DIRECTIVITY SYNTHESIS WITH A 3D ARRAY OF LOUDSPEAKERS APPLICATION FOR STAGE PERFORMANCE

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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. The present study, 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 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. For that, an analysis/synthesis method of the power spectrum is used to provide the automatic design of a correction filter to applied on near field recording before feeding the source. Results are discussed considering room acoustics criteria and examples of musical instruments.

1. INTRODUCTION

A sound source can generally be characterized by three properties : timbre related to spectral and temporal attributes, intensity and directivity providing spatial informations on the sound radiated from the source.

As a matter of fact, current diffusion systems allows rather accurate control on timbre and intensity – even by correcting their spectral and temporal responses with fitted signal processing – but are quiet unable to reproduce faithfully the spatial characteristics (directivity) of the source of which they diffuse the sound. This is mainly due to the fact that most loudspeaker systems are closed to a baffled piston model which radiation can't be significally modified. However, in an enclosed space, such as a concert hall, radiation properties play an important role since it affects the time and spectral distribution reaching at listeners's ears : that's what basically makes the perceptual differences between a real instrument put aside a loudspeaker 'playing' the same signal.

The aim of this study is to simulate the directivity of real sound sources, like musical instruments, or in a larger frame to assign to the sound diffusion a given radiation function. This new step requires to design a new generation of transducers based on a multi-driver architecture that finally can be conceived as a 3-dimensionnal array of loudspeakers.

Several works have yet produced consistent results in the radiation control 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.

2. REPRODUCTION METHOD

2.1. 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 figure 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.

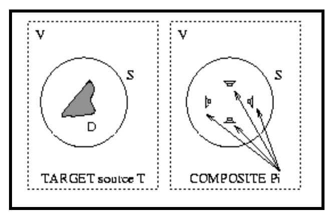

Figure 1. *Radiation reproduction*

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), 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:

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

where $\{a_i\}$ are complex coefficients related to frequency ω.

2.2. 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, in a fitted algebraic space (Hilbert), should lead to N equations/unknowns system giving, for each ω, the values of $\{a_i\}$:

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

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

2.3. DSP implementation

Then, from a signal processing point of view, the computed $\{a_i\}$ coefficients are filters to apply to the composite. Figure 2 shows the implementation used for reproducing a given directivity pattern. The filtering is processed in two main stages. First, a common filter designed from the average magnitude response of the $\{a_i\}$ that leads to obtain a global spectral equalization of the source and can be modeled by a parametric IIR filter. Secondly, specific residual filters that keep in important phase relations between the different drivers and, in consequence, require a FFT convolution process using their FIR characterization.

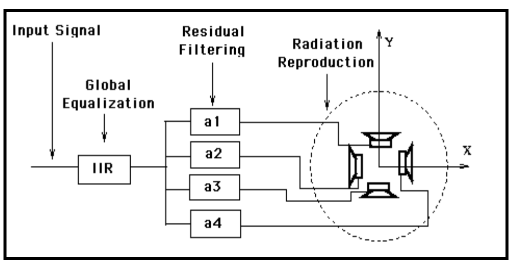

Figure 2. *Digital signal processing architecture*

2.4. First prototype & results

This procedure was first applied to a dodecahedron [3][4] holding, for technological restrictions, only four independent subsets of loudspeakers.

To confirm the merits of the method, a characterization of the synthesized directivities was made, with theoretical directivities targets 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.

3. SUBSET OF DIRECTIVITIES

From these basements, we carried on the work with a rather derived approach and in a musical application aim. The constraints on accuracy of directivity reproduction are fallen in aid of other requirements such as enlarged bandwidth or real-time processing and versatility of the control. For that, we built a new prototype on a cubic shape, getting six independently driven sources [5].

3.1. New prototype & implementation

To increase the bandwidth, this geometrical structure is repeated in three frequency bands – with different size – in order to stay, as far as possible, within the limits of spatial aliasing problems (figure 3). Hence, mid-frequencies (200Hz–2kHz) go through 7" drivers mounted on a 25cm cube while a 8cm cube equiped with tweeters deals with high-frequencies (2kHz–20kHz). Finally, a simplified sub-bass system (4 horizontal drivers) is devoted to low-frequencies.

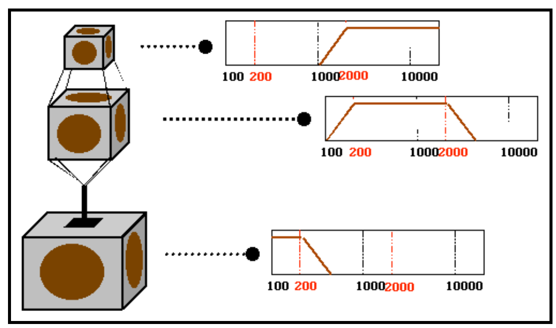

Figure 3. *Three-way multi-loudspeakers device*

As for signal processing, a new architecture is designed. Four elementary directivity patterns are permanently synthesised: the monopole H0 (zero-order spherical harmonic) and the dipoles H1 (first-order spherical harmonic) in the three canonical directions of the cube (X, Y, Z). Moreover, thanks to the specific geometrical symmetry of the cubic shape, the implementation is simplified: H0 will require a common filter for all loudspeakers, while H1 will need identical, but out of phase, filters for two opposite sides.

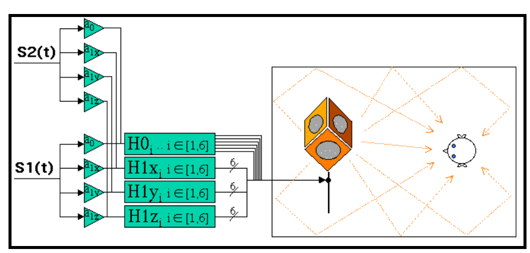

Figure 4. *New implementation: combination of basic directivities (H0, H1x,y,z) by a set of coefficients (a0, a1x,y,z) for two input signals S1(t) & S2(t)*

Once again, to keep coherent relations between the two directivity families (H0 & H1), the design of filters must be carefully done, especially in the phase domain: a header module, common to all elementary directivities, delivers a minimum phase equalization of the composite source and specific residual filters are dedicated to monopole and dipole (figure 5a). These processes ensure a *white* spectral behaviour with regards to the average spectrum produced by the two basic patterns and no difference in phase between H0 and a frontal point of the positive lobe of H1 (figure 5b).

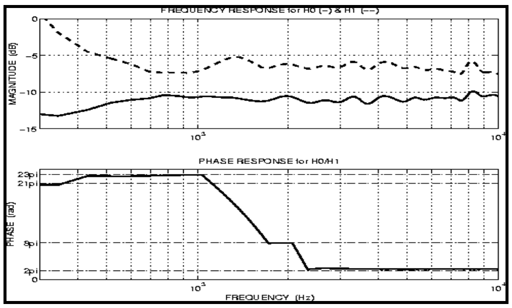

Figure 5a. *Residual filters: magnitude for H0 [—] & H1 [– –] {up} and phase ratio {low}*

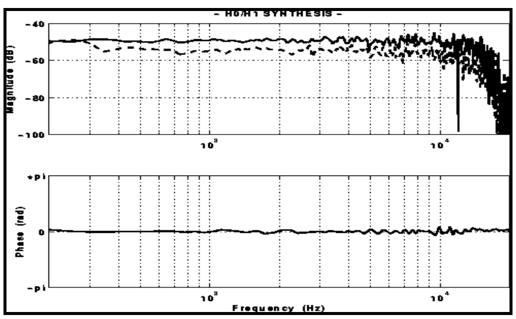

Figure 5b. *Power spectrum for synthesised H0 [—] & H1 [– –] {up} and frontal phase ratio {low}*

From these elementary directivities the composite source can simply generate *intermediate* patterns by weighting them with gains. It avoids complex interpolations between sets of filters related to different directivities: the weight between H0 and H1 controls cardioïcity of the combined pattern and the weight within H1 rules its orientation.

Thus, the overall implementation needs only a limited number of filters, the four filtering modules H0, H1x, H1y and H1z (figure 4) being processed before dispatching the signal to the six drivers. Moreover, this architecture also allows to reproduce simultaneously different source signals, each one having its own radiation: a set of weighting coefficients is simply related to each source signal (figure 4).

Finally, this procedure may be associated to a simple user interface allowing real-time manipulation: for each source signal, a slider for the cardioïcity coefficient and a trackball-like controller for the 3D-orientation (figure 6).

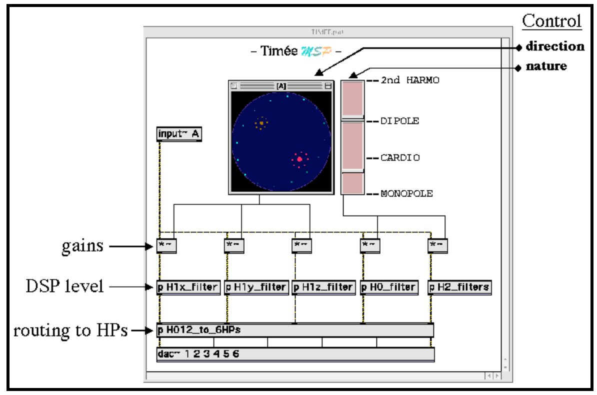

Figure 6. *Real-time algorithm (Max/MSP)*

4. OBJECTIVE CHARACTERIZATION

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.

4.1. Spatial reproduction

Performances of the so-built source are first investigated in term of reconstruction. Figure 7 shows synthesis of basic directivities measured at 650Hz with the mid-range system. Because of spatial aliasing, these results will go degrading around the high cut-off frequency (2kHz). However, beyond this limit, the process hands over to the smaller HF-cube, thus maintaining aliasing problems within acceptable limits.

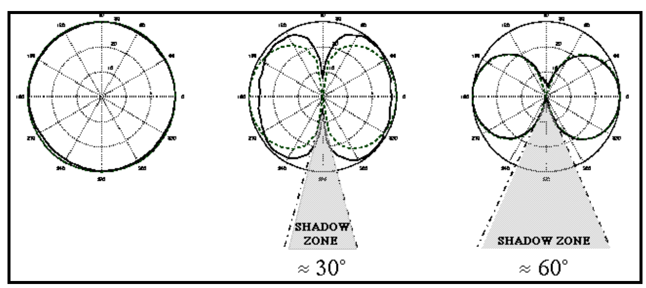

Figure 7. *Reproduction of HO {left}, H1 {center} and R2 {right} at 650 Hz (theoretical reference in [– –])*

4.2. Characterization of the listener's standpoint

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

– ∞ 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, figures 8 present 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. Figure 8a 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.

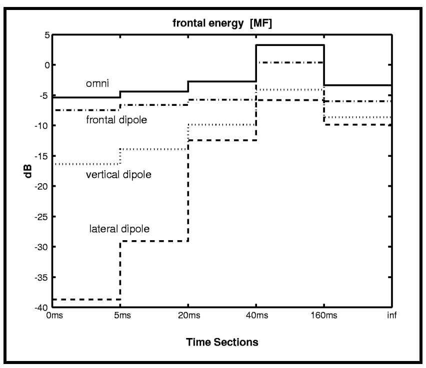

Figure 8a. *Time evolution of the frontal energy (measured with a cardioïd microphone)*

Figure 8b also shows some perceptual properties related 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 (seen in figure 8a), 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.

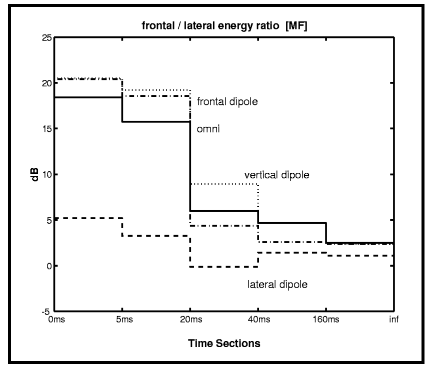

Figure 8b. *Time evolution of the ratio between frontal (cardio mic.) and lateral (bi-directionnal mic.) energy*

This being, the use of H1 appears to be very critical, especially on stage. Actually, this pattern gets a 30° *shadow zone* aperture defining the area where no direct sound is coming from the source (figure 7). Practically, this means that only a reduced part of the audience would *hear* most of directivities as they should be heard. In order to overcome this issue, a third basic directivity R2 is introduced, resulting from the combination of a second-order spherical harmonic (H2) and the monopole. As H1, this new pattern is bi-directionnal but with narrower beams, offering a larger *shadow zone* as shown on figure 7 (≈ 60°). However, because of the limited number of transducers (6), R2 can only be obtained on the three canonical axis X, Y, Z and can't be freely oriented in other directions.

5. 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.

5.1. 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 [6]. 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 9a for different stops of a harpsichord.

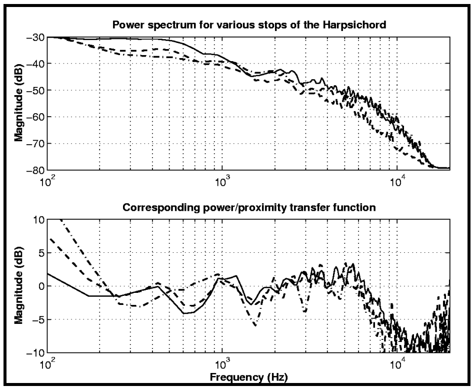

Figure 9a. *(UPPER): diffuse field spectrum for various stops of the harpsichord. (LOWER): diffuse field / near field transfer function.*

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.

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 9b 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 (§ 3.1).

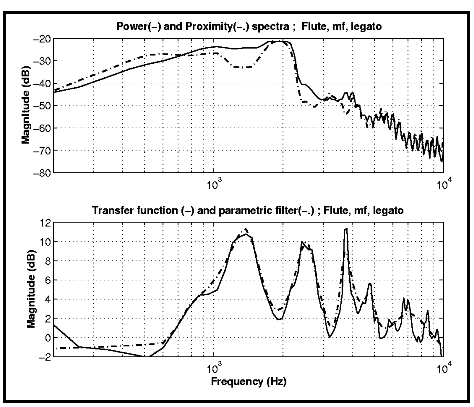

Figure 9b. *(UPPER): diffuse field [—] & near field [– –] spectra of a flute. (LOWER): Transfer function [—] & parametric filter [– –].*

6. CONCLUSIONS

A previous study has provided the basements of a theoretical and applied procedure to the reproduction of given directivity patterns by means of digital signal processing on an electroacoustical device.

From that point, a slightly different way has been explored with the aim of reaching a flexible and musical approach to control the sound diffusion radiation. For this, a new method has been developped and tested on a prototype: it relies on the design of a 3D lay out of transducers and on the permanent synthesis of the first order spherical harmonics. Then, the specific architecture allows the control of two main parameters: the cardioïcity coefficient – continuoulsy between omnidirectionnal to bi-directionnal – and the orientation of the combined directivity. Moreover, the implementation is simplified enough to be compatible with a real-time application and stay efficient with several input signals, each of them getting its own directivity pattern. Finally, the system also affords independent control on the direct sound and radiated power spectra, which is important for the reproduction of musical instruments.

Thus, the most advanced prototype gives yet interesting results: it tends to reproduce quiet faithfully the elementary directivities (first spherical harmonics) and an objective characterization of a listening situation 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.

7. REFERENCES

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