# Convolution Applied to Synthetic Original Music: Some Italian Theatres Examples

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## Abstract

*The convolution of musical pieces with room impulse responses allows the comparison of different halls with a short time delay and under the same conditions. This task can be sometimes difficult due to the restricted choice of proper input signals, which must be anechoic and well-suited to the psychoacoustic analysis to be done. In this paper, the usage of synthetic pieces is suggested and exemplified using an original piece, written to reveal some particular feature of the halls under study. Starting with the same musical piece, in a perfectly anechoic MIDI format, two different convolution algorithms have been applied using the binaural room impulses responses measured, with two different techniques, in three halls with very different acoustic characteristics. This allowed either the comparison of different algorithms on the same halls, either the comparison of the acoustics of different halls listening the same musical piece.*

## 1 Introduction

It is normally quite difficult to compare the acoustics of different halls, because the human memory for sound impressions is not long enough to get from one hall to another. Binaural recordings made in different halls make it possible, provided that the measured binaural room impulse responses are presented to the listeners after processing with the same musical piece [1]. In technical words, the binaural signals presented at the receiver ears *e(t)* must result from the convolution of the binaural room impulse responses *h(t)*, measured in the real halls, with the same musical signal *s(t)*:

$$\begin{bmatrix} e_{left}(t) \\ e_{right}(t) \end{bmatrix} = \begin{bmatrix} h_{left}(t) \\ h_{right}(t) \end{bmatrix} * s(t) \tag{1}$$

The source signal *s(t)* is monophonic and must be “dry” or “anechoic”, i.e. it must contain only the music as it would sound in a free space, without any reverberation which could contaminate the “natural” room reverberation, contained in *h(t)*. Therefore, the choice of a suitable *s(t)* is restricted to few pieces recorded in an anechoic room. An alternative approach is suggested in this paper: the usage of a synthetic musical piece which is perfectly anechoic in principle. The success of such an approach would give to the researchers much more freedom, allowing the usage of pieces purposely written to reveal some particular feature of the halls under study. The technique has been checked using an original musical piece.

## 2 The Anechoic Piece

The anechoic piece used in this work was taken from the lyric scene "La tempesta" (the tempest) by G. Fabbri (1992). The orchestra is composed by 3 flutes, 3 oboes, 3 clarinets, 3 bassoons, 4 horns, 3 trumpets, 3 trombones, tuba, first and second violins, cellos, basses, kettledrums and cymbals. The piece has been loaded into the computer using Cakewalk Apprentice [2], a sequencer running in the Windows environment. This program allows the utilisation of up to 256 different tracks on 16 channels, in the General MIDI format. The piece was then synthesised using the Wave Blaster board, which allows a 32 voices polyphony, omitting any kind of artificial reverberation. The resulting MIDI file has been transformed into a .WAV file acquiring it through another board, a Sound Blaster 16 Multi CD, using a sample rate of 44,1 kHz and a resolution of 16 bits.

## 3 The Room Impulse Responses

The binaural impulse responses (I.R.) were measured in three different halls, using two different techniques: blank shots and MLS [3]. The halls are:

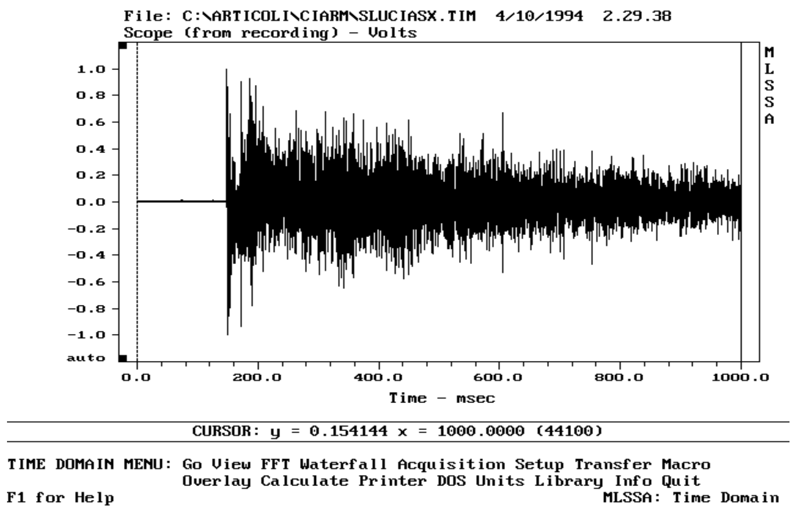
- the main hall of "S. Lucia", Bologna; it is an old dismissed church now used as "Aula Magna" of the University; the hall is aesthetically beautiful, but very large and reverberant, with a poor acoustics; the measurements were performed using the MLS technique (fig. 1);
- the theatre "La Scala", Milan; it is a world-wide famous theatre, with an excellent acoustics; the measurements were performed using blank pistol shots as input signal (fig. 2);
- the "Sala Europa" (Europe Hall), Bologna; it is a modern conference hall with a short reverberation time; the measurements were performed using the MLS technique (fig 3).

## 4 The Convolution Programs

The convolution of the original synthetic piece with the three binaural room impulse responses has been performed by means of software tools. This allowed the comparison of the relative advantages and drawbacks of different programs on the same hardware, i.e. on the same computer. In particular, earing at the final results one can get an impression of the "naturalness" of the output of different algorithms.
Both programs work in the frequency domain, where convolutions become products, i.e. the binaural ear signal is obtained as:

$$\begin{bmatrix} e_{left}(t) \\ e_{right}(t) \end{bmatrix} = F^{-1}\left\{ \begin{bmatrix} H_{left}(f) \\ H_{right}(f) \end{bmatrix} S(f) \right\} \qquad (2)$$

where $F^{-1}$ represent the inverse Fourier transform, *H(f)* is the Fourier transform of *h(t)* and *S(f)* is the Fourier transform of *s(t)*. This requires additional FFT/IFFT operations, but speeds up considerably the core of the computation.



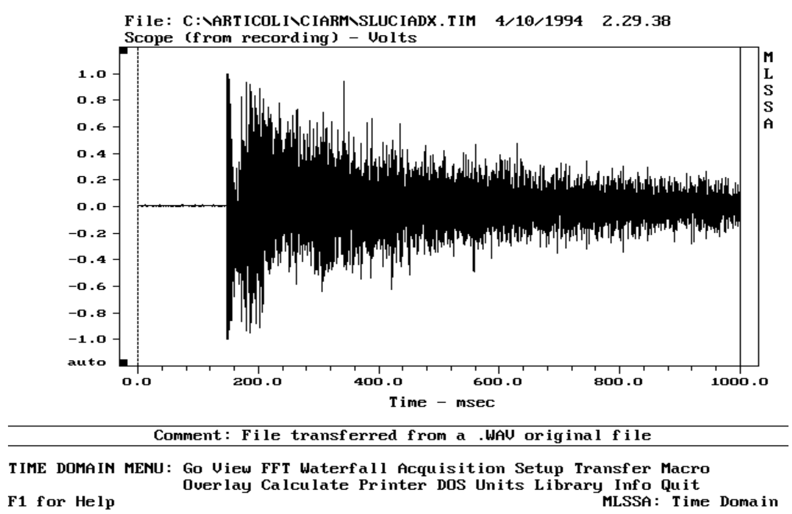




**Figure 1.** I.R.'s (left and right ear) measured in the main hall of "S.Lucia", Bologna.

### 4.1 The MATLAB program

The first program was written for the MATLAB environment [4]. The standard MATLAB routine for convolution has been discarded because, when the musical piece in input is only few seconds long, it works by iteration on the musical piece, Fourier-transforming the room impulse response at every iteration. On the other hand, by writing a suitable program, this Fourier transform can be done only once.

For the newly written program, the anechoic signal and the room impulse response (which can be monoaural or binaural) must be stored in .WAV files, with the same sample rate (in this case 44,1 kHz). The resolutions of the two files can also be different. In a first step, the program reads the room impulse response (if it is binaural the programs reads the left channel first), controls if a zero padding must be applied in order to have a number of points which is a power of two, and then performs a Fourier transform. Then, the convolution is realized dividing the anechoic signal in several blocks, zero padding each of them, Fourier-transforming each of these enlarged blocks, multiplying them by the Fourier transform of the room impulse response and antitransforming the results. Each block must be short enough to avoid the usage

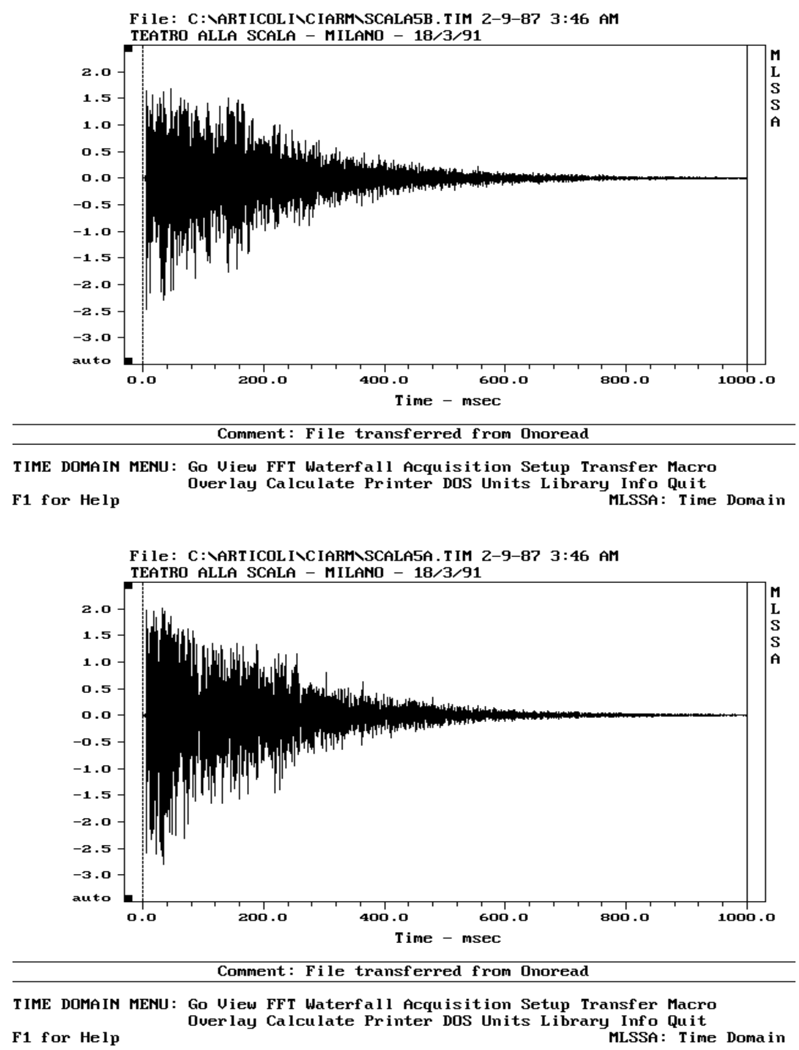



**Figure 2.** I.R.'s (left and right ear) measured in the theatre "La Scala", Milan.

of the virtual memory and long enough to avoid an excessive partitioning of the signal. If the room impulse response is binaural, the described loop is firstly performed on the left channel and then repeated on the right channel. Finally, the convolved signals are normalised with respect to the maximum value and stored in another .WAV file.

### 4.2 The Aurora system

The Aurora software system [5] uses input and output data files stored on the hard disk in standard .WAV format. The convolution is performed through the "select-save" algorithm [6], on impulse responses long up to 200·000 taps.

The actual Aurora version gives computing times, on a i486 DX2-66 PC, varying from 62 s for a .WAV mono input (anechoic) signal of 10 s, a sample rate of 44,1 kHz and a room impulse response of 16 kpoints, to 490 s for a .WAV mono input (anechoic) signal of 30 s, a sample rate of 44,1 kHz and a room impulse response of 180 kpoints [5]. The program takes 10-20 s to initialise, performing the FFT on the room impulse response, then the select-save loop begins. The I/O time due to disk access is always smaller than 10% of the total computation time.

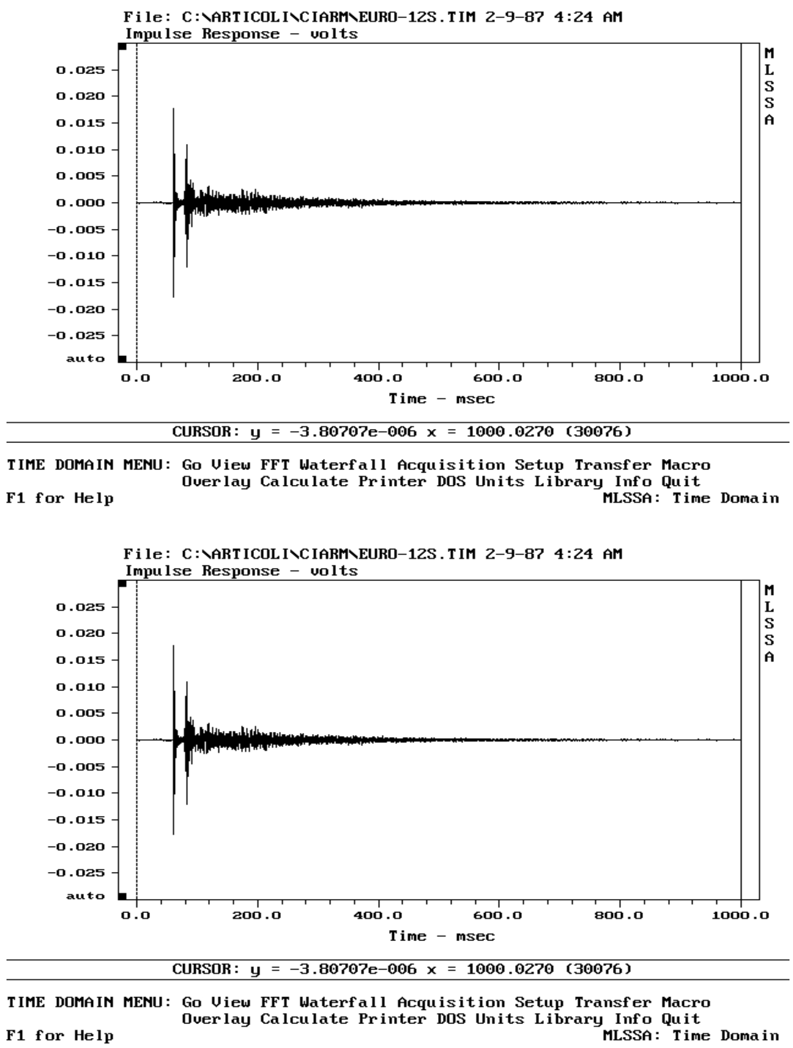



**Figure 3.** I.R.'s (left and right ear) measured in the "Sala Europa" auditorium, Bologna.

## 5 Conclusions

It has been shown in this work that synthetic musical pieces can be used to generate recordings for psychoacoustics tests, with results comparable to those originating from musical pieces recorded in anechoic rooms.
Different software tools can be used to perform the convolution process, giving quite good results on measured room impulse responses, and giving to the user much more control on the details of the process than specialised hardware devices. The Aurora system was found to be fast and reliable, while programming a general-purpose environment, like MATLAB, gives to the user the freedom of changing the way the convolution is performed.
The described procedure allow to distinguish the particular acoustics of each hall by listening to the results of the convolutions: the “S.Lucia” impulse response adds too much reverberation to the anechoic piece, diminishing the clarity; the “La Scala” impulse response gives an output clear and rich in reverberation and the “Sala Europa” impulse response gives a result still too dry.

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