MULTI-LOUDSPEAKER REVERBERATION SYSTEM

Fernando Beltrán and José Ramón Beltrán

Department of Electronic Engineering and Communications
University of Zaragoza, Spain
beltran@posta.unizar.es
http://www.cps.unizar.es/~fbeltran

ABSTRACT

In this paper we present a reverberation system based on a multi-loudspeaker configuration. The aim of this work is to produce a natural sounding reverberation system with a similar pattern to the produced in real rooms. A new method for sound spatialization is presented, and it is used to locate on the virtual room's surfaces the early reflections produced in a reverberant space. Each loudspeaker is fed with its corresponding early reflection sequence after the panning process, and an artificial late reverberation is added in each channel, obtaining a more realistic experience that the obtained with other artificial reverberation systems.

1. INTRODUCTION

Reverberation is a very well known effect that is present in almost all our listening experiences. It creates an ambient space in the sound perception. In reverberant environments, the sounds we hear are accompanied by delayed reflections from many different directions. In most cases, this additional information changes the perception of the sound, modifying it loudness, timbre and spatial characteristics [1]. This effect is particularly important for music listening, because it adds life and sense of space. The reverberation is associated with the architecture and acoustic of concert halls, and this factor is critical for good sounding in concert performances. In recorded music, there has been a great effort is simulating this effect through some electro-acoustic devices. Nowadays, digital signal processing techniques are mostly used for these purposes.

We can study a reverberant space as a distributed sound source element: each wall, ceiling and floor are contributing with thousands of reflections within the reverberation time. Using the geometric model of a reverberant space, and with the help of the image source method or ray tracing method, we can determine the reflections generated in each surface, considering them as distributed sound sources. It is then possible to apply one of the existing panning methods to position a virtual sound source using loudspeakers, obtaining an impulse response for each loudspeaker. With this information is then possible to implement an artificial reverberation algorithm for each loudspeaker. Processing the dry sound with the different reverberation algorithm obtained for each loudspeaker, and reproducing the processed sound in a multi-loudspeaker system, we can get the spatial impression of a reverberant ambient.

This paper is organized as follows: in chapter 2 some ideas about sound panning are summarized, and the method used in our system is presented. In chapter 3 we describe the artificial reverberation algorithm implemented in this work. The implementation of the multi-loudspeaker reverberation technique is depicted in chapter 4, and conclusions are presented in chapter 5.

2. SOUND PANNING IN A MULTI-LOUDSPEAKER SYSTEM

The sound spatialization problem is addressed essentially in two ways: headphone-based systems, using the appropriate HRTF's to process the sound and to get the exact duplication of what the ear would hear in a natural situation, and loudspeaker-based systems, allowing the creation of spatial auditory images with the help of a set of loudspeakers around the listener.

In general terms, the techniques to position a sound source in the space using a multi-loudspeaker layout are based in the method of applying different amplitudes to the loudspeakers [2]. Some of the most interesting are Ambisonics and VBAP. One important restriction is assumed in these systems: it exists only a very restricted portion of the audience area where optimal listening conditions are met (sweet spot). Among all sound spatialization systems, we will focus on systems using several loudspeakers in the horizontal plane.

2.1. Ambisonics

Ambisonics technique was developed by M.A. Guerzon [3], [4]. In this system the B-format encoding is the most widely used. This format allows to represent sounds situated in the horizontal plane with 3 signals: W, X and Y. By applying a suitable transformation matrix (decoding) to these signals, any bi-dimensional array of speakers placed around the listener can be driven. The Ambisonics coding do not provide any distance information. This must be added by controlling different factors such as loudness, direct to reverberant sound ratios, etc.

For a plane wave of amplitude S and angle of incidence θ (azimuth angle), B-format encoding equations can be written:

$$W = S * \sqrt{2}/2$$
$$X = S * cos\theta \qquad (1)$$
$$Y = S * sin\theta$$

X and Y channels contain information of the components of the sound on the x axis and y axis, respectively. W channel contains a monophonic mix of the sound material. X and Y may have negative values, implying that the signal is stored in antiphase when compared with the signal in the W channel.

2.2. VBAP

Vector Base Amplitude Panning was presented by V. Pulkki [5]. In this system, considering an N loudspeaker layout placed in the horizontal plane at the same distance from the listener, a pair of loudspeakers (a base) is selected for each position of the virtual source, and these two loudspeakers are fed with the source signal multiplied by gain factors. The gains of all the other loudspeakers are set to zero. If the virtual source is located in the same direction as any of the loudspeakers, the signal emanates only from that particular loudspeaker. As stated in the Ambisonics system, the VBAP system does not provide any distance information.
When the loudspeaker base is orthogonal, the gain factors are equivalent to those calculated for the Ambisonics system:

$$g_1 = cos\theta$$
$$g_2 = sin\theta \qquad (2)$$

2.3. Distance vector panning system

A new method for sound panning in a multi-loudspeaker system is presented here. This formulation includes both azimuth and distance information of the virtual sound source. In a coordinate system where the x axis is pointing forward and the y axis pointing to the left, and considering a system of N loudspeakers placed in a circumference of diameter *d* (Fig. 1), we can define N vectors $\vec{u}_i$, with their origin in each loudspeaker, pointing to the virtual sound source *P(x,y)*.

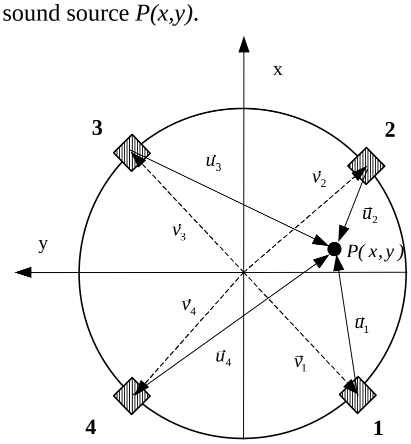

Figure 1. *Four loudspeakers system configuration*.

Considering a circumference of radius 0.5, we propose that the gain corresponding to each loudspeaker can be defined as follows:

$$a_i = 1 - d_i^2 \qquad (3)$$

where $d_i = d(P, \vec{u}_i)$ is the distance from the virtual sound source position P to each loudspeaker.
To assess the quality of localization perception in a multi-loudspeaker system the velocity and energy localization vectors are often used. These measures were proposed by Gerzon [3] for the development of optimal Ambisonics encoders and decoders, and have been used for assessing other 2-D and 3-D panning techniques [6], [7]. The velocity model deals with low frequency theory of human auditory localization (below 700 Hz). The velocity vector definition is:

$$\vec{r}_v = \frac{\sum_{i=1}^{N} a_i \vec{v}_i}{\sum_{i=1}^{N} a_i} = \left(r_v^x, r_v^y\right) \qquad (4)$$

where a_i is the amplitude gain of the i_{th} loudspeaker in the layout, and $\vec{v}_i$ is the unitary vector pointing to the i_{th} loudspeaker (see figure 1). The velocity vector represents a synthetic plane wave with an apparent direction of propagation with angle θ_v. Using this model the factor $|\vec{r}_v|$ must as close to 1 as possible for optimum quality, and $\theta_v = atan\left(r_v^y / r_v^x\right)$ gives the perceived virtual source position.
For frequencies above 700 Hz, the energy vector is defined as a modified version of equation (4), where the loudspeaker gains are replaced by their squared magnitudes:

$$\vec{r}_E = \frac{\sum_{i=1}^{N} a_i^2 \vec{v}_i}{\sum_{i=1}^{N} a_i^2} = \left(r_E^x, r_E^y\right) \qquad (5)$$

In this case the energy vector describes the direction of the apparent sound source, with angle $\theta_E = atan\left(r_E^y / r_E^x\right)$, at high frequencies. Its amplitude $|\vec{r}_E|$ must be as close to 1 as possible for optimum quality.
Finally, it is important to observe the energy preservation constraint, so the sum of squared gains must equal to 1.

2.4. Distance perception

In the panning method proposed here, the information about the distance of the virtual sound source to the listening position is given. This fact is imposing the reinterpretation of the above mentioned quality factors to include the distance perception.
Considerations about θ_v and θ_E are identical to those described in [3], i.e., they represent low frequency and high frequency perceived virtual source azimuth, respectively.

In our system, the amplitude of vectors $\vec{r}_v$ and $\vec{r}_E$ are directly proportional to the distance of the listening position to the virtual sound source. To obtain comparable results with other systems, where values close to 1 are admitted as high quality index for amplitudes of velocity and energy vectors, we have included a normalization coefficient r that is the distance from *P* to the listening position. So, the following velocity and energy vectors are used in this work:

$$\vec{r}_v' = \frac{\vec{r}_v}{r} \qquad \vec{r}_E' = \frac{\vec{r}_E}{r} \tag{6}$$

At all the possible sound source positions at the same distance of the listening position the energy conservation constraint is preserved, but this constant changes with *r*. Thus, a correction coefficient *k* is adopted taking into account the parameter *r*:

$$k = \frac{2}{\sqrt{9+16r^4-16r^2}} \tag{7}$$

2.5. Results

Some results are presented in different graphics. We consider four loudspeakers placed in a circumference with a diameter *d=1*. In figure 2 we can observe a polar diagram showing the gain of each loudspeaker corresponding to a virtual sound source moving over a circumference with radius *r=0.5*, i.e., a circumference passing over the loudspeakers. The gain of each loudspeaker is maximum when the virtual sound source is placed over this loudspeaker, but al least other two loudspeakers are also fed (this is similar to the Ambisonics system). Figure 3 shows the polar diagram for virtual sound sources over a circumference with radius *r=0.3*. In this case all the loudspeakers are fed for every position of the sound source.
In figure 4 we present the evolution of the normalized velocity and energy vectors for distances from *r=0* to *r=0.5* (solid line is the velocity vector and dotted line is energy vector). In figure 5 we present the difference (in degrees) of the perceived direction of the sound and the real one, $\theta-\theta_v$ and $\theta-\theta_E$. In both cases the differences are close to 0 for all the positions of the virtual sound source and for all radius. According with these results, with energy vector amplitude close to 1 for every radius, high frequency perception is very good. Because the values of velocity vector amplitude decrease when approaching to the center of the circumference, i.e., when the sound source is approaching to the listening position, the perception of low frequencies is worst for small radius.

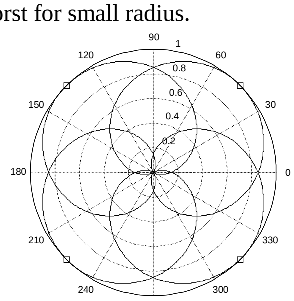

Figure 2. *Polar diagram of loudspeaker gains, r=0.5.*

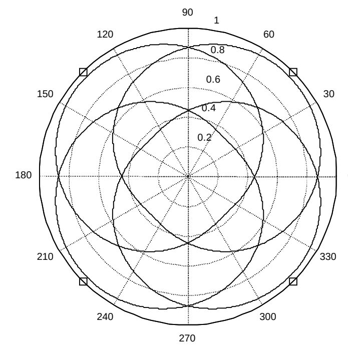

Figure 3. *Polar diagram of loudspeaker gains, r=0.3.*

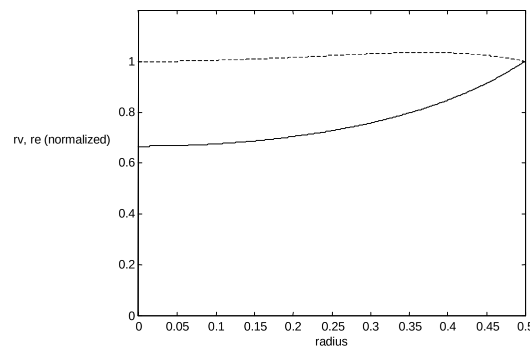

Figure 4. *Evolution of the amplitude of normalized velocity vector (solid) and normalized energy vector (dotted) with the radius.*

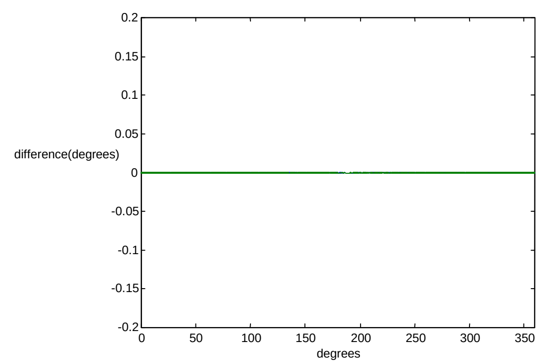

Figure 5. *Difference of perceived and real sound direction.*

3. REVERBERATION ALGORITHMS

Implementing a reverberation algorithm is a well-known subject that involves two processes: the modeling of the early reflection response corresponding to the early echoes and the simulation of the late reverberation or diffuse reverberation. Reverberation algorithms usually include directional filters intended to reproduce localization cues. The best-known algorithms have been summarized in [1]. Figure 5 presents the simplest block diagram of a reverberation algorithm.

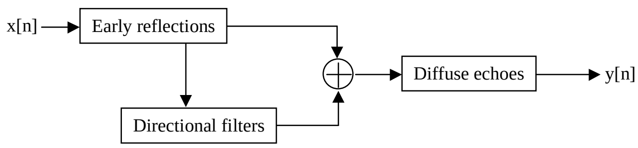

Figure 5. *Reverberation block diagram*.

3.1. Modeling early reflections

In order to model the early reflections a simple FIR filter structure is implemented with a set of delays m_i and tap gains a_i (see figure 6).

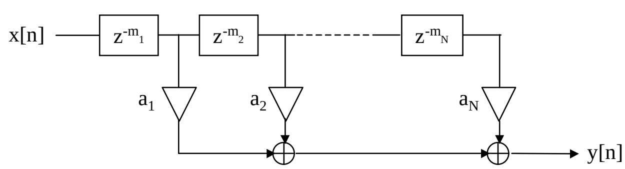

Figure 6. *Early reflections implementation*.

The delays m_i and the gains a_i are obtained computing the impulse response of a virtual space. This impulse response can be obtained with different algorithms: ray tracing and source image methods are the most usual ones. In our case, we are dealing with simple geometry, so the source image method is an efficient algorithm to simulate the early response of a virtual room. This method has been also used for complex geometry models, for instance in the DIVA project [8].

3.2. Simulating late reverberation

A slightly most difficult decision is to choose the algorithm used to simulate the diffuse reverberation. From the first Schroeder work about artificial reverberators [9], many implementation have been presented. The works of Gardner [10], Dattorro [11] and Jot [12] offer different approaches to obtain an artificial reverberation algorithm with some specific characteristics: natural sounding, high echo density and control of the reverberation time; all of them are based on recursive filters. Two important aspects should be taken into account in the election of the reverberation algorithm: the output sound quality and the processing time. In this work we are interested in sound spatialization using the early reflection pattern. The subjective quality of the diffuse echoes algorithm is not our objective in this paper, so we have implemented different late reverberation algorithms [13]. Using this system it is possible to select among different types of those algorithms to check the differences on the final results.

3.3. Directional filtering

Directional filtering in reverberation algorithms is usually based in HRTF functions obtaining a stereo signal to be heard with headphones. In our case we are dealing with a multi-loudspeaker system, so a different approach has been made. In the next section we present our strategy to compute a directional reverberation with spatial impression.

4. MULTI-LOUDSPEAKER SPATIAL REVERBERATION

The reverberation phenomenon implies thousands of sound reflections coming from the different surfaces of the acoustic space. The first step we have to cover is the computation of the impulse response of a virtual space. This process implies the calculation of three parameters for each reflection:

i. The attenuation due to consecutive reflections a_i,
ii. The delay associated with each one m_i, and
iii. The exact position in the room surfaces where the last reflection is produced before arriving to the listening position (x_c, y_c).

Using the source image method for a simple geometry (shoebox), we have obtained the analytical expressions of the last reflections positions:

$$\begin{aligned} x_c &= sign(n)\times min\left(\left|\frac{x_n L}{y_m}\right|, W\right) \\ y_c &= sign(m)\times min\left(\left|\frac{y_m W}{x_n}\right|, L\right) \end{aligned} \tag{8}$$

where (n, m) is the n-th reflection on the x axis and the m-th on the y axis, (x_n, y_m) is the virtual source position for the (n, m) reflection and (W, L) are the room width and length respectively.
This information allows us to allocate each reflection in the space with the help of the panning algorithm shown in section 2.
An important remark has to be made: to locate the reflections at the room walls and to locate the direct sound in the space, absolute position is important, so it is necessary to perceive azimuth and distance. This perception is reached with the algorithm proposed in section 2.
From each reflection a_i corresponding to the set of early reflections of the virtual room we obtain, after the distance vector panning process, N ones: $(a_i^1, a_i^2, a_i^3, \ldots, a_i^N)$. In this case, a_i^j is the corresponding weight for each loudspeaker associated with the early reflection i, and it is computed using expression (3). So, we obtain a different impulse response for each loudspeaker in the sound reinforcement system.
In figure 7a we can see the impulse response associated with a room which width and length equal to 20 m. and 30 m., respectively. The reflection coefficients are 0.85 for all the surfaces. The sound source position is 14 m. behind and 9 m. to the left of the listening position (the center of the room). Figures 7b to 7e present the early reflection pattern associated with each loudspeaker in a four loudspeaker layout after the sound panning method.

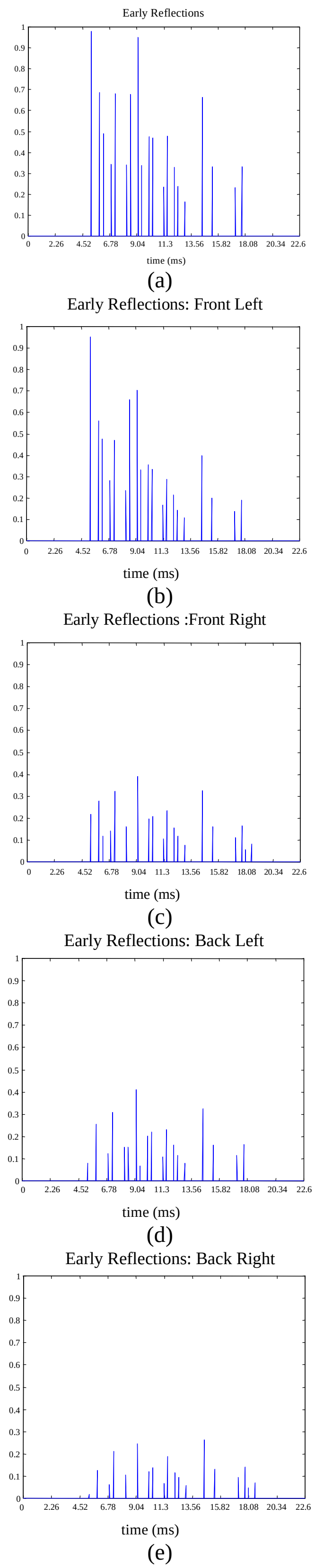

Figure 7. *Early reflections for a virtual room before (a) and after (b)-(e) the sound panning process (see text).*

Our reverberation implementation is depicted in figure 8. To process the early reflections associated with the corresponding loudspeaker we need N FIR filters, each one similar to the filter presented in figure 6. The delay associated with every early reflection is the same for the N filters, but the gains are now a_i^j depending on the loudspeaker *j*. Directional filtering is not used because directivity is obtained from the early reflection position.

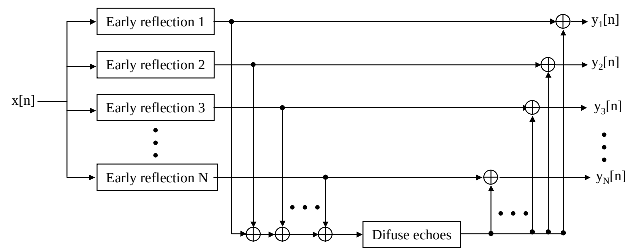

Figure 8. *Directional reverberator block diagram*

A FIR filter is included for every loudspeaker in the system. The algorithm for obtaining the diffuse reverberation tail could be shared. Some kinds of decorrelation between loudspeakers are obtained in the same ways described in [1].

5. CONCLUSIONS

A method for multi-loudspeaker reverberation implementation has been introduced. This system combines techniques to characterize the acoustical behavior of a virtual room, sound spatialization techniques and a reverberation algorithm implementation. The virtual room reverberation pattern is characterized by means of the source image method, obtaining its impulse response. For sound spatialization, we have developed a new sound panning method, taking into account the distance and the azimuth of the virtual sound source. The reverberation algorithm used for a multi-loudspeaker layout is built based on the perception of the early reflections coming from the different surfaces of a virtual room, considering each of them a virtual sound source. Because of this model we need to consider their distance and azimuth referred to the listening position. This sound panning method has been subjected to some validation tests, obtaining very good results in sound source azimuth and distance perception. The output signal fed to each loudspeaker is obtained processing the input signal with a different impulse response for each loudspeaker. These impulse responses are obtained applying the panning method to each early reflection in the virtual room, and using an artificial reverberation algorithm to create the late reverberation.

6. REFERENCES

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