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# APPLICATION OF REAL-TIME RECURSIVE FILTERS TO ACTIVE CONTROL OF SOUND IN FINITE LENGTH DUCTS

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## ABSTRACT

This paper discusses the active control of noise in a finite length duct. Although the case of infinite ducts is efficiently treated by a controller designed as a finite impulse response filter (FIR), it is shown that finite length arrangements require an infinite impulse response controller (IIR). A laboratory air system comprised of a home made (IIR) controller is presented. A calculation of the optimal controller transfer function obtained from system responses measurements is shown to result in a broad-band noise attenuation of 10-20 dB. Similar investigations were made in a water system, leading to up to 10 dB attenuation of broadband noise. In the aim of implementing adaptive systems, recursive least mean squares (RLMS) models of system responses are also presented.

## INTRODUCTION

Active control of noise in single degree of freedom systems has often been studied and today, numerous laboratory experiments are being carried out leading to convincing results <1>,<2>. These first experiments are conducted on long ducts -with a low stationnary waves ratio- in which the acoustic feedback is substancially reduced by using unidirectional detectors and an anechoïc design of upstream terminations. With that kind of quasi-ideal set-up, the propagation time makes it possible to design the electronic controller as a Finite Impulse Response (FIR) digital filter. The implementation of adaptive controller using frequency domain algorithm is well suited to FIR filters and provides reliable active absorbers. However the problem is more intricated, if we attempt to achieve active noise control in a short open ended duct or in water systems. The presence of stationnary waves together with the higher sound speed cause instabilities which cannot be avoided by FIR filters. As a result, performances are drastically reduced. It is therefore necessary to use Infinite Impulse Response (IIR) filters although they require more sophisticated electronics and control algorithms.

In this paper, we describe an experiment in a short open ended duct, using a home made versatile controller. It is shown that the use of IIR filters can cope with many cases of acoustic feedback in which the FIR technique had failed before. This system has been used in two cases of air and water, and a significant attenuation has been obtained.

In order to respond to eventual changes in sound speed and noise characteristics, the system must have a great adaptability. Insofar as there is no systematic way to determine the coefficients which directly provide the convenient controller transfer function, we have done recursive LMS identification of the main elements of the plant. The whole controller has then to be a combination of these elements. Its adaptability will be obtained by the adaptability of each element.

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## 1. CONTROLLER DESCRIPTION

A microprocessor is needed for the implementation of a real time IIR filter whose basical behavior law takes both input and output samples into account. A powerful signal processor (8 MIPS) has then been used in a home made digital signal processing system. Its main advantage consists in a high speed processing due to a simultaneous data-memory and program-memory access at each clock cycle. The analog signals are processed with 12 bits conversions. The entire system is driven by a personal computer which achieves both computation and management of the filter coefficients. The signal processing is controlled by the sampling clock which handles a microprocessor interrupt. The program performs a cascade biquad IIR filter which guaranties a better accuracy for the coefficients than the direct form realization of an IIR filter does.

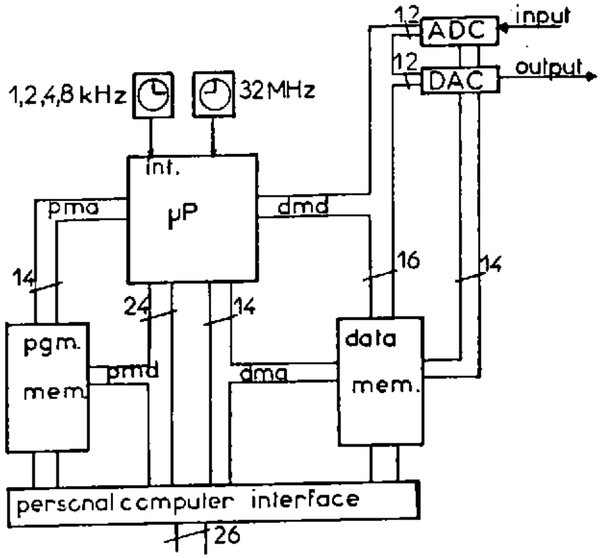


Figure 1 : Controller block diagram

## 2. AIR SYSTEM EXPERIMENTS

The experimental set-up (Fig. 2) consists of an open ended duct. The total length is 1,60m and the cross section 0,20m * 0,30m giving a cutoff frequency of about 850 Hz. The detector and error sensor are electret omnidirectional microphones. The sources are currently available loudspeakers, mounted with fiberglass filled enclosures.

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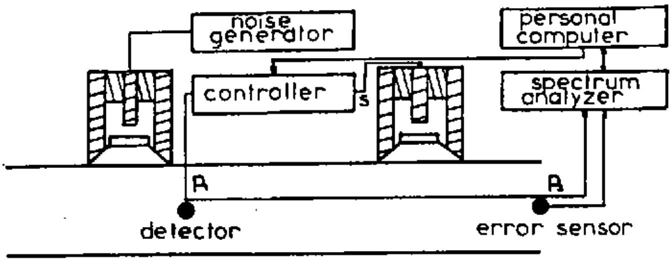


Figure 2 : Air System arrangement

The main problem here is to determine the optimal controller characteristics in order to implement the best active cancellation in a given configuration. The methods we can choose here are either a frequency identification of the controller optimal transfer function or a time identification. Insofar as this second method requires a complete modelling of the system basical elements, we have first implemented a frequency technique.

The calculation of the optimal transfer function which takes into acount the feedback has often been developed : for exampie by Ross <3>. The value of this transfer function, H(f), may be written as

$$H(f) = H_0(f)/(H_0(f)H_1(f)-H_2(f))$$

where
- $H_0(f)$ is the transfer function between detector and error sensor when the attenuator is off;
- $H_1(f)$ is the transfer function between detector and cancelling source signal, s (see Fig. 2), when the cancelling source alone is on;
- $H_2(f)$ is the transfer function between error sensor and cancelling source signal, when the cancelling source alone is on.

Given the value of the optimal transfer function, the problem lies in the determination of the IIR filter coefficients which allow the best agreement between the obtained and desired transfer function. This has been achieved by an iterative empirical programme. This programme allows the user to place poles and zeros in the frequency domain and to adjust their module. The main advantage of this method is that it necessarily provides a stable digital IIR filter. It is then easy to define an IIR coefficients set capable of a good approach of the optimal transfer function (Fig.3). It is worth noting here that the synthesis of the optimal transfer function could not be realized by a FIR filter.

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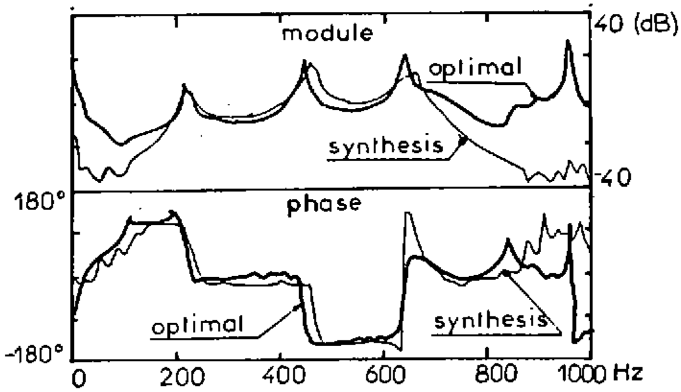


Figure 3 : Optimal transfer function

Indeed, an inverse discrete Fourier Transform shows an impulse response that could not be reached by a real time FIR filter, because of the non-zero values in the negative time domain (Fig.4).

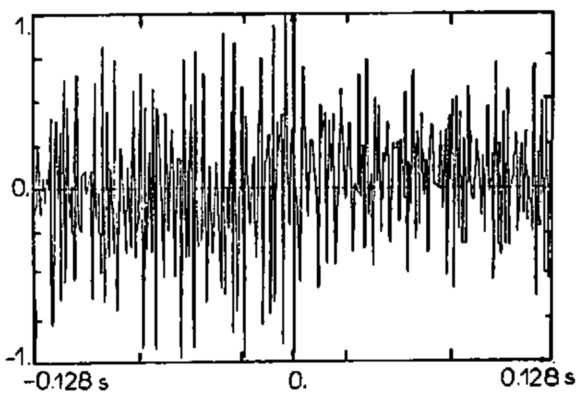


Figure 4 : Optimal transfer function's inverse Fourier Transform

The application of this identification to our electronic programmable controller with a sampling frequency of around 4 kHz and 30 coefficients in 6 biquad cells result in a significant attenuation (up to 20 dB) of a broadband noise (Fig.5). The frequency domain in which the system is efficient correspond exactly to the domain where the error between desired and obtained transfer functions is sufficiently low. It is to be noted that here, acoustic feedback is well taken into account by the system which does not generate any stability.

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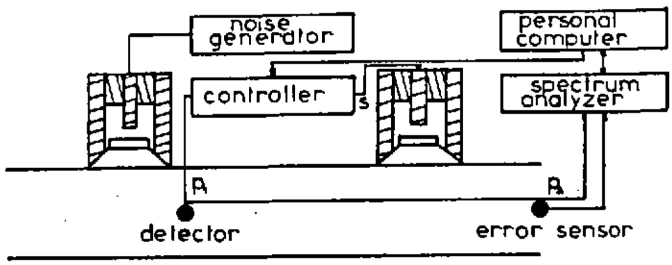


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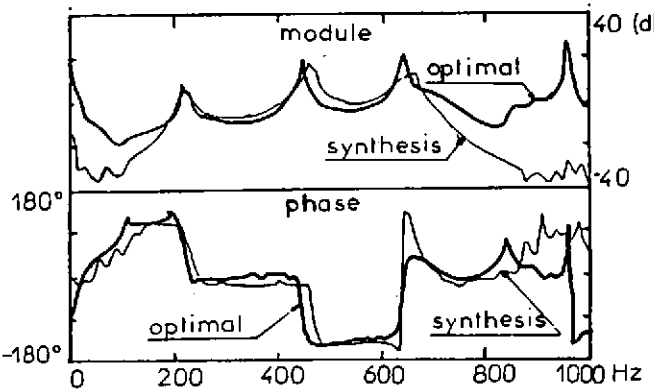


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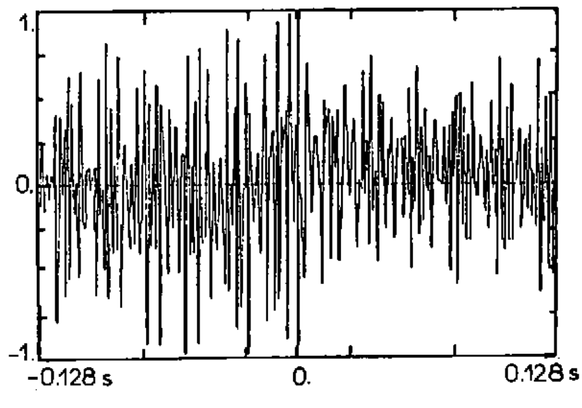


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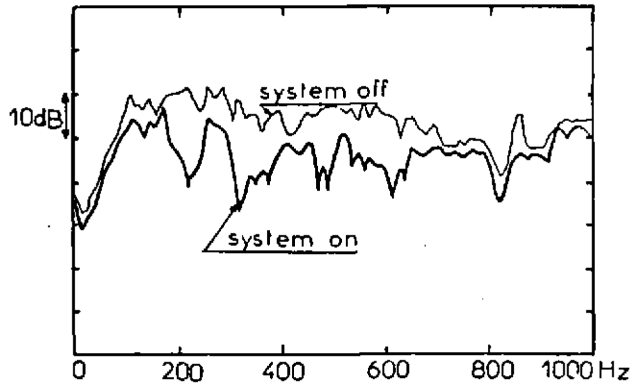


Figure 5 : Acoustic pressure power spectra at the error sensor

## 3. WATER SYSTEM IMPLEMENTATION

Water systems are of great interest since in many hydraulic circuits, there is almost no possibility of passive attenuation. Thus, attempt has been made of reducing broadband noise in a steel water-duct with the system depicted in figure 6. The detection is taken directly as the motion of the primary source.

The difficulty here successively lies in the high sound speed, the important density of acoustic and vibration modes, and the presence of strong reactive components in the pressure field. We have then limited this study to lower frequency ranges in order to improve the transfer function measurement accuracy.

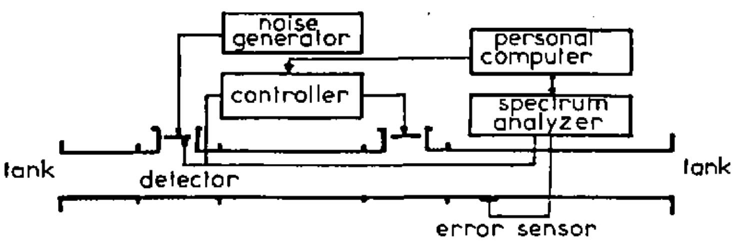


Figure 6 : Water System arrangement

A technique similar to that shown above has provided an attenuation of around 10 dB (Fig.7). Here again, the error between the desired and obtained transfer functions has a direct effect on the attenuation (Fig. 8).

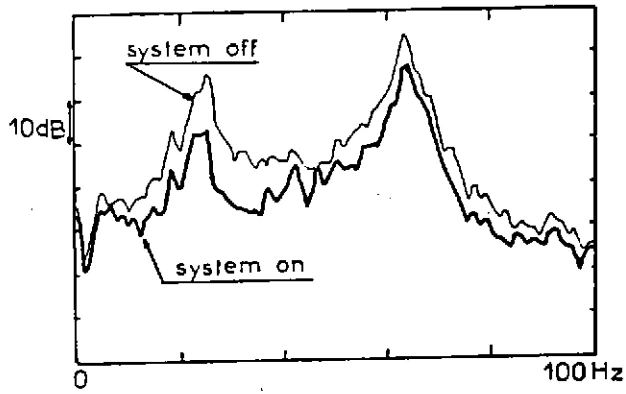


Figure 7 : Acoustic pressure power spectra at the error sensor

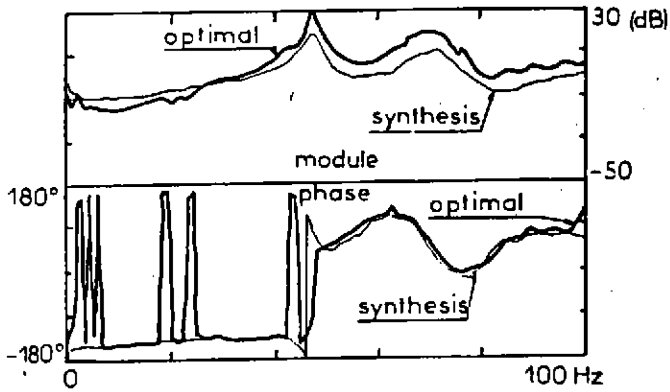


Figure 8 : Optimal transfer function

4. ACOUSTIC PLANT RECURSIVE LMS MODELLING

Since physical systems are generally prone to the drift of their mean characteristics, an active absorber has to be adaptable. It is then necessary to implement adaptive IIR filters, for which theoritical studies have been developed (Feintuch <4>,Parkish <5>). The first step of these methods requires a model of the plant that will be adapted to algorithms using time samples of input, output, and error. We have thus made first investigations in the RLMS modelling of acoustic paths in the duct, represented by $H_0$, $H_1$, and $H_2$ in §2.

Given the input-output typical equation of an IIR filter in the matrix form

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$$\underline{Y} = U \, \underline{B}$$

where $\underline{Y} = ( y_n , y_{n-1} , \ldots , y_{n-L} )$
output vector

$\underline{B} = ( -b_1 , \ldots , -b_N , a_0 , \ldots , a_M )$
coefficients vector

U is a block Toeplitz (N+M,L) matrix containing input and output samples

Sorenson <6> shows that the LMS coefficients which can fit an experimental process $\underline{Ye}$ may be written :

$$B_{LMS} = ( {}^tU \, U )^{-1} \; {}^tU \, \underline{Ye}$$

Application of this method has been achieved in the laboratoty air system. A case with 30 feedforward and 30 feedback taps has been treated. The training signal used in the system for this identification is a periodic impulsive signal with a fundamental frequency of 16 Hz. The signals were measured with a sampling frequency of 4 kHz and a sample length of 1024 points included in a home made data acquisition system. Satisfactory agreement between the actual and the modelled transfer function has been obtained. An example of the results for $H_0$ is given in figure 9.

The next step is the combination of these elementary modules in order to obtaine the operative controller. It is worth to notice that an adaptive system will be then provided by an available gradient method (Ffowcs-Williams <7>).

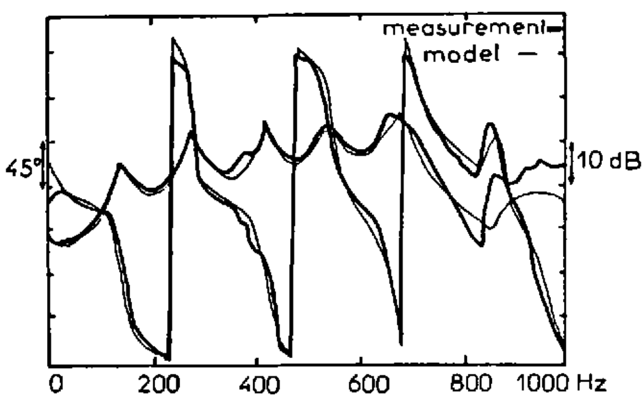


Figure 9 : Error sensor to detector transfer function

## 5. CONCLUSIONS

In this paper, we have examined the use of IIR filters in the implementation of active noise control. We have shown that this filtering technique is particularly adapted to systems where feedback between the cancelling source and the detector occurs and to arrangements comprising a high stationnary waves ratio (such as water systems). Significant attenuation has been obtained in both situations. A LMS modeling of the air system has provided promising results which will be applied in the near future to an adaptive complete system.

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