Realisation of “virtual” musical instruments: measurements of the Impulse Response of Violins using MLS technique

Angelo Farina*, Andreas Langhoff**, Lamberto Tronchin***

* Dipartimento di Ingegneria Industriale, University of Parma, Italy
** Scuola Internazionale di Liuteria, Cremona, Italy, and Institute for System Engineering and Informatics, E.C. J.R.C., Ispra (VA), Italy
*** Istituto di Fisica Tecnica, Facoltà di Ingegneria, University of Bologna, Italy

Abstract

In this paper the realisation of “virtual” musical instruments is analysed, in which the instruments are treated as linear systems, characterised by their impulse response.
Various measurements techniques of the impulse response have been tested, employing different transducers and numerical analysis. The better one resulted the direct Maximum Length Sequence excitation signal, applied to the bridge of the violin by a current-to-force transducer. The acoustic pressure radiated from the violin’s body is then sampled by a microphone located in an anechoic chamber, and the Impulse Response is obtained by cross-correlation of the acoustic signal with the excitation signal.
The impulse responses measured with the technique presented in this paper contain all the timbric and reverberant characteristics of the violin. Langhoff already developed a rating technique that make it possible to extract objective informations about the timbric quality of violins [1]: the scope of the present work is to evaluate also the reverberant quality of the instruments.
Furthermore, the measurement of the impulse response enables the creation of a “virtual” violin, which is a numerical filter that applied by convolution to an “anechoic” signal add to it all its information. The techniques to obtain such samples of “anechoic” input signals are also discussed here.

1. Introduction

The violins are not mechano-acoustic linear transducers, but their non linear characteristics are due to the interaction between string and bow, and to the vibration of strings. However, the most important difference in timbric perception among ancient and modern violins depends on the body of the violins that behaves as a source of radiation, in dependence of the characteristics of the wood sound chest of each instrument.

The problem was analysed assuming the system included from the bridge of the violin (in which the strings are in contact with the body) to the sound field in an anechoic environment: this system is surely linear, as a consequence of the small displacements of the structure. The input signal, coming from the strings to the bridge, must be measured separately with an appropriate technique.
The impulse response can be obtained directly (excitation of the violin on the bridge with a known **force F** and measurements of the impulse response as **sound pressure p**) and inversely, using the reciprocity technique (excitation of the sound field with a known **volume velocity Q** generated by a loudspeaker, and measurements of the impulse response as **velocity u** of vibration of the bridge). These two different techniques give the same results, provided that the requirements dictated by Fahy in its mecano-acoustics reciprocity theory are met [2]: these requirements demand for a point, omnidirectional sound source located exactly in the same place where the microphone is located, and a velocity transducer located exactly in the same position of the structure where the force load is applied.

$$\mathrm{FRF} = \frac{p(\omega)}{F(\omega)} = \frac{u(\omega)}{Q(\omega)} \tag{1}$$

Both techniques were employed in the present work, but actually the direct method seems capable of superior performances than the reciprocal one, due to the better signal-to-noise ratio and to the greater linearity of the transducers.
The second problem is to properly measure the input signal (force), to make it available the “anechoic” input signal to be used for convolution. The placement of force transducers (piezoelectric load cells) between the strings and the bridge during the violinist’s performance resulted unfeasible. However, a velocity transducer can be placed over the bridge, causing only a limited disturbance to the player: this is made with a phonograph needle, supported by a flexible arm. The sound track coming from the velocity transducer is not, however, what is required: in fact this is the response of the mechanical system, excited in a point with a very complex mobility function (velocity versus force).
Two indirect techniques were developed to reconstruct the “anechoic” signal: the first is based on an inverse filtering of the velocity track, the second of the acoustic signal recorded by the microphone in the anechoic chamber. In the first case the inverse filter is the inverse of the mobility function, that is the mechanical impedance of the excitation point (force vs. velocity). In the second case, the inverse filter is the inverse of the mechano-acoustical radiation impulse response (acoustic pressure vs. force). The inversion of these complex functions is not easy, as they are mixed-phase type. The inversion of long, mixed-phase responses is still an unresolved mathematical problem [3], so approximate solutions have to be used. In this case the better performances were obtained with the Neely and Allen [4] technique, that invert only the minimum phase component of the impulse response, obtained taking the modulus of its Fourier transform and forcing the phase to zero. This perfectly removes the timbric character of the violin on which the music sample was played, but still leaves in the signal an “all-pass” reverberation that can be heard.

2. Impulse Response measurement techniques

The measurements of the mechano-acoustic impulse responses have been obtained using a MLS (Maximum Length Sequence) signal, generated by PC fitted with an A/D board equipped with an hardware MLS generator and a software for deconvolution of the response.

For direct measurements this signal, properly amplified, was sent to a Dunnwald-like copper wire force transducer [5]: the force exherted over the bridge is proportional to the current passing through the wire, as it is located in a strong, permanent magnetic field.
The violin was placed in the anechoic chamber of the Cremona's Violin Making School, that is already fitted with proper supports, microphones and preamplifiers. The signal coming from the preamplifier is sampled by the A/D board, and cross-correlated with the original MLS signal to obtain the impulse response directly in the time domain, thanks to the Alrutz fast deconvolution algorithm [6]. Fig. 1 shows a schematic diagram of this direct measurement technique.

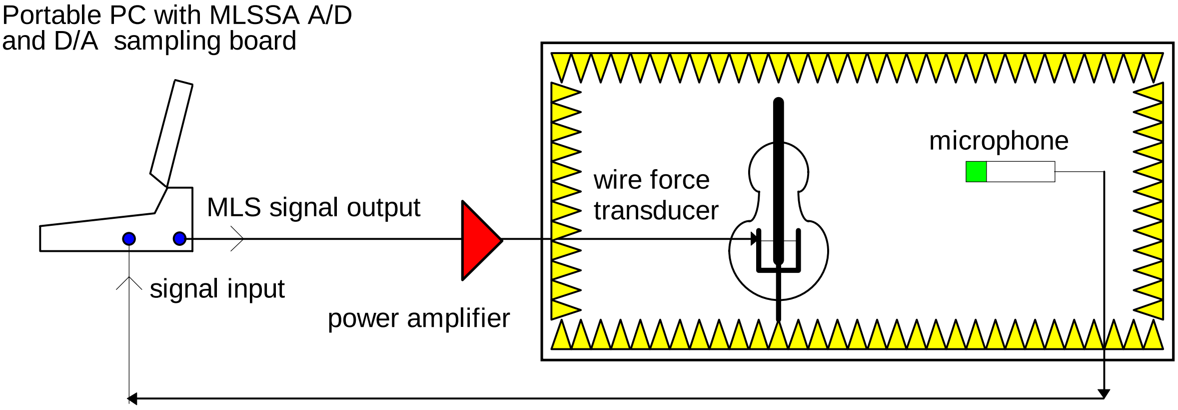

Fig. 1 - schematic measuring system for direct method

Also a reciprocal scheme was used, as shown in fig. 2. In this case, the MLS signal is fed to a loudspeaker, placed in the anechoic chamber approximately in the same position previously occupied by the microphone. The velocity response of the violin's bridge is detected by a phonograph needle, whose electric output is cross-correlated with the original MLS signal.

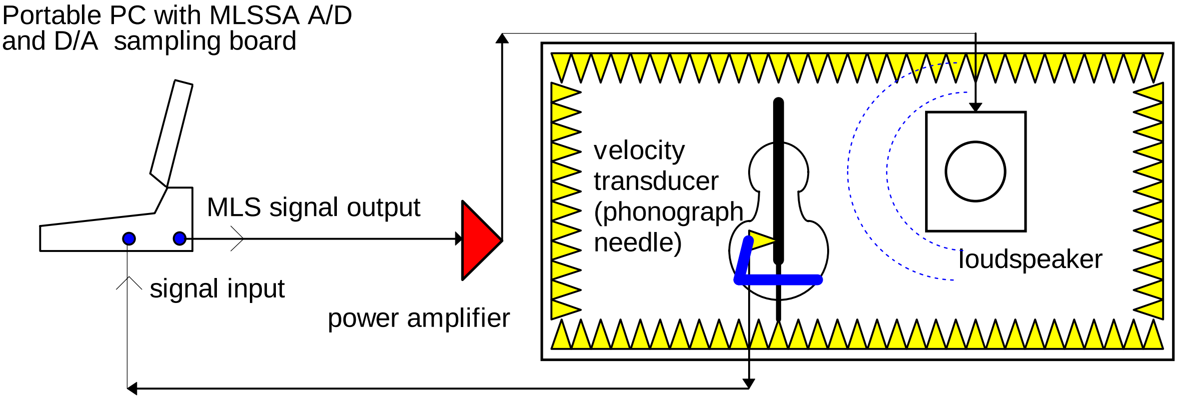

Fig. 2 - schematic measuring system for reciprocal method

Fig. 3 shows a typical measurement result by the direct technique, while fig. 4 shows the result of a reciprocal measurement on the same violin. Both the time domain and frequency domain representations are shown. It is evident that the results are not equal: this is certainly due to noise contamination problems evident in the reciprocal measurement. The authors think that this "noise" is not actually acoustic noise present in the environment (the anechoic chamber has a background noise lower than 20 dB(A)), but it is a mathematical artefact caused by non-linear distortion in the transducer chain, mainly in the loudspeaker and in the phonograph needle pickup [6].

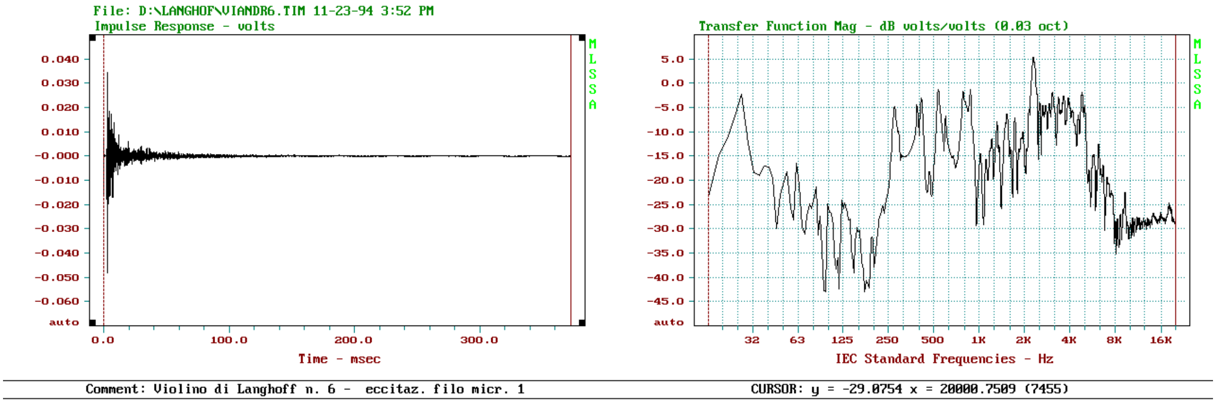

Fig. 3 - Impulse response and Frequency Response Magnitude - Direct method

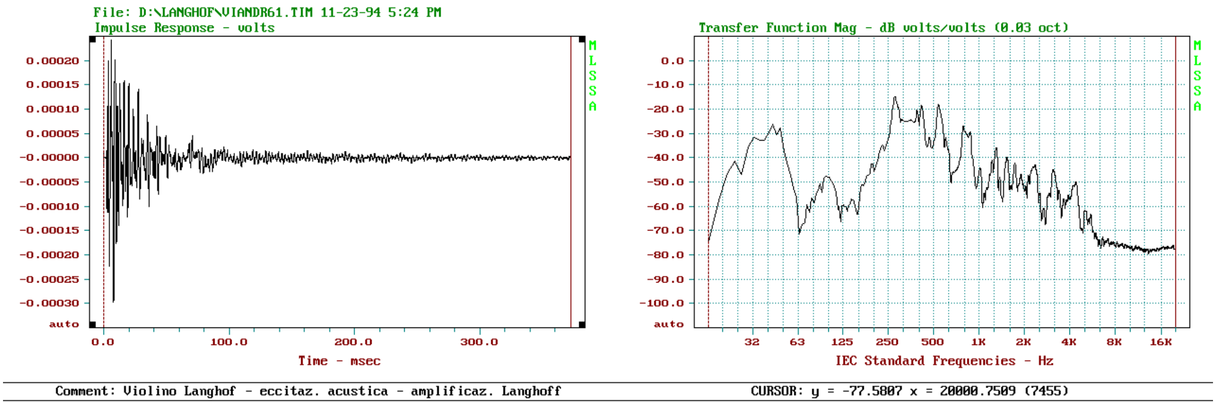

Fig. 4 - Impulse response and Frequency Response Magnitude - Reciprocal method

For this reason in the following only the direct-type Impulse Response measurements are considered.

Note also that other measuring systems, as FFT 2-channel analysers or similar, were not used in this work, although available at the Cremona's laboratory. This is due to the limited length of FFT analysis available with these instruments (actually 2048 points), which is not enough for a complete characterisation of a violin impulse response, as it requires usually at least 16384 point at a sampling frequency of 44.1 kHz.

3. "Anechoic" recording of input force signals

To compare the responses of different violins, an anechoic input signal is required, which is then convoluted with the impulse response of each violin, producing a "filtered" signal containing the whole characterisation of that particular instrument. Then a pair comparison technique can be used to subjectively assess the perceptible differences between the responses of different violins, as reported in a separate paper [7].

The obvious solution consists in measuring the force applied from the vibrating chords to the bridge, during a musical performance. However this is not an easy task: miniaturised load cells have to be inserted between the chords and the bridge, and their mass and stiffness are anyway too large to avoid any change in the dynamic response of the instrument. For this reason, it was chosen an indirect technique, in which the input force signal is reconstructed by inverse filtering of response signals. Actually we have measured two kinds of response signals during musical performances: the velocity of the top of the bridge (by the phonograph needle

pickup) and the acoustic pressure inside the anechoic chamber. The two tracks were simultaneously digitally recorded on a 2-channel DAT machine and transferred to .WAV format through a low-cost 16 bit audio board installed in the PC of the Cremona's laboratory.
In principle, the input force can be recovered by both these response signals (pressure or velocity), provided that the input/output transfer function of the system is accurately measured, and that it is possible to create an inverse filter to deconvolve out this transfer function from the response signal.
The transfer function for the acoustic pressure signal is the impulse response already measured as explained in the previous paragraph; the velocity vs. force transfer function is the mechanical mobility of the bridge, and this can be measured exciting the bridge with the Dunnwald force transducer and measuring the velocity output by the phonograph needle. Operating in this way has the advantage that, during the deconvolution process, the response of the transducer (the phonograph pickup) is filtered out together with the response function of the particular violin over which the music was played.
In theory these deconvolution techniques recover an "anechoic" input signal which has lost any colouring coming from the particular violin employed, and this signal can be used for convolution with any other violin's impulse response. Obviously this does not happens in the reality, as the performer changes its playing depending on the sonority of the instrument used, as he tries to correct for defects of it, or is anyway conditioned from some particularities of the violin (not necessarily acoustic, but for example related to the instrument embracement, to its tactile feeling, to the vibrational feedback through the shoulder and the chin). However, a performance on a medium-quality violin, with a not particularly evident character can produce a reasonably "universal" recovered input signal, suitable for further convolution with different violins. The employment of the same input signal on different violins is like to produce more evident acoustic differences than separate performances of the same music piece over these instruments, as no "compensation" is made by the performer.

4. Inversion of mixed-phase impulse responses

The latter question that remains open is how to invert long, mixed phase impulse responses, producing inverse filters that are causal, stable and finite length. The question was addressed in the last years by many authors, but the most efficient results are those of Mourjopoulos [8,9]. He proposes two general techniques: the minimum/maximum phase decomposition with separate inversion, and the least squares approximation.

4.1 Minimum and maximum phase signal decomposition

Following the first technique, the original impulse response is first decomposed in two components: a minimum phase one, containing all the poles which fall inside the unit circle on the Z-plane, and a maximum phase component, containing all the poles which fall outside the unit circle (it is assumed that no pole falls exactly over the unit circle).
The decomposition of a mixed-phase impulse response in the minimum and maximum phase component is not easy. It was tempted both by homomorphic decomposition [8] and by complex cepstral separation [9], but in general the results are poor.
Once the components are separated, the minimum phase component can be inverted directly, because simply taking the IFFT of the reciprocal of its FFT transform yields a finite, stable and causal inverse impulse response. The same approach is unsuccessful for the maximum-phase component, as its inverse is unstable (the response is not decaying to zero with increasing time, but the values get larger and larger..). However, if the maximum-phase component is time-reversed, then its response becomes stable, but infinite and acausal; after

the inversion, the inverse impulse response is time-reversed again. If the time window is long enough, the truncation of the inverse response does not cause any appreciable error; furthermore, the acausality can be corrected adding a simple time delay, that cause no practical problem to not-real-time processing. The inverse of the minimum and maximum phase component are then convolved, producing the final approximate inverse filter.

4.2 Least squares technique

The second technique set up a classic least squares problem, with an unknown inverse impulse response (made of N unknowns) that, convoluted with the original impulse response, has to approximate a delayed Dirac's delta function. Imposing the minimisation of the sum of the squared differences between the convoluted result and the wanted delta function, a set of N linear equations is formed:

$$\left[R(i,j)\right]\cdot\left\{h_{inv}(i)\right\}=\left\{h(d-i)\right\} \tag{2}$$

in which each row of the matrix [R] is simply a sample of N points taken from the autocorrelation function of the original response h(N), starting at time 0 for the first row and one sample left for each subsequent row (-1 for the second row, and so on); the column of known terms is a time-reversed, delayed copy of the original impulse response h(N) (d is the delay in samples, usually taken around N/2).
The solution of the system of eq. 2 is possible only for a limited length N of the impulse response, due to storage space limitations. These limits actually prohibit the inversion of impulse responses of the length encountered in the present work, so this method, that is usually judged the optimal one, was not applied in this case.

4.3 Approximate inversion of the zero-phase component

On the other hand, the decomposition of the impulse response in minimum and maximum phase components resulted difficult, as the phase response of the system exhibits many wraparounds, and it is difficult to unwrap correctly these jumps.
So none of the two known methods of complete inversion resulted applicable in this case. Nevertheless, the Neely and Allen [4] approximate technique resulted a viable method to remove at least the timbric coloration caused by the particular violin used. Following this technique, the original impulse response is decomposed in a zero-phase component, obtained taking the modulus of the frequency response, and an all-pass component, containing the phase of the frequency response with unitary modulus. The first is obviously a minimum phase component, and it can be inverted directly by taking the inverse Fourier transform of its reciprocal. The second is again mixed-phase, and cannot easily be inverted: it carries just "reverberation" information, not any timbric coloration, and it is left not inverted.
Applying the zero-phase inverse filter to the measured response signals, the timbric coloration of the violin is removed, but the "reverberation" is still present. When this "anechoic" signal is convoluted again with a violin impulse response, the spectral coloration is restored properly, but the reverberation is a little too big. However this effect is noticeable only comparing signals directly recorded in an acoustically-anechoic space with violin convolutions. If the comparison is repeated in a non-anechoic acoustic space (i.e. in a room), the directly recorded acoustic signal becomes indistinguishable from the original input signal convoluted with both the violin's impulse response and the room's impulse response; this because the room's reverberation is usually larger than the violin's reverberation (1s against 0.2 s), and it masks completely the latter.

5. Results

The impulse responses of three violins and of one viola were measured with the direct technique presented in paragraph 2. The three violins were identified by the name of the builder:

- Calcanius
- Klotz
- Langhoff

The viola was introduced only to have a very different instrument, making audible to everyone the different timbric coloration.

Fig. 5 reports the time-frequency responses of the 4 instruments. It can be observed that the viola is noticeably different, whilst the differences between the three violins are not so clearly evident from these waterfall representations.

Also the mobility function of these 4 instruments was measured. However, musical trials were conducted only over the three violins, and furthermore the velocity music tracks recorded with the phonograph needle were noisier than the acoustic pressure tracks. So the inversion of the mechano-acoustic impulse response function was preferred to the inversion of the mechanical input mobility.

Two music samples, played on the Langhoff's violin, were filtered with its inverse impulse response to produce two "anechoic" input samples. As a verification, the convolution of these input samples with the direct impulse response of the Langhoff's violin produced results almost indistinguishable from the original anechoic acoustic pressure recordings. On the other hand, convoluting the same input signals with the impulse response of the other three instruments produced noticeably different results, as it is reported in a separate paper [7].

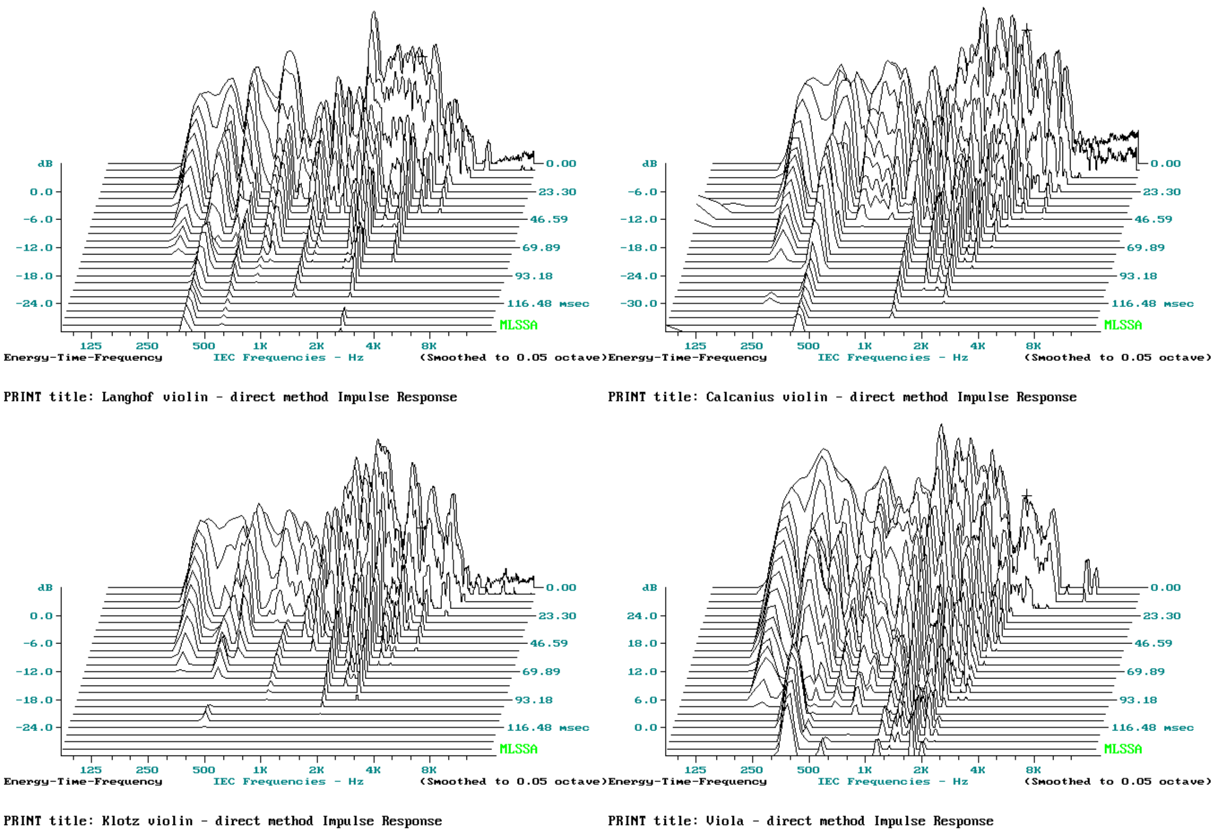

Fig. 5 - Energy-Time-Frequency responses of the four instruments

6. Conclusions

The measuring techniques implemented were able to accurately measure the mechano-acoustical transfer function of different violins, represented by the time-domain impulse response. The MLS excitation system circumvented the limitations previously encountered, caused by limited time window length, poor noise rejection, limited sampling rate and long measurement time. Furthermore this technique avoids the use of large excitation amplitudes, that possibly cause non-linear distortions in the vibroacoustic system.
The approximate inversion of these impulse responses made it possible to reconstruct the force input signal applied from the strings to the bridge during a music playing, starting from an anechoic acoustic recording. This “anechoic” input force signal can then be used as a starting point for producing music samples played on “virtual” violins, by the well known fast convolution process [10]. These music samples can then be used for subjective comparisons between different violins, without the bias caused by the player’s reaction to different instruments. By numerical manipulation of the impulse responses, the subjective effect of removing/adding spectral components or damping the reverberant character of an instrument can easily be assessed.
The research is going on, searching a different non-approximate inversion technique of the impulse response that avoids the limitations that actually affect any known scheme for the inversion of long mixed-phase impulse responses.

References

[1] A. Langhoff - “Measurement of acoustic violin spectra and their interpretation using a 3D representation” - Acustica vol. 80, pp. 505-515 (1994).
[2] F. Fahy - “Sound and structural vibration : radiation, transmission and response” - London, Academic Press Limited, 1985.
[3] J.N. Mourjopoulos - “Digital Equalization of room acoustics”- J.A.E.S. vol. 42, No. 11, pp. 884-900, November 1994.
[4] S.T. Nelly and J.B. Allen - “Invertibility of a Room Impulse Response”, J.A.S.A. vol. 66, pp. 165-169 (1979).
[5] H. Dunnwald - “Ein Verfahren zur objektiven Bestimmung der Klangqualitat von Violinen” - Acustica vol. 58, pp.162 (1985).
[6] D. Rife and J. Vanderkooy - “Transfer function measurement with Maximum-Length Sequences” - J.A.E.S. vil. 37. no. 6, pp.419-443, June 1989.
[7] A. Farina, A. Langhoff, L. Tronchin - “Subjective comparison of Virtual Violins obtained by convolution” - Proc. of CIARM 95, Ferrara, May 1985.
[8] J. N. Mourjopoulos - “The removal of reverberation from signals” - Ph.D. Dissertation, University of Southampton, UK (1984).
[9] P.M. Clarkson, J. Mourjopoulos, J.K. Hammond - “Spectral, Phase and Transient equalization for audio systems” - J.A.E.S. vol. 33, pp. 127-132, March 1985.
[10] A.Farina - “An example of adding spatial impression to recorded music: signal convolution with binaural impulse responses” - Proc. of “Acoustics and Recovery of Spaces for Music”, Ferrara (Italy) 27-28 Oct. 1993.