RADIATION CONTROL ON A MULTI-LOUDSPEAKER DEVICE

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Abstract

The control of directivity represents a new stake in the reproduction of sound sources by an electroacoustic device. This paper first reminds the principle of a general method for reproducing the acoustical field radiated by a source, using a multichannel device controlled by digital signal processing. It approximates a given directivity pattern by linear combination of the fields radiated by a set of loudspeakers laid out on the faces of a polyhedron. The present study, derived from previous work, is devoted to the bandwidth improvement and to the optimization of the directivity control. It rests on the preliminary constitution of a set of basic directivities, namely the first spherical harmonics, from which it is possible to tune the cardioicity and the three-dimensional orientation of the synthesized directivity pattern. For the simulation of musical instruments, it is important to maintain independent control on the spatial and spectral characteristics and to put focus on a faithful reproduction of the power spectrum. For that purpose, an analysis/synthesis method of the power spectrum is used to provide the automatic design of a correction filter to be applied on near field recording of the instrument before feeding the source. The results are discussed considering room acoustics criteria and examples of musical instruments.

INTRODUCTION

The state of the art in loudspeakers technology allows accurate control on timbre and intensity criteria. Moreover, if needed, signal processing affords easy mean to correct their spectral and temporal response. On the contrary, reproducing a given directivity remains difficult because most systems are close to a baffled piston model and, therefore, their radiation cannot be significantly modified. However, when the emission occurs in a closed, or semi-closed space, directivity plays an important role for the perception since it affects the time and spectral distribution of the sound reaching the listeners' ears.

Radiation control requires then designing a new generation of transducers involving a multi-driver architecture. Several works have yet produced consistent results in this domain such as creating a sound beam adjustable in frequency and width [1] with 1D and 2D transducer arrays or super directive devices using ultrasonic transducers [2]. However, the aim of such development is generally to focus the sound information on a limited audience area, and to minimize the room effect. In the present paper, we propose a slightly different approach for the intent is here to reproduce or, at least, to control the way sounds will interact with the room.

REPRODUCTION METHOD

Theoretical basis. Let's consider an acoustical source T, within a domain D, radiating the pressure $T(\vec{r},\omega)$ in a domain V outside D (see fig. 1 for notations). Propagation laws tell us that this radiation is entirely determined by the pressure distribution (magnitude and phase) on an arbitrary surface S around D.

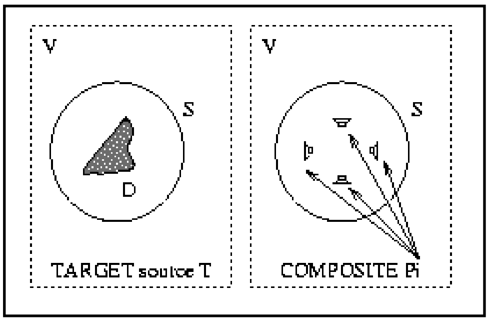

Figure 1: *radiation reproduction*

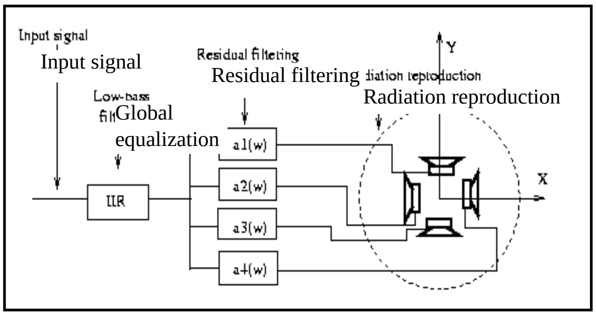

Figure 2: *DSP implementation*

Let's try now to synthesize the directivity of the target source T with a model. From the above principle, we can state that if the latter reproduces exactly the pressure on a given surrounding surface, both target and model radiation will be equal outside this surface. Actually, the model is a composite source built by combination of N elementary sources (loudspeakers or transducers), each producing the field $P_i(\vec{r},\omega)$. Then, the equality condition on pressure will lead to realize a decomposition of T on $\{P_i\}$, according to the equation:

$$\sum_{i=1}^{N} a_i(\omega)\cdot P_i(\vec{r}_0,\omega) = T(\vec{r}_0,\omega) \qquad ,\forall \vec{r}_0 \in S \tag{1}$$

where $\{a_i\}$ are complex coefficients depending on frequency ω.

Optimization procedure. However, as $\{P_i\}$ is not, at first glance, a base of the free-field radiation problem, existence and uniqueness of $\{a_i\}$ are not ensured at all. Thus, the method only leads to an approximation of T that is optimized to get the 'best' reproduction with respect to a given criterion. For this purpose, we choose to read equation (1) as a distance between target and model so as its minimization, within a fitted algebraic space (Hilbert), should lead to N equations/unknowns system giving, for each ω, the values of $\{a_i\}$:

$$(T-\sum_{j=1}^{N} a_j \langle P_j | P_i \rangle) = 0 \qquad \forall i = 1\ldots N \tag{2}$$

where $\langle\cdot|\cdot\rangle$ is a hermitian form of the functional space defined by $\langle f|g\rangle = \oint_S f(\vec{r}_0)\overline{g(\vec{r}_0)}\cdot dS$

Composite characterization. Obviously, the method involves a preliminary modeling or experimental step to define both, the radiation behavior of the elementary sources $\{P_i\}$, and the target directivity pattern T to be reproduced. Theoretically, they are performed by the same protocol: defining a spatial sampling of a surrounding sphere where the response of the source is measured. In practice, it may be convenient to reduce the experimental cost, taking advantage of axial and structural symmetries of the composite source (regular polyhedron, identical transducers). The measurement of the composite may be then limited to a bi-dimensional spatial sampling with a single active transducer.

Prototype development. This procedure was first applied to a dodecahedron [3][4] holding, for technological restrictions, only four independent subsets of loudspeakers. The $\{a_i\}$ coefficients resulting from the optimization process represent a set of filters applied to the input signal before supplying the composite source. From a signal processing point of view, their implementation can be split into two stages. The first is a common header module, controlling a global magnitude equalization of the composite source, and can be implemented with a minimum phase IIR filter. The residual filters carry important phase relations between the different transducers and are preferably implemented with FFT-convolution (fig. 2).

Results. To confirm the merits of the method, a characterization of the synthesized directivities was made, using the same measurement protocol. As for the targets, theoretical directivities were chosen among the first spherical harmonics and some typical combinations of them. In parallel, simulation software was developed in order to predict the behavior of the filtered composite source. When comparing measurements and prediction, we could notice that the error was quite negligible: 0.3 dB on average, for several target. In others respects, although no psycho-acoustical tests were fully undertaken, the results of this first study were also encouraging in term of perception. Although being limited in a narrow frequency range (200 – 2000Hz), the spatial rendering of the radiation reproduction appears to be relevant in many cases. These intermediate conclusions allow us henceforth to rely on numerical simulations for studying next evolutions of the prototype.

INTERPOLATION WITHIN A SUBSET OF DIRECTIVITIES

New implementation. The research being carried out in the context of musical application, i.e. stage performance mixing real and "virtual" instruments, the development is conducted with a slightly modified approach. The constraints on the accuracy of directivity reproduction are fallen to the benefit of other requirements such as an enlarged bandwidth, real-time processing and versatility of the directivity control. For this purpose a new processing architecture is designed. A subset of elementary directivity patterns is permanently synthesized, namely the first order spherical harmonics: the monopole and the dipoles in the three canonical directions of the cube (X, Y, Z).

Although, these first harmonics could be synthesized using a four drivers system (tetrahedron), we choose to build a cubic source with 6 independently drivers. As a matter of fact, thanks to the specific geometrical symmetry of the cubic shape some observations can lead to the simplification of the signal processing implementation of these different canonical directivities. The monopole will require a common filter for all loudspeakers, while the dipole will need identical, but out of phase, filters for two opposite sides. Thus, the overall implementation needs only a limited number of filters. Once again, a header module, common to all elementary directivities, processes a minimum phase equalization of the composite source. Figure 3 shows the magnitude and phase response of the specific residual filters dedicated to the monopole and dipole directivities. Recently, a fourth directivity (one of the second order spherical harmonic) was added to the subset.

With these basic directivities the composite source can generate a directivity family simply by weighting them with gains (or frequency dependent gains). This avoids complex interpolation process between the set of filters corresponding to different directivities. The weighting function between the 0-order (monopole) and the 1^{rst} order (dipoles) directivities controls the cardioicity of the combined pattern. The weighting within the 1rst-order family governs the orientation of the pattern. This procedure may be associated to a simple user interface allowing real time manipulation of the combined directivity (cardioicity coefficient and 3D orientation).

Experimental results. In order to increase the bandwidth of the system, this geometrical structure is repeated in three frequency bands and with different size in order to stay, as far as possible, within the limits of spatial aliasing problems. Hence, the mid-frequency (250Hz-2kHz) system is made of 7" drivers mounted on a 23cm cube, while the high-frequency (2kHz-20kHz) system is an 8cm cube. Current sub-bass system has been simplified keeping only 4 horizontal drivers.

Performances of the so-built source are currently investigated to determine its possibilities. Figure 4 shows the reconstruction of the monopole and dipole with the mid-frequency system. It can be noticed that, approaching the higher cut-off frequency (2kHz), spatial aliasing already creates progressively secondary radiation lobes. Beyond this frequency limit, the process hands over to the smaller high-frequency cube, thus maintaining aliasing problems within acceptable limits.

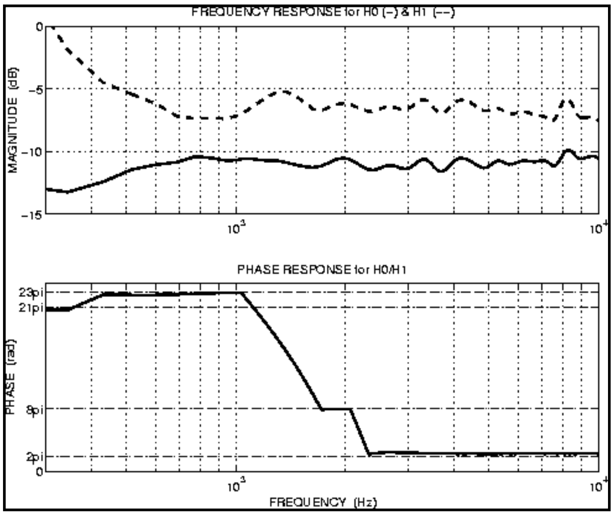

Figure 3: Upper*: magnitude of residual filters corresponding to the monopole (solid), and the dipole (dashed).* Lower*: phase difference between the two filters.*

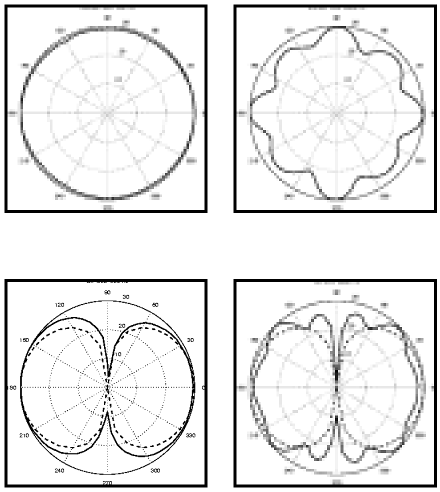

Figure 4: *Monopole (up) and dipole (down) reconstruction with the mid-frequency system at 500Hz (left) and 2kHz (right). Dashed line is the theoretical dipole (dB scale).*

OBJECTIVE CHARACTERIZATION OF THE LISTENER'S STANDPOINT

As expected, the approach brings a significant gain in terms of control commodity and perceptual efficiency. The architecture also allows working with several signal sources, each one being sent with different weights on the basic directivities. A composer or sound designer may then shape the radiation behavior of several sounds that are played simultaneously on the same source, keeping them perceptually segregated, thanks to the different directivity perceptual properties.

In order to give an objective characterization of the perceptual sensation linked to the different canonical directivities, their corresponding impulse responses were measured in a small concert hall (Ircam's Espace de Projection). Considering the first order spherical harmonics, it is possible to control the directivity index of the source (DI) from +4.7 dB, when the cardioid or dipole patterns face the audience, to theoretically $-\infty$ when the listener is in the zero of the dipole. This directly controls the perceived direct to reverberated energy ratio. However, it is known that the critical distance (beyond which the reverberation level is predominant compared to the direct sound) is often small in concert halls, when compared to the dimension of the audience area. Hence this would, at first glance, cast doubt on the interest of such control in real situations. However, the spatial distribution of the early part of the response is also known to drive important perceptual attributes such as *Apparent Source Width* and *Room Envelopment*.

These effects being linked to the time vs spatial distribution of the energy, results are analysed using different microphone directivities. In order to check the potential of the chosen basic directivities, figure 5 presents the time evolution of the frontal energy measured with a cardioid microphone facing the source, and the ratio between frontal and lateral energy, the latter being measured with a bi-directional microphone. Left graph of figure 5 shows that, when the source dipole is oriented vertically or laterally, the lack of frontal energy persists as late as 160ms, where it tends to converge to the behavior of other orientations or directivities. Right graph of figure 5 also shows some perceptual properties that relate to the frontal to lateral ratio of energy according to time. It is, for example, interesting to compare this value for the different orientations of the dipole. As expected, the frontal dipole will give the more precise localization of the source, this tendency being also supported by the first reflections. Also expected, the lateral orientation of the dipole presents low values of the frontal/lateral balance from the very beginning of the response, which will give the sensation of a wide source and high room envelopment. The vertical dipole, in spite of a low direct sound level (fig.5, left graph), is associated to a late frontal balance providing the weakest source width impression and low room envelopment. This comes from the zero plan of the dipole that is horizontal in this case. Hence no lateral reflections can be created. The resulting impression is like a monophonic sound. These properties can be easily noticed when listening to sound examples, and allow assigning perceptually relevant identities to the sounds created or reproduced by the composite source.

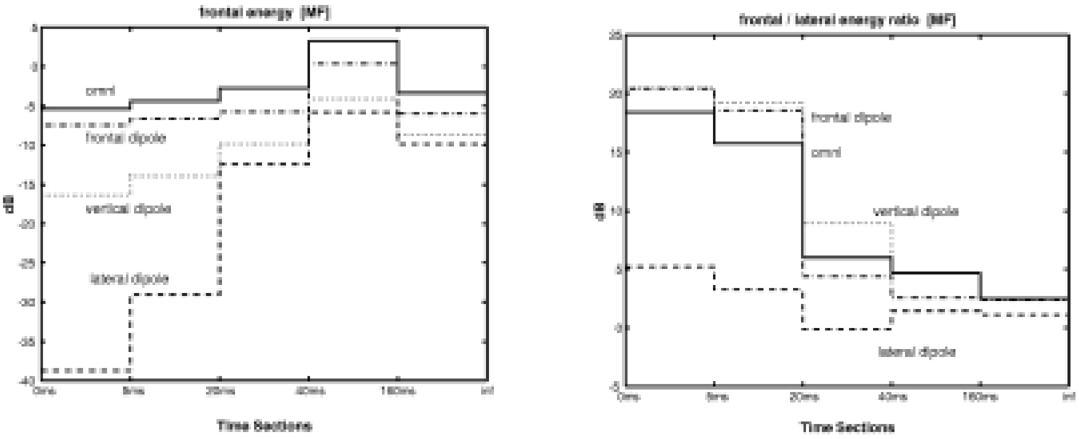

Figure 5: *Analysis of the room impulse response for various loudspeaker directivities.* Left: *time evolution of the frontal energy measured with a cardioid microphone facing the source.* Right*: time evolution of the ratio between frontal (cardio) and lateral (bi-directional) energy.*

ANALYSIS / SYNTHESIS OF POWER SPECTRUM

In the specific case of a musical instrument simulation, the spatial reproduction needs also to be frequency dependent. In a first implementation, to comply with this fundamental requirement, the parameters used to control the combination of the basic directivities are replaced by frequency dependent coefficients. Two symmetrical equalizers (3-band filters) are weighting the monopole/dipole ratio so as to allow, for instance, low frequencies to be omni-directional, medium frequencies to be mono-directional (cardioid) and high frequencies to be bi-directional (dipole). The reason for using two symmetrical equalizers is to ensure a constant power spectrum. Thus, it is possible to control independently the spectrum emitted in the direction of the audience (responsible for the direct sound) and the power spectrum, responsible for the frequency content of the room effect. This contribution yet improves significantly the handiness of the system.

Synthesizing the musical instrument power spectrum. Although the chosen basic directivities cannot relate for the directivity details of a musical instrument, room acoustics properties tell us that a faithful reproduction of its power spectrum is crucial. For that purpose, we use a traditional acoustical power analysis method (using a reverberant room) adapted (although not exact) to a common concert hall [5]. A chromatic scale played on the instrument is used as a reference signal recorded in the far field by one or several bi-directional microphones, in order to avoid the capture of the direct sound. Reverberation time compensation and diffuse field equalization of the microphones allow computing an estimation of the power spectrum of the instrument. Examples are shown on figure 6.a for different stops of a harpsichord. However, in order to feed the composite source, a far field microphone cannot be used, for it carries a room effect. Then, an approximation of an anechoic power spectrum signal of the source must be synthesized from a close microphone signal processed by a correction filter.

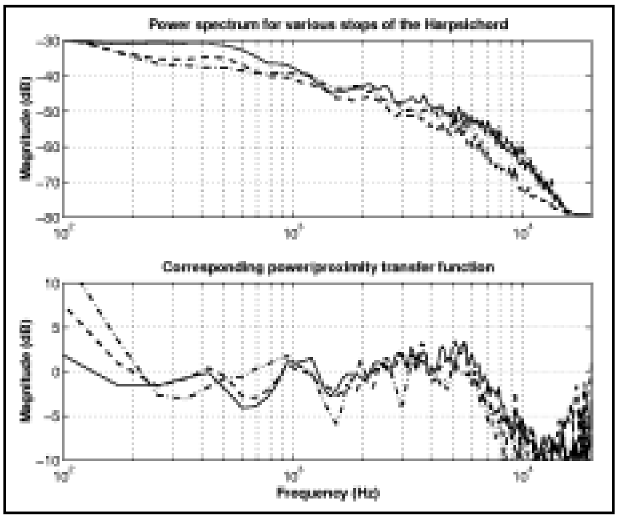

Figure 6a: Upper*: diffuse field spectrum for various stops of the harpsichord.* Lower*: diffuse field / near field transfer function.*

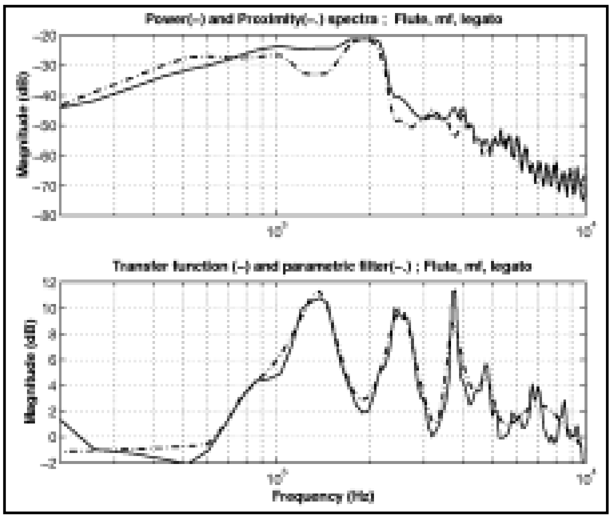

Figure 6b: Upper: *diffuse field (-) and near field (--) spectra of a flute.* Lower*: Transfer function (-) and parametric filter (--).*

This filter is derived from the ratio between the above estimated power spectrum with the spectrogram of the chromatic scale captured with the close microphone. Figure 6.b shows the correction filter magnitude in the case of a power analysis of the flute. This power signal will then be able to 'feed' the composite source, which has been previously equalized in order to deliver a flat diffuse field response for any basic directivity.

CONCLUSIONS

Following a previous research dedicated to the reproduction of radiation properties, a new method has been developed and tested on a prototype. It is based on the design of a 3D lay out of transducers and on a prior synthesis of the first order spherical harmonics. This architecture allows governing continuously the cardioicity coefficient and the orientation of the combined directivity. Signal processing implementation is simplified and stays efficient when several sound signals are simultaneously played with different directivities. It also affords independent control on the direct sound and radiated power spectra, which is important for the reproduction of musical instruments. Objective characterization of a listening situation was performed with this directivity subset. It showed various and significant modifications of the time vs spatial energy distribution according to the chosen directivities, and proved their potential to create a perceptually relevant radiation vocabulary in a musical context.

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