Measure and identification of acoustic impulse responses by NLMS-DC

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Abstract

In this work, we have the practical results for a realization of a system of measurement of acoustic impulse responses. The realization of a measuring bench makes it possible to measure the acoustic impulse response of a room with a dynamics of 48 dB by the method of identification by inter correlation of the signals tests. The latter proved to be better than the identification by the algorithm of the stochastic gradient with decreasing step (NLMS-DC) especially for the identification of impulse responses in the noised mediums.

Key words

Impulse response, acoustic Channel, Intercorrelation, adaptive Algorithm, Identification, Acoustics.

1. Introduction.

The acoustic phenomena in an acoustic medium are measured starting from the notion of the acoustic channel (fig.1). This last depends on three principal elements[1]:

- the shape and acoustic properties of the walls and the objects of the room.
- the source with its spectral emission diagram and its position (LS: loudspeaker).
- the receiver with its diagram of directivity and its position (MIC: microphone) [2].

In the approximation of linear acoustics this channel is a linear filter whose entry is the signal x(t) and the output y(t)[3]:

$$y(t) = h(t) * x(t) \qquad (1)$$

Where *h(t)* is the impulse response (IR) acoustic channel..

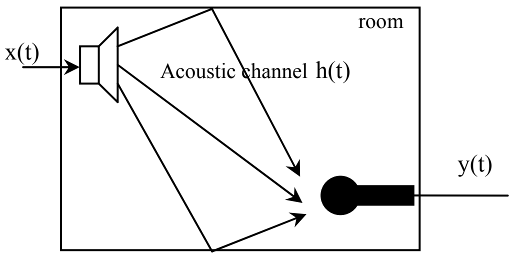

Figure 1 : Acoustic channel.

The goal of measurement is to find the IR *h(t)* which completely characterizes. In rooms acoustics , these IR have a duration of about a second ,and the desired dynamics are about 60 dB. The traditional measurement technique which generates an impulse and records the IR does not make it possible to answer all these requirements simultaneously[4]. Moreover, the risk increases considerably when the transducers (LP and MIC) are subjected to impulses. We describe two measurement methods which are better adapted to the requirements of the measurement of the IR.

2. IR measurement principle.

The general flowchart of the method of measurement is given on figure 2. The essential element of this method is the identification of the impulse response *h(t)* starting from the two signals *x(t)* and *y(t)* [5].

2.1 Identification by the inter correlation method.

The function of inter correlation between the signals x(t) and y(t) is written:

$$R_{YX}(\tau) = R_{XY}(\tau) * h(\tau) \qquad (2)$$

When the function of autocorrelation of the input signal is proportional to the impulse of Dirac, we have:

$$R_{yx}(\tau) = h(\tau)\sigma_x^2. \quad R_{xx}(\tau)=\sigma_x^2.\delta(\tau) \qquad (3)$$

with σ_x^2 is the energy of the signal *x(t)* .

Consequently, the IR is obtained by a simple calculation of the function of inter correlation of the input-output signals. For the continuation, one notes *x(n), y(n)* and *h(n)* the sampled versions, at the frequency F_E, of the signals *x(t), y(t) and h(t)*.

In practice, the signals whose function of autocorrelation approaches a distribution of Dirac are the random pseudo binary sequences (RPBS) [6]. The signal of excitation x(n) is built starting from a periodic sequence RPBS of value 0 and 1 generated by an m-stages feedback register (fig.3) . In suitably choosing the configuration of the shift register, one can generate a maximum length RPBS sequence of period $L = 2^m - 1$. This sequence of 0 and 1 is then converted into of $\pm V$ volts rectangles. The signal x(n) obtained is periodic of period $T = L/F_E$.

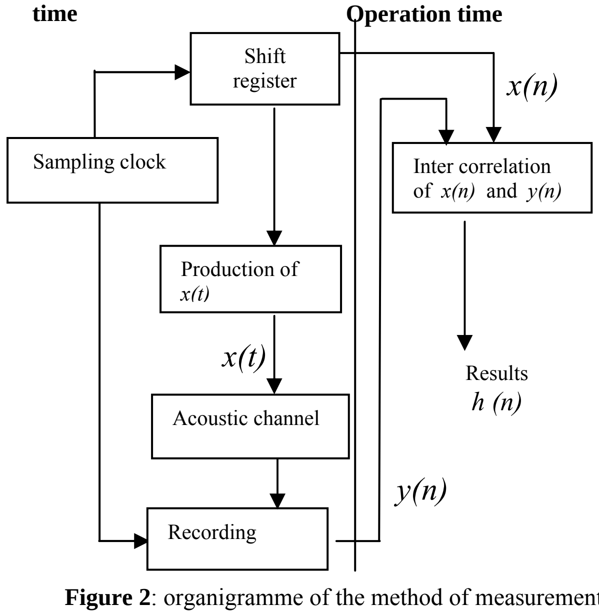

Figure 2: organigramme of the method of measurement

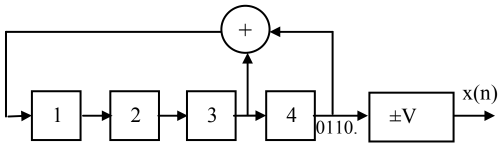

Figure 3 : Maximal RPBS sequence , m=4

The output signal y(n) is in absence of auditive noise, of the same period as x(n). The function of inter correlation $R_{YX}(k)$, being of the same period as x(n) and y(n), can be calculated on an average of several periods L:

$$R_{YX}(k) = \frac{1}{pL}\sum_{n=1}^{pL} y(n)x(n-k) \quad (4)$$

where p indicates the number of periods L.
The estimate of the impulse response is obtained by the following formula:

$$R_{YX}(k) = \frac{R_{YX}(k)}{\sigma_x^2}, \quad k=1,\dots,N \quad (5)$$

where N is the estimated number of points. The estimate of σ_x^2 is given by:

$$\sigma_x^2 = \frac{1}{pL}\sum_{n=1}^{pL} x(n)^2 \quad (6)$$

The parameters in this method are the length L of the sequence, the number p of periods and the size N of the identified IR.

2.2 Identification by the stochastic gradient method.

The adaptif identification of the IR system is represented on figure 4.

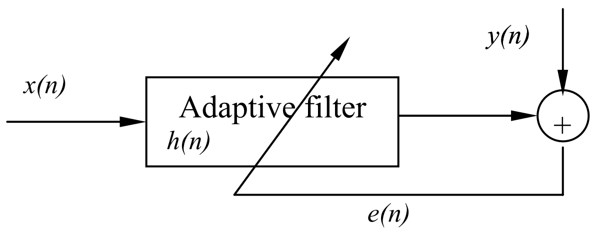

Figure4 :Identification adaptive

The error at the output of this system is written:

$$e(n) = y(n) - H^T(n-t)X(n)$$

where H is a vector column which gather the first N values of the identified impulse response and $X(n)$ a vector which summarizes the past of the signal [7] :

$$X^T(n) = [x(n), x(n-1), \dots, x(n-N+1]$$

The IR is calculated by minimization of the average quadratic error EQM:

$$EQM = E(e(n)^2)$$

The recursive solution of this problem is given by the normalized stochastic gradient algorithm (NLMS)[3]:

$$H(n) = H(n-1) + \frac{\alpha X(n)e(n)}{X^T(n)X(n)} \quad (7)$$

where $0 < \alpha \le 1$ is the step of adaptation of the algorithm[8]. To weaken the average quadratic error of the steady operation of the algorithm, we used a version of algorithm NLMS with decreasing steps (NLMS-DC) towards 0 for which the adaptation step is given by :

$$\alpha = \begin{cases} 0.5 & \text{for } n \le \frac{n_x}{2} \\ \frac{0.5}{n - \frac{n_x}{2}} & \text{for } n > \frac{n_x}{2} \end{cases}$$

where N_X indicates the total size in samples of the signal $x(n)$.
For N = N_X / 2, the choice of the step of adaptation α = 0.5 allows a fast convergence of the algorithm. After the moment N_X / 2, the decreasing step towards zero makes it possible to obtain a weak average quadratic error in steady operation[9].
In this method, the signal $x(n)$ is not obligatorily a RPBSA sequence. To allow the comparison thereafter, we use the same signal of excitation $x(n)$.

3. Realization of the measuring bench.

The measuring bench that we carried out in practice is illustrated by the synoptic of figure 5.

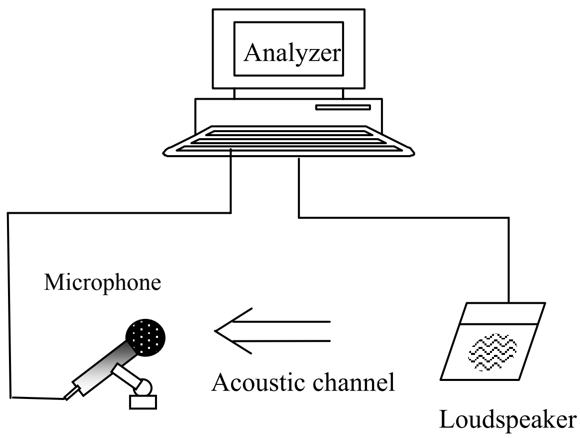

Figure 5: S chema of the measuring bench

Equipment used the measuring bench is described as follows:

- A standard bi-channel real time frequency analyser (2133B&K),(B&K=Bruel & Kjaer).We used this analysis for, on the one hand, to generate the signal of excitation which is a pseudo-random white vibration $x(n)$ of frequency band equal to 5KHz and of effective amplitude 2volts. In addition, this analyzer allows acquisition in real time and synchronism with the excitation x(n), of the signal of response of the room $y(n)$ [10].

- A loudspeaker of the type (4224B&K) transforms the signal of excitation x(n) into a sound pressure which will be diffused in the room [11].
- A microphone with standard rotary arm (3923B&K) which makes it possible to convert the acoustic pressures into exploitable electric signals on the spectrum analyser. The microphones - high speaker couple defines the acoustic channel of the room [12].
- An integrating modular sonometer of standard precision (2131B&K) which makes it possible to measure the existing ambient sound level in the room during the measure [1].

The measurement of the impulse response of the acoustic channel is taken in two stages:

- Acquisition in synchronism of the signals x(n) with the materials described above.
- Identification in delayed time of the impulse response by the methods of the stochastic gradient and method of Interco relation.

4. Experimental results.

We always used the same standard two-channels of frequency analyzer (4224B&K) as well as the acoustic medium (room of measurement). But we tested several couples of microphones/loudspeakers.

Dimensions of the room used are: 8 meters of width, 0meters length and.5meters height. The criterion of performance used for the comparison between various measurements is the signal to noise ratio (SNR) which is calculated as follows:

- Method of inter correlation: the SNR is calculated by the ratio of the energy of the beginning of the IR on the energy of the tail of the IR.
- Algorithm NLMS-CD: the SNR is calculated by the ratio of the energy of the signal *y(n)* on the energy of the signal of error[9].

4-1 Influence of the type of LS on the measure

In this manipulation, we evaluated the effect of the type of LS to the measure. For that we used 3 types of LS, the first is a professional sound source of type (4224 B & K) the second of mark (MICDIS of power 100w is) and the third is of mark (TNGY of power 80w).

The parameters of measurement are:

- x(n) is a sequence of white noise (6 sequences of 1 second).
- The distance which separates the LS and the MIC is d= 3meters.
- Sensitivity of the analyzer of frequency is of 50mV/pa .

The identified impulse response is of size 2500 points.

The results of this measurement are summarized on table 1. This manipulation shows that the use of professional LS gives an identification of the impulse response better than the use of the general public LS which reproduces the low frequencies badly.

Table 1: SNR Calculation (dB) of the measure

Type of LS used	SNR (dB)	
	Inter correlation	NLMS-DC
LS TNGY 80w	46.68	32.00
LS MICDIS 100w	46.40	34.50
LS (4224 B&K)	47.67	45.00

4-2 Influence of the distance (d) which separates the LS and the MIC on the measure.

In this manipulation, we evaluated the effect of the distance (d) which separates the LS and the MIC on the measure. For that we used:

- LS (4224B&K).
- White noise sequence: 6 one-second pseudo-random sequences .
- One MIC type (4165 B&K)
- Sensitivity of the analyzer of frequency is 52mv/pa. Impulse response length : N=2500.

The results of this manipulation are given on table 2. During this manipulation we obtained results which show that the Interco relation identifies the IR with constant SNR for all the distances.

That is due to its robustness to the noise. On the other hand algorithm NLMS-DC identifies better the IR for small distances where the MIC receive only the direct waves (great powers) which shows that this algorithm is sensitive to the noise.

Table 2: SNR Calculation (dB) of the measure

distance d (meter)	SNR (dB)	
	Inter corrélation	NLMS-DC
d = 1.5	47.07	48.00
d = 3	47.67	45.00
d = 6	46.92	45.00

4-3 Influence of the length of the noise sequence on the measure.

We evaluated the effect of the length of the noise sequence of the signal of excitation on the measure by using the following parameters of measurement:

- LS (4224).
- One MIC type (4165).
- distance D = 3 meters.
- Sensitivity of the analyzer of frequency is 50mv/pa.
- Impulse response length N = 2500.

During this manipulation we obtained results which shows that a better measurement of the impulse response must be done with a signal of excitation x(n) represented by a sequence of noise of significant duration (order one second) and several periods for the method of inter correlation.

The identification by stochastic gradient does not require a periodicity of the signal of excitation x(n) for table 3.

Table 3: SNR Calculation (dB) of measure.

Sequence of the noise	SNR (dB)	
	Inter correlation	NLMS-DC
6s de 1s	47.67	45.00
3s de 2s	47.70	40.27
6s random	34.32	37.00

Conclusion .

We realised a measuring bench of acoustical impulse response using professional acoustic material (material Bruel & Kjaer). The practical impulse responses obtained have an average dynamics of 48 dB. The same measurements were taken with two cartes (SoundBlaster), the dynamics of the measure in this case does not exceed 10 dB.

The realization of this bench measuring enabled us to conclude that the method of inter correlation identifies better the acoustic impulse responses in the noised mediums than the algorithm of the stochastic gradient with decreasing step.
These results can be improved by taking more precautions to minimize the effect of the ambient noise present in the room during the measure.

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