Software Toolbox for Multichannel Sound Reproduction

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Abstract

This paper describes a versatile software toolbox, which has been developed for researching, teaching and developing in the field of multichannel sound signal processing. The software system runs on a PC and consists on 5 modules covering the main stages and aspects of multichannel sound reproduction using loudspeakers. A number of new and efficient algorithms have been specially implemented for this software.

1 Introduction

The software system is composed by a set of modules designed for researching, teaching and developing in the field of multichannel sound.

In binaural systems, the reproduction of 3D sound through loudspeakers adds more difficulties to the traditional method using headphones: cross-talk effect and room transfer functions between speakers and listener ears have to be taken into account.

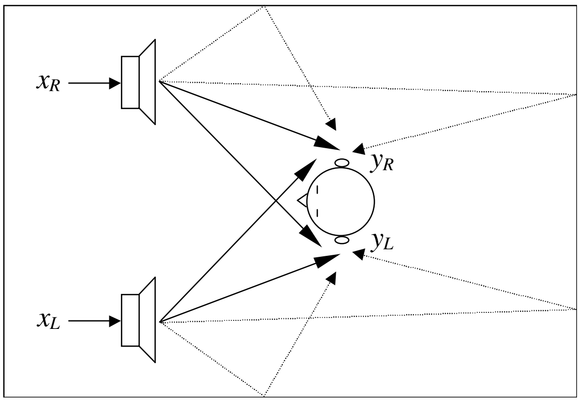

Figure 1. Cross-talk effect and wall reflections in binaural sound reproduction.

The theoretical way of removing this distortion is to perform the inverse of the Room Impulse Response between sources and receivers [4]. Thus, there is an unfulfilled need for a multichannel inverse-filtering method for sound reproduction using loudspeakers.

In general, the computational complexity of the multichannel room acoustics inverse filtering techniques increases when the number of sources (loudspeakers) grows. On the other way, the mathematical solution accuracy improves [7].

Figure 2 shows the different parts that compose a general multichannel sound reproduction system using loudspeakers. Let's assume that it wished to reproduce K recorded signals (two for a stereo system) at L points in the listening space using M loudspeakers. The matrix C(z) in *Figure 2* characterizes the electroacoustic transmission paths. The matrix H(z) of digital (generally inverse) filters operates on the recorded signals $x_k(n)$ prior to their transmission via the loudspeakers for obtaining the desired signals $d_k(n)$. Signals $e_l(n)$, commonly named error signals, measure the resemblance between the desired and the reproduced signals.

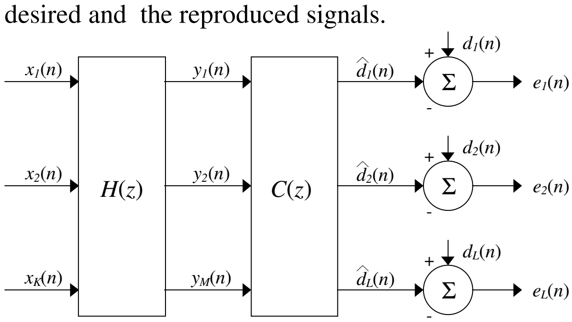

Figure 2. Multichannel sound reproduction system using loudspeakers. Filter bank and acoustic paths.

Using the block diagram of *Figure 2*, a number of papers related to cross-talk cancellation and invertibility of room impulse responses have been recently published [6]-[9], but there exist few publications reporting practical measures and implementations. Also a versatile system has been shown to be needed in practice since the number of loudspeakers, recorded signals, points in the listening space, the listening space or even the kind of desired and recorded signals can vary between experiments.

Also, as it can be seen in *Figure* 2, the implementation of a bank of digital filters H(z) is needed in practical multichannel reproduction systems. For general purposes, a number of algorithms for designing such a bank of filters are needed as well.

Several methods proposed in the literature have been integrated in this software system. Computationally efficient signal processing and matrix analysis methods have been employed, having special care during their implementation, to allow them to work on a personal computer.

2 System architecture

The software tool has been programmed to carry out test and measurements of multichannel reproduction systems. The software runs over Windows 95 OS (98 or NT are also accepted) and a standard Personal Computer.

The system has been developed employing standard operating system sound drivers allowing to use any compatible sound card, although a high-quality multichannel sound card is recommended for scientific or professional applications. Actually the system is running using a Gina card from Event, that has 2 analog inputs, 8 analog outputs, 1 digital SPDIF input and 1 digital SPDIF output.

Figure 3 shows how the different parts of the system interact. The application software drives the sound card through the standard OS and sound card drivers. Therefore, the application software can be used with any sound card. The number of loudspeakers and microphones that can be commanded by the application software depends on the hardware capabilities and the available computational resources.

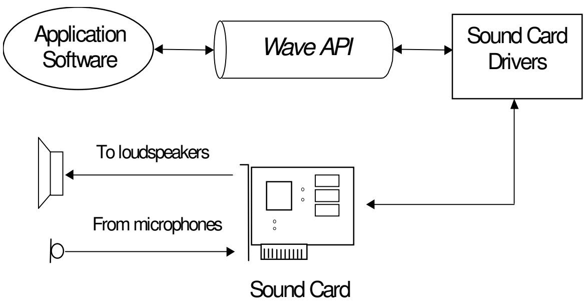

Figure 3. System architecture.

3 Software modules description

Each of the software modules can be accessed from the main window. This windows also allows to configure the working environment: sampling frequency selection, kind of card and number of input output channels.

The software system consists on the following 5 modules: multichannel real-time digital filtering, acoustic impulse response measurement using Maximum Length or MLS sequences, computation of multichannel inverse filter banks, evaluation of multichannel reproduction systems using MLS and placement of virtual sources using HRTF database.

3.1 Multichannel real-time digital filtering

The use of a fast convolution method (overlap-save) allows us to run a bank of digital filters between sound sources and loudspeakers.

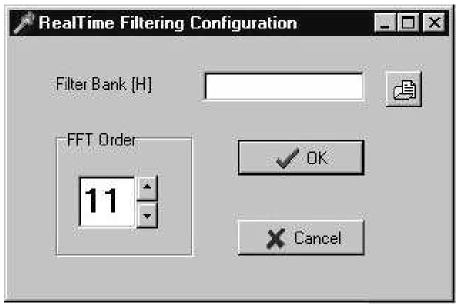

Figure 4. Real-Time filtering configuration

Figure 5 shows data entry window. Files containing the bank of FIR filters and order of the FFT used in fast convolution must be entered. In practice, there is no limit for the number of filters and their length, in number of coefficients, but the computational resources limit: processor speed or RAM size.

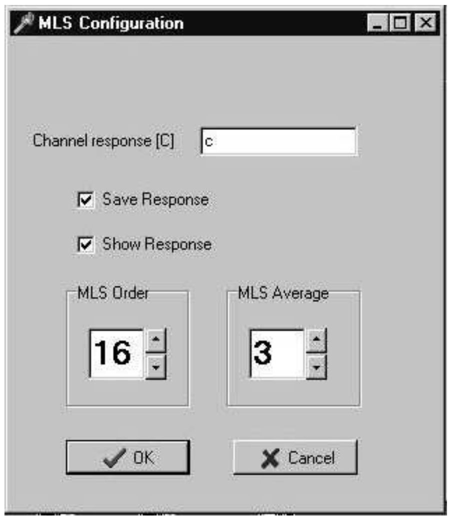

Figure 5. MLS impulse response measurement config.

3.2 Acoustic impulse response measurements using MLS

A versatile MLS system is provided for measuring acoustic responses between speakers and microphones (listening points), [1,2,3,11]. Length of the MLS sequence, sampling frequency and number of averages are easily user-definable as it can be seen in *Figure 5*.

3.3 Computation of the multichannel inverse filter bank

In order to cancel cross-talk between speakers, reduce the room reverberation or, generally speaking, obtain a desired response, it is necessary to pass the recorded sound signals through a specially designed filter bank prior to feed the loudspeakers.

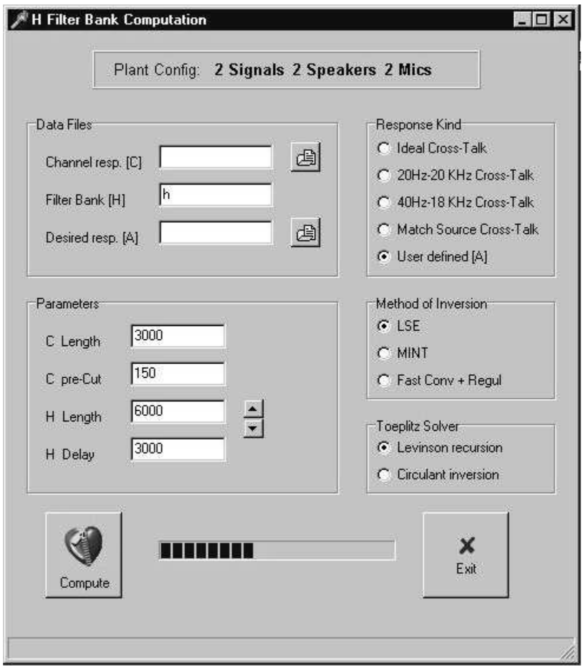

Figure 6. Computation of filter bank configuration.

Figure 6 shows the configuration window for the bank of filter design. This window is composed of five different zones. The previously measured plant response bank of filters is requested by the data file zone. Other section allows us to introduce the name of the destination file of the calculated filter bank. The desired signals are also requested, they are generated by the software system passing the recorded signals through the bank of filters A. Bank A is user supplied and it will be different depending on the application: cross talk cancellation, room acoustics inverse filters design, multipoint equalization,... On the other hand, user can choose a previously defined desired filter bank, A, from the kind of response selection zone.

Length of filters and responses are also requested and user selected. The next parameters can be defined: length and transport delay of the plant response (C length and C precut in *Figure 6*); and length and modeling delay of the computed bank of filters (H length and H delay in *Figure 6*).

Three main methods of computation of the inverse filter bank have been implemented and they can be selected in the zone called "Methods of Inversion" in *Figure 6*. Two methods are available in time domain: a least squares optimization procedure, named LSE method [8], and a technique for the perfect inversion of multichannel systems, called MINT in the multichannel audio processing literature [5]. An additional method is supplied that performs a least squares optimization in frequency domain, called Fast deconvolution using regularization [10].

Finally, the kind of matrix inversion method can be selected. Both time domain methods need at some step to compute the inversion of a quite large block Toeplitz matrix, symmetric or non symmetric, therefore two computationally very efficient algorithms have been implemented to carry out this, other way cumbersome, calculation [12].

3.4 Evaluation of multichannel reproduction systems using MLS

This module is necessary in order to measure the goodness of a computed filter bank for a given sound reproduction system and acoustic environment. It has to be noted that usually least squares optimization methods have been used to obtain the filter bank response, A, but there is not clear how this solution is performing in practice. Even if the solution were mathematically exact, by example a perfect inversion, their behavior has to be tested in practice since some physical limitations in the practical system could have not been taken into account by the designer. Finally, this measure is also needed in order to check the obtained filter bank robustness against certain changes of the acoustic environment.

In this module, MLS sequences are pre-filtered through the previously computed filter bank and then used to measure the multichannel system. As in the measure of acoustic responses using MLS case, the number of measure averages and the length of the MLS sequence are chosen by the user. However, the computed or selected filter bank, A, have to be supplied in this case.

3.5 Placement of virtual sources using HRTF database

This module is currently under development. It provides the desired responses between virtual sources and listening points by means of a selectable database of HRTF's. These responses together with actual channel responses between loudspeakers and listening points are then used by module 3 to compute the filter bank. The suitable combination of the desired responses with a cross-talk cancellation design can place virtual sources at any space position.

4 Algorithms employed

Fast Hadamard Transform [2], has been used for acoustic impulse response measurement using MLS, so the number of operations needed for computing sequence correlation's decreases.
Fast convolution using short time Fourier analysis and synthesis is employed for real-time filtering. Overlap-save method is used for window treatment of samples. Time and frequency domain methods have been employed to calculate the bank of inverse filters. The multichannel correlation matrix are efficiently calculated and stored. Fast and super-fast Toeplitz solvers efficiently compute the desired filter bank in time domain [12]. Frequency domain method comprises fast deconvolution using FFT and regularization [10].

5 Conclusions

A very useful and versatile software toolbox for multichannel audio reproduction has been presented. It runs in a PC with affordable sound cards and uses a fashioned and easy to use windows interface.

As a difference from previous works [6] [7] [9], this software toolbox comprises the main stages needed for developing, computing and testing multichannel reproduction audio systems using loudspeakers.

Advanced computation algorithms and procedures have been implemented in order to compare their performances and achieve the best results for a given acoustic environment. This software has been already used in real experiments [13].

6 Acknowledgments

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