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Acousmonium and sound spatialization. Study and musical implications in Nouvel´s Auditorio400 at Museum Reina Sofía in Madrid

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Abstract
This article presents new ways to listen to the music using the parameter of sound spacialization. In this paper we present results of measures realized in Auditorio400 at the National Museum and Center of Art Reina Sofia in Madrid. These measures indicate SPL in each point of the room and show us zones with acoustic problems. Hereby the placement, directivity and power of loudspeakers that compose the Acousmonium system in the hall had been corrected.

The Auditorio400 of Madrid, designed by Jean Nouvel, has installed the system named "Acousmonium" that was developed in GRM's studies of the ORTF in Paris. It consists of 32 loudspeakers distributed suitably to give acoustic signal to the whole room. This system allows to simulate acoustic properties of the traditional instruments or to change its characteristics of radiation, musical clarity and acoustic harmonics distribution. By means of this system it is possible to create a virtual space where the sound can be recreated in any geometric point of the room with the use of computer simulation.

Keywords: Sound spacialization, Auditorio400, Acousmonium,

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1 Musical Space

The parameter “space” was not considered part of music concept until recently. Nevertheless, we need to make mention several authors that make interesting use of this parameter. The term space makes reference to 3 basic concepts: space and localization of sound sources, the moving of trajectory of sound, and the sense of a virtual space where the audience is just inside the sound.
First, Gabrielli make concerts for several choirs at the Venice cathedral. Lately, in the Twentieth century the North American Charles Ives and Henry Brant used several orchestras distributed in different spaces. In the second half of the past century, Stockhausen, Carter, Cage, Xenakis, and others uses several groups and orchestras to have the sensation of music into the space. However, the big revolution took place when electronic and computers introduce technology into the music. Xenakis designed the POLYTOPE space in Paris, where he used sound and light to produce a new concept of

spatiality. Stockhausen developed at Osaka World Exhibition a semi-spherical hall with 50 speakers. And a new spatial designed hall for Space "Salle de Projection" was constructed at IRCAM (Institute of Recherche pour la Composition et Acoustique Musicale) in Paris.

1.1 History

The first examples of space as a parameter in musical properties, is found in the "call - response" in the antiphonal music from the very beginning of history of music. In the twentieth century music, Charles Ives often called for the spatial separation of performing forces in an effort to facilitate the differentiation of two or more simultaneous—relatively independent—layers of sound. In the music of Ives, the divided ensembles are often not separated within the performance space but instead are demarcated by registration, which amounts to separation in pitch space. The second movement of his Fourth Symphony is such a case as Harley [1] describes in his PhD dissertation. Ives likewise supplied only general instructions as to the physical separation of instrumental groups. He viewed the spatial distribution of musical forces in performance as an "aspect of interpretation" and therefore leaves it somewhat open (Erickson, [2]).

Brant [3] points out that the harnessing of physical space as a compositional resource is considered "optional" and ancillary by most composers, and he emphasizes that in his own music, "the spatial distribution of the performers throughout the hall is a planned, required, and essential element of the music."

One of Brant's primary arguments is that the spatial separation of sound sources may be used to "disentangle" dense textures. In short, the unwanted or inharmonious blending of tones may be thwarted by the spatial diffusion of performers. He equates the spatial distribution of sound sources to the composing of multiple melodic parts that do not overlap in octave range, essentially conceiving of sectors in actual space as discrete segments of pitch space.

Brant makes some mention of *moving* sonic events. He states: "When instruments placed in fixed positions [adjacent to one another] begin playing one at a time and *accumulating* (staying in)...there is a compelling impression of the hall tangibly *filling* up with sound [and also] of the sound *traveling* gradually."

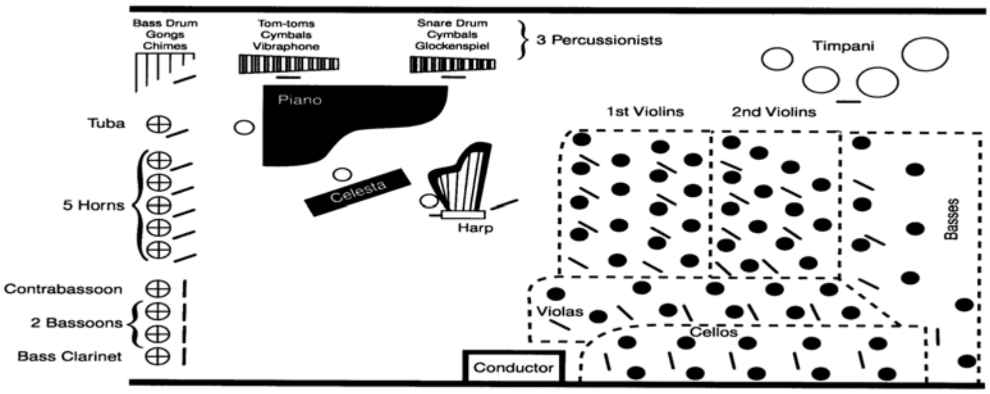

Figure 1- Settings for H. Brant "Desert Forest"

Many twentieth-century composers began to concern themselves with the actual motion of sound sources or, more commonly, with the composite motions and spatial shapes arising from the consecutive activity of multiple sound sources. In this case, can be useful to mention the contributions of Stockhausen ("*Gruppen*" for 3 orchestras), Cage ("*Pieces for 5 orchestras*"), and Carter ("*Symphony for 3 orchestras*").

Ives, Brant, Stockhausen, and Berlioz, all make the point that by the proper positioning of instruments and instrumental groups the composer can delineate, separate, and clarify relationships between textures and their component sound elements. Brant's work was composed for a specific performing space; Berlioz and Stockhausen composed somewhat independently of any particular space; and Ives, while keenly aware of its effects, considered the management of the performing space part of the interpreter's task. Brant composed the vertical spatial dimension of pitch, Stockhausen asked for movement (wandering) of the sound between groups, and Ives drew attention to the effect of distance.

Although Berio's "*Circles* "(1960) is "often cited as the earliest work that requires the soloist to move from one stage position to another," Brant's "*Hieroglyphics I*" of 1957, which likewise calls for performer mobility, predates "*Circles*" by three years. Brant's "*Windjammer*" (1969) includes "specific walking plans for the performers of the wind quintet". In Musgrave's "*Concerto for Clarinet and Orchestra*" (1969), the soloist is required to traverse the entire orchestra— intermittently activating and participating in small, concertino groups.
The following examples came from one percussion piece by Iannis Xenakis, ("*Persephasa*") where the players are placed around the audience, and in this way, as the score shows, the sound moves from one place to another

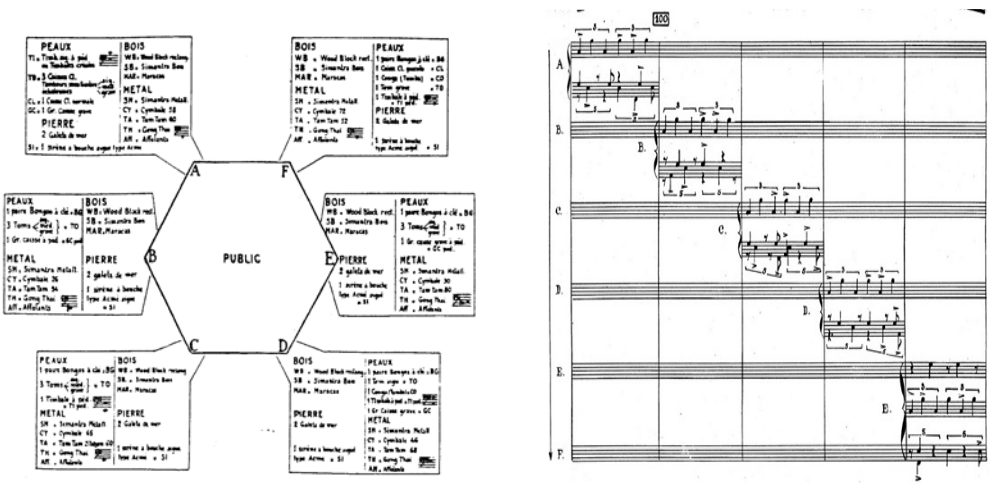

Figure 2 - Xenakis' settings and example of "Persephasa" score mm 99-103.
Counterclockwise circular gesture

1.2 Electronic

The more sophisticated use of spatialization came from the field of electronic music an computer music. Harley notes: "The experience of hearing spectacular movements of sound images in space during the concerts of "*musique concrete*" in the early 1950s played a formative role in the development of Stockhausen's and Boulez's theories of spatialization."
Stockhausen [4] introduces the term *Kinematic relief* to describe the manipulating the trajectory of sounds between loudspeakers and corresponds to Stockhausen's practice of mobilizing both acoustic and electronic sounds.
One of the first highly successful multichannel works, Stockhausen's "*Gesang der Jünglinge*" (1956) made effective use of the space "surrounding" the audience—both immediate and distant. According to Stockhausen, this work was his first attempt to make "the direction and movement of sound in space" evident as a "new dimension for musical experience." Other important early electronic works include: "*Poème Electronique*" (1958) by Edgard Varèse, which was transmitted over approximately

400 loudspeakers in the Philips Pavilion at the Brussels World's Fair in 1958, and Stockhausen's "*Spiral*" (1969), which required the construction of a spherical auditorium at the Osaka World Fair in 1970. During performances of "*Spiral*", the audience sat upon a transparent platform in the middle of the sphere, and Stockhausen controlled the projection of sound over 50 loudspeakers dispersed throughout the sphere. At this point, it can be interesting to note that the Ambisonics system for sound localization, has returned to this concept of sphere.

1.3 Computer Simulation

In the 1970s, John Chowning worked diligently to design computer algorithms that accurately control and modify the azimuth (horizontal and vertical angle of displacement), distance, and velocity (speed and directionality of motion) of a sound within a quadraphonic environment. While endeavoring to simulate a wide array of sounds in a variety of replicated acoustical spaces, Chowning likewise concerned himself with simulating the actual movement of those sounds within the illusory spaces. Chowning [5] states: "A special program has been written which allows the user to specify an arbitrary *path* in a two-dimensional space [one exclusive of elevation] by means of a light pen or a computed geometry. The program evaluates the trajectory and then derives the time functions which control distance and angle for simulation...Using interactive graphic display techniques, a user can specify the location and movement (trajectory) of a sound. The program computes the control functions for azimuth, distance, and velocity which are used to modulate the signal to be applied to the loudspeakers". Chowning's program likewise factors in cues to mimic the *Doppler Effect*, which is the phenomenal shift in frequency experienced as a moving sound source passes the listener. When a sound-emitting source approaches and passes the listener, the subject experiences a decrease in the sound's pitch. The amount of decrease is dependent upon (and directly proportional to) the speed of the traveling source: the higher the speed, the greater the change in frequency. Thus, the extent of a frequency shift provides a cue for gauging the speed of a moving sound source.
Chowning's work is important in part because he achieves remarkable spatial effects with only four speakers. Many other composers and sound engineers have developed extensive diffusion systems involving multiple loudspeakers.

2 Space and localization

Computers use the advantage of specially design algorithms to simulate a virtual space. Auditory localization is the human perception of the placement of a sound source. In listening to music in a concert hall, a listener receives cues for sound location from the placement of the actual sound sources in the hall. With an understanding of the perception of sound source location, the placement of the apparent source of sound can also be used as a compositional element. A 3-D sound system uses processes that either complement or replace spatial attributes that existed originally in association with a given sound source. The space can be simulated with an appropriate array of speakers
The sense of space in an auditorium came from the acceptance of three characteristics that can define the feeling of 3D sound: Direction, Distance, and Motion
The direction is commonly expressed in terms of two angles: the azimuthal angle which is measured in the horizontal plane passing through the center of the listener's head; and the elevation angle, which is measured in a vertical plane bisecting the listener.
The distance, is defined by concepts derived from SPL, Reverberation (D/R, Early Reflection, late Reverberation) and for the way the hall adapt the sound due to air absorption and walls material.
The motion of the sound uses the consequences of the Doppler Effect.
In addition, it is necessary to mention the effect that the pinna (outside ear), produces in the sense of the sound.

2.1 Cues for direction

In order to define the sense of space in simulation by , it is necessary to use some cues that can help to produce that feeling. ITD, IID, ITD ENVELOPE.

- ITD, Interaural time difference is any delay a listener perceives between the time that a sound reaches one ear and the time that it reaches the other.
- IID, Interaural Intensity difference is the level difference that is perceived in each ear. It indicates the distance and intensity panning of the source.
- ITD ENVELOPE, this cue is based on the hearing system's extraction of the timing differences of the onsets of amplitude envelopes, rather than of the timing of the waveform within the envelope.

2.2 Cues for distance

Three principal cues help a listener judge the distance of a sound source: a) the intensity of the sound b) the effects of reverberation (the ratio of reverberated to direct sound (D/R ratio), the early reverberation, and the late reverberation) and c) the amount of high-frequency energy in the sound.

2.2.1 SPL

The characteristic of speakers must be similar for sound spatialization and the SPL must have a value almost uniform in the entire hall.
One of the main objectives in the design of auralization is the design of correct loudness or power of a speaker.

2.2.2 3-D Sound, Distance, and Reverberation

The inclusion of distance and environmental context effects within a 3-D audio system is almost imperative for maintaining a sense of realism among virtual acoustic objects. In fact, the same physical cue—reverberation—is a key for the effective simulation of distance and the perception of an environmental context. Distance and environmental context perception involve a process of integrating multiple cues, including loudness, spectral content, reverberation content, and cognitive familiarity.

- ***The D/R ratio and Distance Perception***

The reverberant-to-direct sound ratio (D/R) has been cited in many studies as a cue to distance, with varying degrees of significance attributed to it. Von Békésy [6] observed that when he changed the D/R ratio, the loudness of the sound remained constant, but a sensation of changing distance occurred. He noted that "though this alteration in the ratio between direct and reverberant sound can indeed be used to produce the perception of a moving sound image, this ratio is not the basis of auditory distance".

- **Specific Perceptual Effects of Early Reflections**

-Echolocation
It is possible to cue distance differences corresponds to a time delay range of about 4–6 msec.

-Timing and Intensity
The perceived timbre and attack qualities of sound sources, especially of musical instrument sounds, can be changed substantially by early reflections without affecting intelligibility. For example, one hears a cello, not only from vibrations emitted from the instrument itself, but also from early reflections from the floor and walls.

- **Late Reverberation**

Late reverberation is informative perceptually as to the volume of a particular space occupied by a sound source.
Santala [7] states that room reverberation is able to mask the phasing effect. Spatial aliasing is further reported to be capable of masking the side-effects of ideal order truncation, which normally occur at the border of the area of best listening.

2.3 Motion of Sound Sources

The preceding discussion has been concerned with the creation of an illusory location for a stationary source. When the source location is moved about rapidly, another acoustical phenomenon comes into play. The Doppler Effect describes the change in pitch that results when the source and the listener are moving relative to each other.
The simulation of a "natural" situation that includes a Doppler shift of frequency can be problematic, because it alters pitch succession, which is often employed as one of the principal means of affecting musical continuity. The most successful applications of Doppler shift in music are those in which the shift of frequency is an integral part of the musical continuity and not simply an "special effect" that can be expected to hear. Figure 3 shows two musical pieces with motion in its structure.

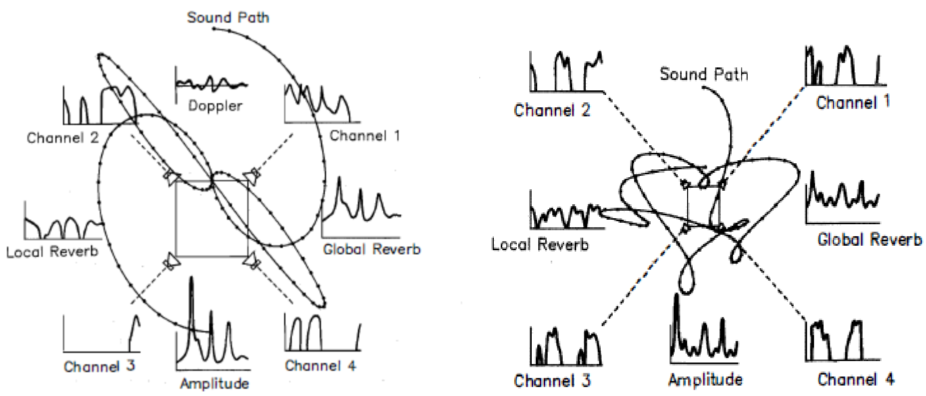

Figure 3 - Chowning' "Turenas" and Reynolds's "The Palace"
Examples of moving trajectories

2.4 Effects of the Pinna

In addition, the pinna performs filtering that varies with the direction from which the sound reaches the listener, particularly at higher frequencies. This filtering helps the listener determine whether a sound comes from above, below, in front, or behind.
The spectral modification of the pinnae can be thought of as equivalent to what a graphic equalizer does in one's stereo system: emphasizing some frequency bands, while attenuating other bands. This is notable from the standpoint that both timbre and spatial cues depend on the spectral modification of a sound source.

2.5 Characteristics and Utility of cues

The usefulness of the cues depends on the frequency content of the sound. Both ITD and IID cues are practically ineffective when the spectral energy of the sound resides below about 270 Hz; hence, the direction of such sounds cannot be determined. ITD cues are most effective in the frequency range of

270 to 500 Hz but contribute little above 1400 Hz, although differences in the arrival time of the attack portion of the envelope of a high-frequency sound can be used for localization. The amount of intensity difference increases with frequency, so that above 1400 Hz IID cues predominate.
When a sound is long enough, listeners uncertain about its location commonly turn their heads so that one ear is closer to the source.

3 Software Algoritms

The Processing the audio signal for real or virtual space can take advantage of some packets for DSP that are available for the composer sound engineer, or any other person that can be interested in this subject. In the early 80´s appeared the C program and with this development, the old music software as MUSIC V, developed for Max Mathews, took new look and C-music (San Diego University), C-sound (MIT) and C-mix (Princeton University) give the technicians a big power to manipulate the audio signal. In the 90's, real time technology, and C++, change again the universe of sound processing, and programs like MAX/MSP and Pure Data, offer a new look to process the sound.
Inside this program, the theory derived from space distribution of acoustic signal, allow the design of patches and algorithms for the auralization of a sound source.
Several techniques are been used to simulate the sense of space of a audio signal: HRTF (Head Related Transfer Function), Ambisonics, Holophony, Binaural Sound, Q-sound, and the last one and maybe the best one WFS (Wave Field Synthesis).
Among them, there are some specific for headphones, as binaural and HRTF, an others, were designed for speaker situation. HRTF, can work well in a hall settings, if the transfer function are well selected, but it is difficult to introduce in a hall with dimensions like a concert hall. It is very interesting because with convolution and digital filters HRTF is able to simulate the harmonic change produced by the pinna section of the ear.
The best situation for the simulation of space in concert hall setting is accepted the Ambisonics technology. The first use of this technology was a recording with 4 microphones that give information of the sound localization inside a hall.
At the present moment, the computer can generate information that is introduced in the information of the sound, and give good sense of localization.
In 1997, Pulkki [8] introduce the concept of VBAP (Vector Based Amplitude Panning), and it is the first approach to the concept of sound spatialization in programs as Pure Data and MAX/MSP.
These properties imply that VBAP produces virtual sound sources that are as sharp as it is possible with any loudspeaker configuration and amplitude panning.
The gain factors corresponding to each loudspeaker are summed up to form one gain factor for each loudspeaker. The resulting gains are normalized. The listener perceives still a single virtual source.

3.1 Historical survey

The first examples came from algorithms developed inside the C-program. The composer Roger Reynolds uses the unit generator ESPACE, inside C-Music, for several of his pieces with good results. Other example is John Chowning's *Turenas*. He uses C-Sound software to have the possibility to have different arrangements for the control of the out signal in its algorithm.
We can see how this software works in the figure 4.

```
#include >carl/cmusic.h>

SPACE (input [b1], 1) RV*;

Where
RV = X[bvpn,] y [bvpn], theta[bvpn], amp [bvpn,
]back [bvpn]
```

RV stands for radiation vector consisting of five pieces of informationspecifing the x coordiantes the y coordinates in meters, the direction theta in radians, the aplitude sacalar called amp, and a back radiation parameter indicating the apmlitude opposite to theta direction.

```
nchnls = 3
instr 1
i2pi = 2*3.141592
ifi1 = 0
ifi2 = i2pi/3
ifi3 = 2*i2pi/3
kp line 1,p3,10
kfi line 0,p3,2*i2pi
kpan1 = (.5+.5*cos(kfi-ifi1))^kp
kpan2 = (.5+.5*cos(kfi-ifi2))^kp
kpan3 = (.5+.5*cos(kfi-ifi3))^kp
a1 rand 30000
outc a1*kpan1,a1*kpan2,a1*kpan3
endin
```

Figure 4 - Examples of C-Music and C-Sound

3.2 Panning with VBAP in MAX/MSP

MAX/MSP is a graphical programming environment. MAX is designed for processing of events, and MSP is an extension of it designed for real-time audio applications.
The VBAP implementation consists of three objects: “define_loudspeakers”,” vbap”, and “matrix~”. An schematic representation of the implementation is shown in Figure 4
A vbap object is attached to each generated sound signal. The user may design controls for direction and spreading for it. The loudspeaker setup is controlled by using “define_loudspeakers”. The matrix~ object performs distribution of sound signals to loudspeakers. When the patch is applied for different loudspeaker setups, only the settings of “define_loudspeakers” and” matrix~” object has to be updated.

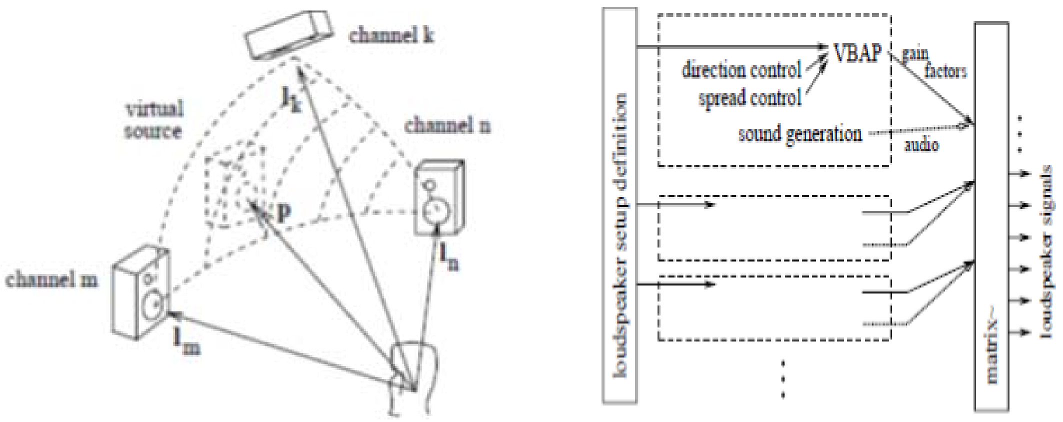

Figure 5 -VBAP vector and Schema of Spacialitacion MAX/MSP

3.3 Spatialisateur

The IRCAM developed a packet to work as part of the MAX/MSP called SPATIALISATEUR. The Spatialisateur project started in 1991 as a collaboration between *Espaces Nouveaux* and *Ircam*. Its goal is to propose a virtual acoustics processor which allows composers, performers or sound engineers to control the diffusion of sounds in a real or virtual space.
Spat~ is an effort to organize and optimize the experimental patches developed in the Spatialisateur project, in order to make them accessible to musicians and researchers who work with Max/MSP. The current release allows reproduction on multi-channel loudspeaker systems in studios or concert halls. It also integrates 3D stereo reproduction modes for headphones (binaural) or 2/4 loudspeakers (transaural), as well as Vector Based Amplitude Panning (VBAP,) and Ambisonics.
The library of Max objects which compose Spat~ is divided in three main categories of objects: DSP objects, low-level control objects, high-level control objects (see figure below). The purpose of this

organization is to allow easy configuration and construction of custom remote control panels or mechanisms for Spat~.

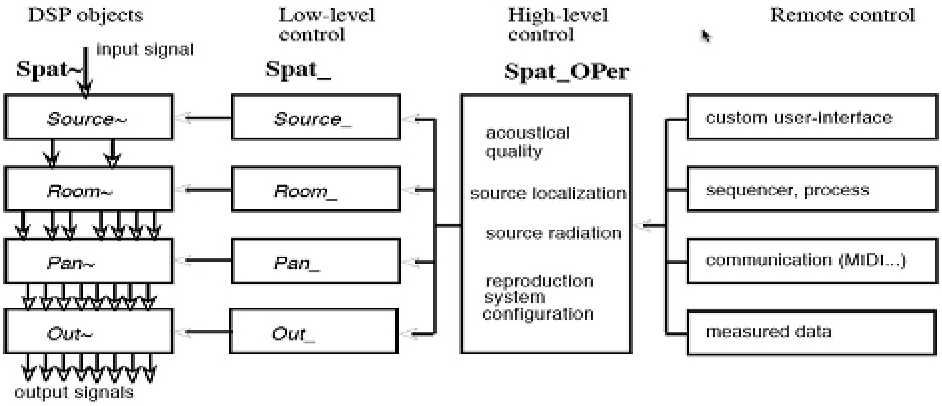

Figure 6 - Schema of Spatialiteur

The signal processing in Spat~ is divided in four successive stages, separating directional effects from temporal effects:

- Pre-processing of input signals (Source~)
- Room effects module (reverberator) (Room~)
- Directional distribution module (Pan~)
- Output equalization module (Decoder~)

3.4 Other options

Many other composers and sound engineers have developed extensive diffusion systems involving multiple loudspeakers. Jonty Harrison, famous for his work with BEAST (Birmingham Electro-Acoustic Sound Theatre), proclaims that eight speakers (the “main eight”) are “the absolute minimum for the playback of stereo tapes.” In 1974, François Bayle created a “loudspeaker orchestra” called the *Acousmonium*. This construction “consisted of eighty loudspeakers of various sizes placed across a stage at different heights and distances from the proscenium.”
In the late 1980s, Peter Otto and Nicola Bernardini (and others) developed a system called TRAILS (Tempo Reale Audio Interactive Location System), a “multi-loudspeaker network [consisting of] a twenty-four-by-eight spatialization matrix.” This configuration involves 192 speakers.
There is an special program designed by Ton Erbe, SOUNDHACK, where the sound can be manipulated to give any of the formats for sound spatialization: Binaural. Ambisonics, HRTF,
Other program inside the MAX/MSP packet, is VIMIC (virtual microphone) where it is possible to simulated the position of several different microphones, with different information of distance, reverberation.

4 Auditorio 400

The Auditorio 400, part of the ampliation of the Museum Reina Sofia in Madrid, adapted the system Acousmonium for the diffusion and sound design for the concerts that every season the Spanish Ministry of Culture offers in this hall.
The system has 32 speakers from different models and names: Meyer, Bose, Genelec, Yamaha, etc.
On the stage there are 16 speakers: 4 Bose, 10 Meyer, (including 2 subwoofer) and 2 Genelec,

On the audience area there are 12 Meyer around the public, as the crown in Acousmonium setting, and 4 Yamaha in the middle of the audience.
This can suggested an Area of Best Listening, just in the center of the Auditorium.
The authors of this paper realized some testing and simulations to see how the speakers work in the hall. The SPL in the hall, with the entire system working, has a reasonable acceptable level.
In order to study how the Acousmonium works for the space, we simulated and study the behavior of several speakers. Some examples of this simulation is presented in the figure 7.
We can see that the SPL close to every speaker is bigger than in the rest of the area, if only this speaker is radiating energy. The area close to each speaker is not adequate for 3D-sound.

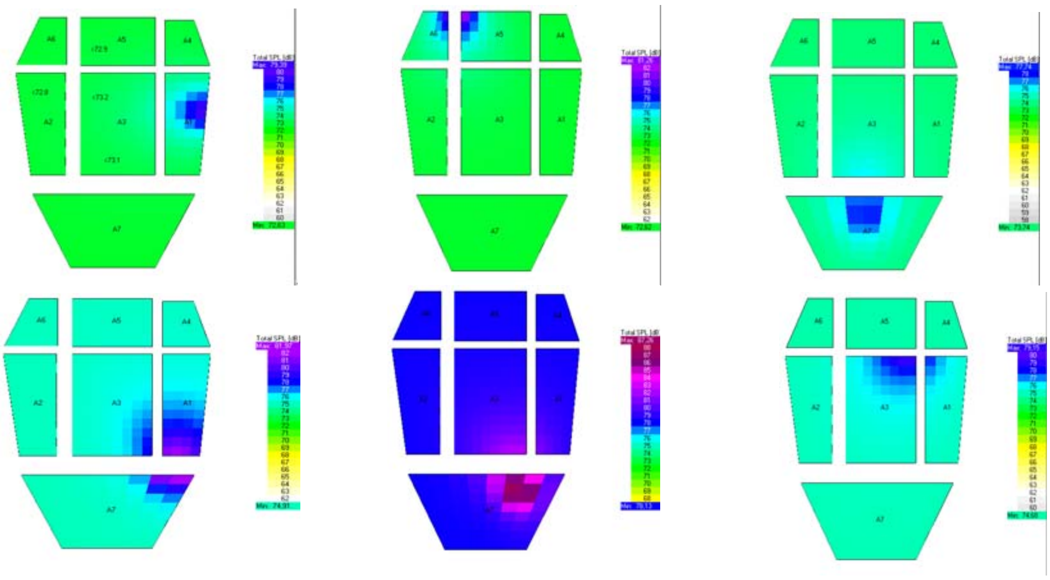

Figure 7.- Examples of SPL for a single speaker

This originates a problem for spatialitation and localization. Nevertheless, the big problem in the Auditorium is the reverberation that the hall presents. Its values is elevated for the music that the center normally presents: Chamber Music and electroacoustic music.

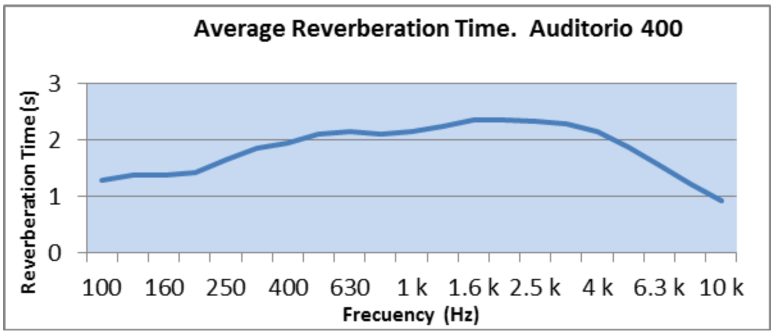

Figure 8- Reverberation Time vs. frequency

The reverberation of the hall, as we stated earlier, is one of the big references in localization of source. The early, cluster, and late reverberation give us cues in the direction of the source. From these values, the hall present problems with speech clarity C50, especially when the source sound presents fragments with spoken voice, that is normal in contemporary music. From the same reason, the musical clarity C80, present problems with harmonics and definition of the qualities of a specific instrument. This is also accentuated, because the absorption in the area reduces harmonic content, dependent on situation of the listener.

5 Acousmonium

The Acousmonium, as new instrument for music diffusion was thought for "tape music" and was conceived after the ideas of Pierre Shaffer in the book "Traité de objets musicaux" where he explains the techniques for "musique concret" used from 1948. It was presented in Paris by François Bayle in 1974 inside "L´ Espace Cardin" as a new Acoustic Experience.
The Acousmonium system allows two different ways to present a piece:

- Offers the piece as it is, as the final product without sound manipulation. According with the original recording, the composer or sound engineer, only need to choose which track goes to which speaker.
- Offers a real time version inside a specific space, changing some parameters of the piece, as localization of sound sources, equalization, and the introduction of some effects.

This last possibility gives the option to have a real player or group as sound source. The Acousmonium is like introducing a degree of freedom inside a fixed media like is the recording of a musical piece. The acousmatique concert offers the composer a final stage in composition process with the possibility of multiplication and diversification of listening points.
With these new possibilities, the composer or the sound technician has the possibility to change in real time some parameters of the work and have a "*mise in scène*" for every space and time. Among the parameters that the Acousmonium allows to control are:

- o Dynamic Variation that it can depend on the hall, the public, and some external aspect outside the essence of the music. The faders of the mixer desk allow the possibility to emphasize some moments of the music.
- o Color or equalization, the change of the spectrum, both, for adaptation of the hall and for coloration of the music.
- o Space: The Acousmonium gives the degree to create a 3D virtual sound space, modify the distance, angles and reverb of the sound source, and change de density of the musical discourse, vertical density (harmonic content) and horizontal density (pulsation or tempo of events)

The localization of the sound comes from speakers characteristics and for the form of the music.
In this sense, we can speak about:

- ▪ Localization of the speakers, stage, side, back, audience level, upside audience, etc.
- ▪ Density of the music, changing the "orchestration" of a musical source.
  - Tutti as layers of orchestral sound, like Ives proposed
  - Solo as an unique source, moving and transforming itself inside the hall
- ▪ Velocity of the motion of elements of the sound. This concept defines the degree of elements that transform its trajectory, as Stockhausen proposes in his early electronic pieces and in the last opera "*Light*".

5.1 Lutherie of the instrument

The Acousmonium is based on:
- An Orchestra of Speakers with different color and characteristics, which will be located at stage, and around the audience. This last localization is known as "crown"
- A Mixing Desk, that will be used not for control the total sum of the sound, but the solo and individual speakers of the instrument.
- A Desk of several DSP elements: Equalization, Gates, Compressor, Expander, Reverb, etc,..

The speakers must be used after a special study of power, radiation, and answer in frequency. Its implementation requires some special considerations.
The crown can be used as reference, for power amplification, for proximity of sound and for distribution of sound sources. The speakers must be able to create a virtual sound space, for what is required a dissymmetry.
The Mixing Desk and some DSP associated with the mixer, allow Effects: (sound planes, Solo source); EQ (sustain medium and high area of spectrum, Presence high area of frequency) Cross Planes and velocity (distance, trajectories) and Inclination, (delay, and other considerations).
The Acousmonium, is like a musical instrument, that, besides its origin, can be used for new sound algorithms for spatialization, as VIMIC (Virtual Microphone), Ambisonics, HRTF, Spatialisateur, etc.

6 Conclusions

The Auditorio400, designed by Jean Nouvel, has dimensions that make it appropriate for chamber music and electroacoustic music. The introduction of Acousmonium was a decision of the LIEM (Laboratory for Computer and Electronic Music), center created for the Spanish Ministry of Culture, and works reasonably well with electroacoustic music.
The Acousmonium, in collaboration with appropriate software and hardware, can address in an appropriate way the concept of space in a musical discourse. Every single speaker of the crown setting creates a problem of proximity for auralization simulation. It was solved, in first instance, with the introduction of four speakers just inside the hall around the central area of the audience. This solution originates an additional problem, due to the directivity of these four speakers. A new solution can came from the use of speakers with option of change of directivity. This property is being investigated in these moments. Nevertheless, the Auditotio400 has its big problem in the reverberation time in the hall, derived from its geometry and its absorption levels. This is especially serious when the sound source includes spoken voice, because of the clarity of the source is reduced.

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