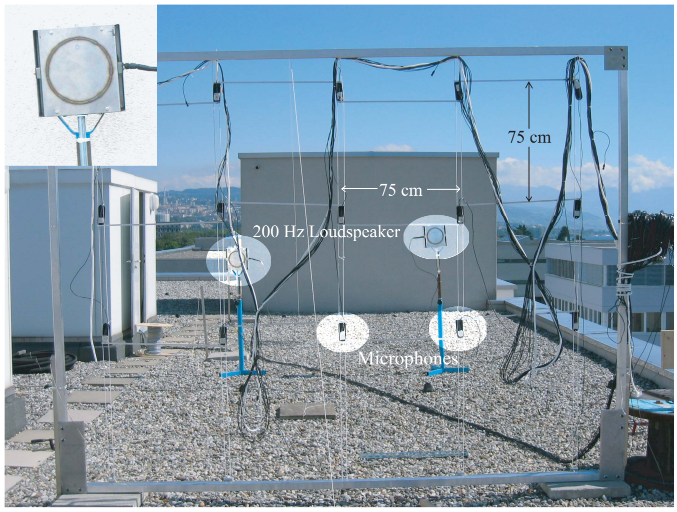


Fig. 5. Picture of the microphones antenna with two RADs in the background (one RAD in the left corner)

### A. Beamforming principles

The acoustic mapping is obtained by "focalizing" the antenna on every cell of a virtual grid coinciding with the transformer. The spatial filtering is obtained by adequately delaying the 15 signals, compensating the level through a "1/r" energy decrease law and summing the results [4].

(4) gives the propagation time $\tau$ between the cell and the microphone:

$$\tau_{cell,mic} = \frac{1}{c}\left(d_{cell,mic} - d_{cell,center}\right) \tag{4}$$

where $d_{cell,mic}$ is the distance between the cell and the microphone and $d_{cell,center}$ is the distance between the cell and the antenna center. $Gain_{cell,mic}$ (5) is the ratio between $d_{cell,mic}$ and the distance of focalization $d_{foc}$.

$$Gain_{cell,mic} = \frac{d_{cell,mic}}{d_{foc}} \tag{5}$$

For computational efficiency, the time delay operations are performed in the Fourier Domain (6).

$$\boldsymbol{XFFT}_{cell,mic} = \mathrm{fft}(\boldsymbol{x}_{mic})\, Gain_{cell,mic}\, e^{-j\,2\,\pi \boldsymbol{f}\,\tau_{cell,mic}} \tag{6}$$

where $\mathrm{fft}(\boldsymbol{x}_{mic})$ is the fast Fourrier transform (fft) of the microphone signal $\boldsymbol{x}_{mic}$ and the frequency vector $\boldsymbol{f}$ is given by (7) and (8).

$$\boldsymbol{f} = \begin{pmatrix} f(0) \\ \dots \\ f(N_{fft}-1) \end{pmatrix} \tag{7}$$

$$f(i) = \begin{cases} f_s \frac{i}{N_{fft}} & \text{when} \quad 0 \le i < \frac{N_{fft}}{2} \\ -f_s \frac{N_{fft}-i}{N_{fft}} & \text{when} \quad \frac{N_{fft}}{2} \le i < N_{fft} \end{cases} \tag{8}$$

where $N_{fft}$ is the number of samples.

The contribution of a given microphone (9) for a given cell is obtain by the inverse fast Fourrier transform (ifft) of (6).

$$\boldsymbol{Xmic}_{cell,mic} = \text{real}(\text{ifft}(\boldsymbol{XFFT}_{cell,mic})) \tag{9}$$

The level of a particular cell (10) is the sum of the square of each microphone contribution (9):

$$Level_{cell} = 10\log_{10} \sum_{mic=0}^{N_{mic}} \sum_{i=0}^{N_{fft}} Xmic^2_{cell,mic}(i) \tag{10}$$

The output of the beamformer is then expressed in dB relative to the maximum value and mapped on the cells grid. A mapping simulation of a 3-sources modelled transformer is presented on Fig. 6.

### B. Experimental results

Acoustic maps have been realized with the experimental setup described above (Fig. 5). The 15 microphones and two RADs were connected to a single module. The two RADs simulated the transformer noise at a frequency of 200 Hz. Three cases are presented: one RAD on at a time (Fig. 7 and 8) and both RADs on (Fig. 9).

For each case, three maps are presented: the first obtained by simulation, another illustrates the experimental results and the last shows the differece between them.

The error of the source center position is linked to:

- Imprecision of the position of the source and the microphones.
- Inaccuracy of the hypothesis on the speed of sound ($c = 343[m/s]$): the speed varies with the temperature.
- Non-anechoic environment: reflections are present on the walls and ground.
- Meteorological conditions: the wind creates turbulences and moves the microphones inducing a phase error.

The larger size of the source in the experimental plots is due to:

- Imprecision of the position of the microphones.
- Pairing of the 15 input channels in amplitude and phase: since the gain is programmable, good calibration is very difficult to achieve.

The errors observed are of the same order of magnitude as the sum of all the above inaccuracies.

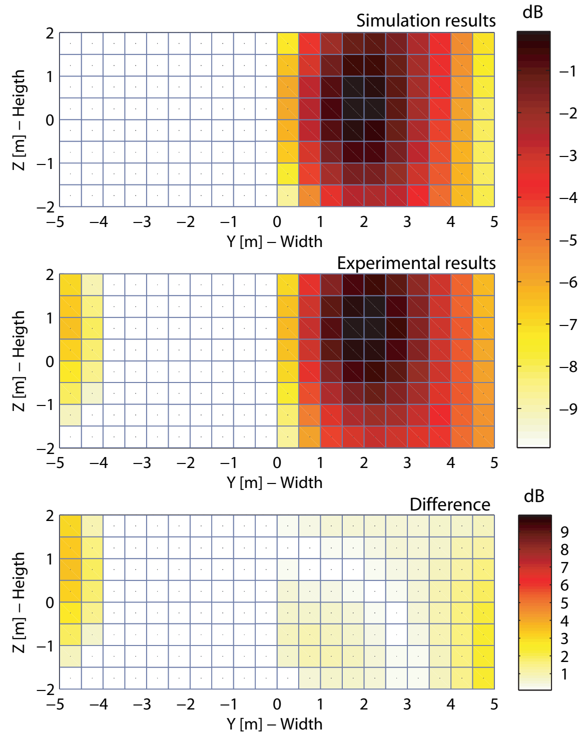


Fig. 7. First case: the right RAD is one (200 Hz). The first map is the simulation results, the second is the experimental results and the third shows the difference between the two.

## IV. Conclusion

This paper has presented an acoustic mapping application based on a modular DSP platform. The frequencies used match those found in T&D transformers. The beamforming algorithm has been explained and experimental results have been presented and compared to simulation results. This application demonstrates the functionalities of the modular hardware developped for the active transformer quieting system.

The modular controllers are now ready for the next step: the implementation of an active noise control algorithm inside the DSP. In the mean time, the acoustic mapping could be particulary useful to reduce the effort necessary to the dimensioning of a noise reduction solution (optimization of the number and positioning of the actuators).

## V. Acknowledgment


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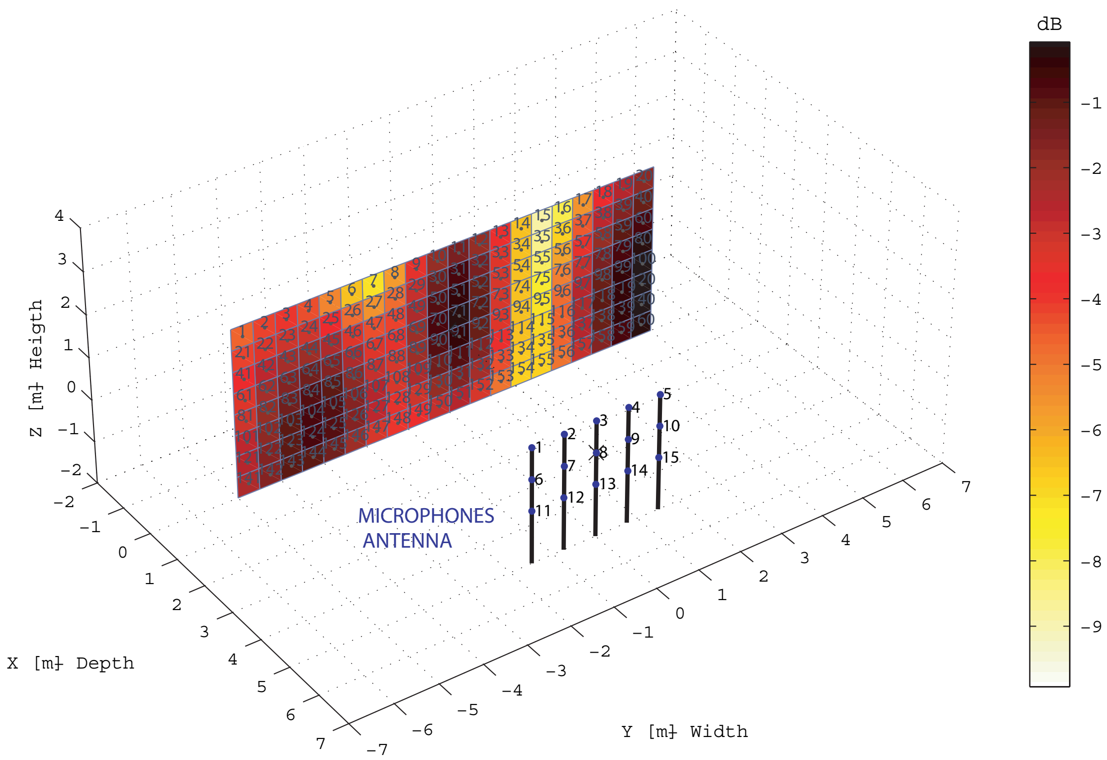


Fig. 6. Mapping simulation of a 3-sources modelised transformer

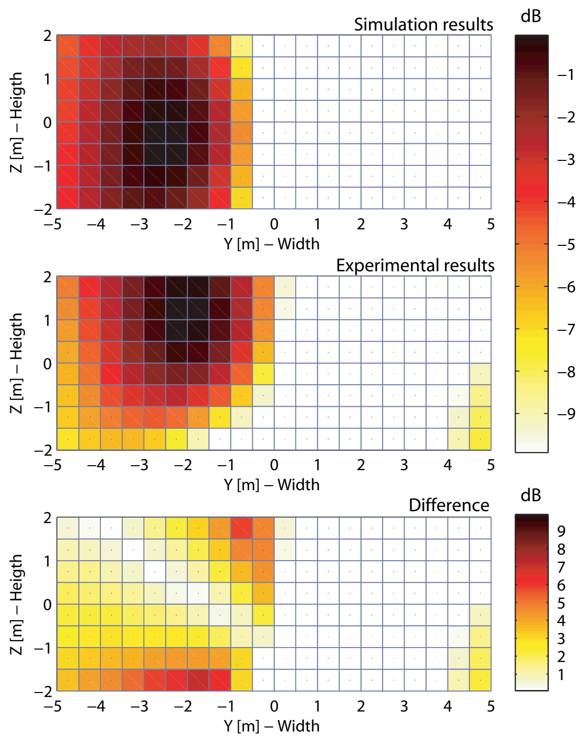


Fig. 8. Second case: the left RAD is one (200 Hz). The first map is the simulation results, the second is the experimental results and the third shows the difference between the two.

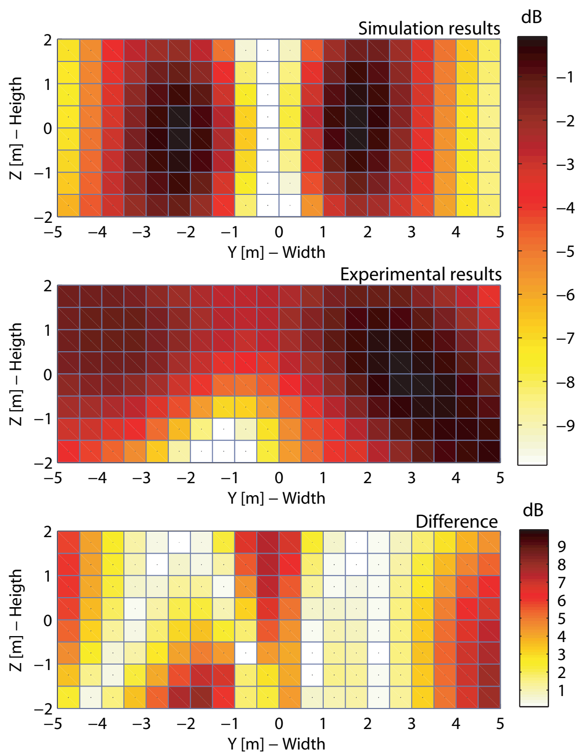


Fig. 9. Third case: both RAD are one (200 Hz). The first map is the simulation results, the second is the experimental results and the third shows the difference between the two.