

AML | ARCHITECTURE AND MUSIC LABORATORY

A MUSEUM INSTALLATION

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A museum-exhibit gives the visitors an opportunity to experience the relation between sounds and performance space. The impulse responses of real rooms have been measured and a simulator (Spatialisateur) has been programmed to imitate these spaces. Elementary volumes are imitated using simpler algorithms. A tracking system enhances the spatial illusion by compensating for the movements of the listener's head.

ARCHITECTURE FOR THE EYES AND FOR THE EARS

When we describe an architecture, we first make use of pictures and we often underestimate that buildings as well as halls and rooms have peculiar acoustical properties. Nevertheless the acoustics contributes to the ambience of a room and can be defined as a unique signature of this room. The architecture "for the eyes" is already represented in the ZKM | MediaMuseum by several exhibits showing virtual architectural environments. In order to give the visitor of the museum an auditory grasp for architecture, the ZKM | Institute for Music and Acoustics was commissioned to design an installation that deals with architecture "for the ears" [1, 2].

In order to make apprehensible the acoustic properties of various rooms, we must give the visitor the opportunity to enter, in a virtual meaning, the rooms. With different dry sounds or musical excerpts he can explore, "light up", the unknown rooms. The relation between sound and room is in the forefront of the installation. The peculiarities of existing and non-existing rooms should be pointed out. For example a piece of classical music can be played in a concert hall, in a church, in a bathroom, in a roman theater as well as in basic shapes such as a small cube, a cylinder or a sphere.

The purpose of this paper is to give an overview of the exhibit AML | Architecture and Music Laboratory which is now in operation, to explain how the system was designed to meet the requirements of the museum, to report how some real rooms were chosen and measured and to

describe how elementary volumes were selected and implemented within the exhibit.

1. OVERVIEW OF THE INSTALLATION

The visitor can select a room by touching one among 12 illustrated buttons, each of them representing a different room. He can also select a sound like music, speech or natural sounds, including his own voice which he can project into the virtual room (Fig 1). The plastic design of the exhibit is due to the dutch artist Frank den Ouden, while the technical development and environment is in charge of the ZKM | Institute for Music and Acoustics.

The components of the installation can be divided into four parts, the user's interface, the sound source, the reverberation system and the amplification. The visitor controls via the interface which sound should be combined with which selected space. The signal resulting from these choices is amplified and presented to the visitor by the mean of headphones. We have chosen to use headphones instead of loudspeakers for several reasons:

- headphones give a better calibrated reproduction, with no additional reverberation due to the listening room;
- the listeners can better concentrate on the sounds because they are acoustically isolated from the museum's background noise, which can be fairly high when many interactive installations are simultaneously in operation;
- the large available dynamic range enables the visitors to perceive the fine structure of the reverberation.

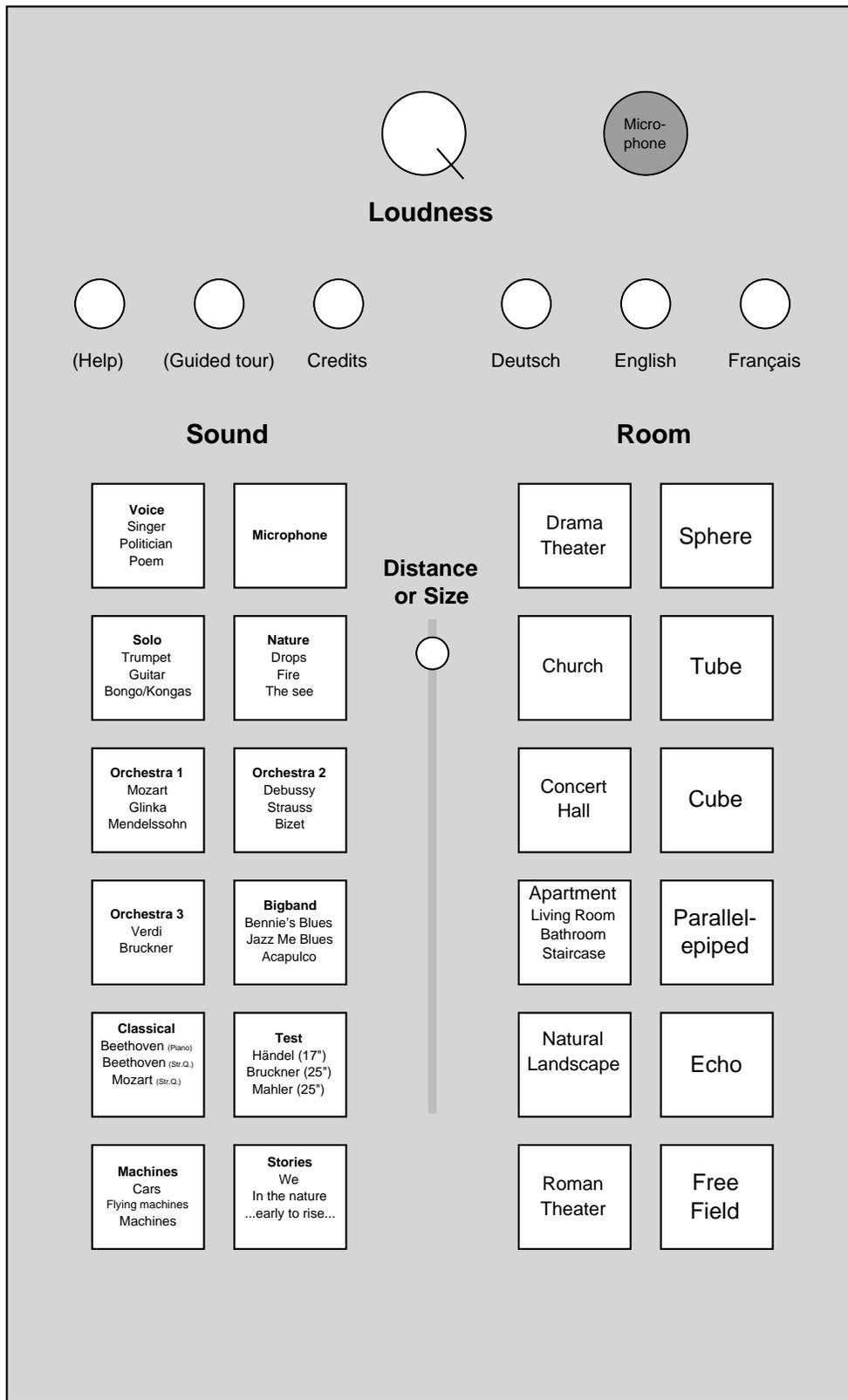


Figure 1: The AML control desk

ated signals;

- each headphone can be set at a different reproduction level in order to suit the differing expectations of the visitors;
- the contribution of the AML to the level of background noise of the museum is kept at a minimum.

Additionally the visitor should get the illusion that the source of the sound is always coming from the same location in space, although he moves his head. Therefore a tracking-sensor is integrated in the headphone, which measures the position and the direction of the visitor's head [3].

The exhibit features 33 sound sources, one of these is the voice of the visitor himself, and 14 acoustical spaces for selection and combination at will. The sounds come not only from the musical domain, such as pop and classical music but also come from common daily situations (such as noise or speech). In this catalogue, it is possible to find unique couples (sound, room) that appear to be intimately and ideally related. The visitor can come across these couples by combining and listening.

The rooms are divided in elementary and complex shapes; the visitor can project sounds in rooms like a cylinder, a sphere, etc... as well as in a church or in a roman theater. The basic rooms are scalable in one dimension, so that the visitor may, for example, control the length of a cylinder by a slider. While trimming the length of the cylinder, he can actually trim the resonant frequencies of the cylinder to the sound that he is listening to.

1.1. Catalogue of dry sounds

If the sounds presented within the AML had been recorded within a traditional studio the reverberation of which cannot be neglected, the museum visitor would hear rather the sound as it was in the recording studio than in the room that he has selected in the AML. That is why the sounds proposed within the AML have been recorded in a very dry room, anechoic whenever possible. We have recorded some of them ourselves (Poem, Drops...). The solo instruments were borrowed from the Archimedes Project [4] and the orchestral pieces were contributed by recording companies [5].

These sounds are considered within the AML to originate from a punctual monophonic sound source. This is convenient for most applications but it can lead nevertheless to paradoxical situations such as putting a whole orchestra within a shoe box or processing each instrument of the orchestra identically. An accurate simulation of the real acoustical situation in a concert hall would require that each instrument be processed differently according to its position and that of the listener. We have considered that this fact can be neglected within the AML in com-

parison to the large sonic differences that can be heard between one room type and another.

1.2. Catalogue of complex shapes

Whereas each room has its own acoustical signature, it is often difficult to identify a room solely from its reverberation pattern. A sound engineer or an acoustician could be better trained at this exercise than the usual visitor of the museum. In order to simplify the task of the visitor, we have selected a set of rooms which had to fulfill the following requirements:

- the reverberation should give the impression of being in a room of that kind;
- the number of rooms should be low so that the visitor can get an overview within a reasonably short period of time;
- the set of rooms should be representative of a wide range of applications;
- the acoustics of the rooms should be well differentiated from each other;
- the chosen buildings should be significant as far as their architecture is concerned.

A concert hall, a church, a drama theater, a roman theater, a natural landscape and an apartment were selected. The apartment was selected in order to give the visitor a point of reference to his daily life.

- **Concert hall:** the large symphony orchestra as we know it today was formed during the 19th century. It needed large spaces for its sounds and for the audience. The different instruments are to blend well but still be distinguishable and each seat in the audience should be reached by a balanced sound. We have chosen the Berliner Philharmonie (Fig 2).
- **Church:** the cathedral of Strasbourg was chosen but we could not yet measure this place. We have implemented the acoustics of the Saint-Stephan church in Karlsruhe instead (Fig 3).
- **Drama theater:** the Burgtheater in Vienna is very famous for the german repertoire, that is why we have selected it. We could not yet measure it. We have implemented instead the acoustics of a medium-size drama theater of Karlsruhe (Fig 6).
- **Natural landscape:** the museum wanted to present a landscape that would be beautiful but very difficult to reach for the public. We found in Mark Twain an accomplice in our search: he wrote a story about an American who developed the obsession of collecting echoes [6]. We "collected" the echo of the calanque En Vau on the coast of southern France. We had to reach this place by boat and, not before the dusk of a 6-day waiting time, we had

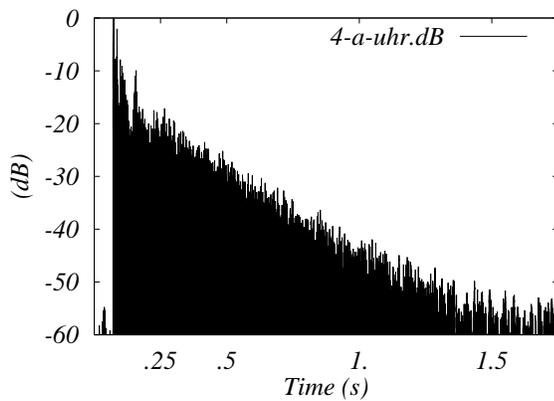


Figure 2: Echogram of the Berliner Philharmonie.

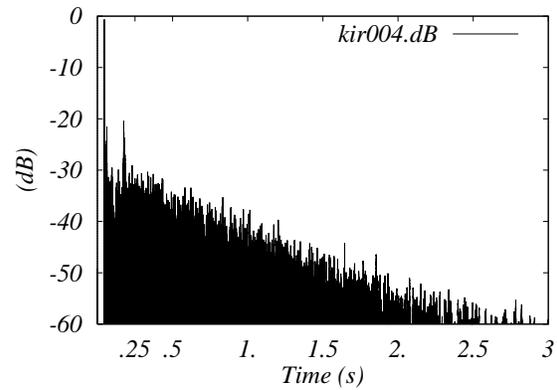


Figure 3: Echogram of the Saint-Stephan church.

the opportunity to make a valid measurement [7]. The calanque is about 100 m wide and 1 km long. It has a 100 m high cliff on the northern side, the southern side is neither as high nor as steep. We could measure an echo corresponding to the width of the calanque. This echo is blurred by the many rifts and wrinkles of the cliff that produce a diffuse reverberation (Fig 4 and 5). An interesting combination within the AML is with jazz music: it sounds as if the jazz orchestra were playing in an open-air venue in the distance.

- **Roman theater:** we selected the roman theater of Orange. This theater was built 2000 years ago in southern France. It is open air, with a high wall behind the stage, the audience sitting on rising stairs forming a half circle. We could measure a small echo for each step of the half circle. Since these individual echoes are as regularly spaced as the steps, they induce a characteristic coloration of the sound. The half circle sends the sounds back to the scene (Fig 7). The overall sound color is fairly aerial, it sounds "outside". The AML allows to check that the music of Debussy comes especially well in such a place.
- **Apartment**
In this apartment, three locations are proposed within the AML:
 - the **living room** because it is the most usual place. It has a fairly short reverberation time and might have discernible resonances because of the small size of the room (Fig 8);
 - the **bathroom** because it is well known that it is pleasant to sing in it. It has a fairly long reverberation and the high frequencies are little dampened because of the flat and hard boundaries (Fig 9);
 - the **staircase** because we figured out during

our experiments that it presents a very peculiar reverberation pattern. The steps work like diffusors and the sound source is often hidden. These peculiarities produces an extremely diffuse reverberation without any direct sound (Fig 10).

1.3. Catalogue of basic shapes

The basic shapes are meant for the visitors who wish to better understand how a volume can modify a sound. Beyond this analytical approach it is possible to use the basic shapes as tunable resonators and the exhibit turns out to be a place to perform elementary experiments with acoustics and instrument building. The basic shapes have one scalable parameter, so that, for example, the visitor can control the length of a cylinder by a slider. While trimming the length of the cylinder, he can actually trim the resonant frequencies according to the sound or to the music that he is listening to. The basic shapes had to comply with the following criteria:

- limited number of basic shapes (for the same reason as for the complex shapes);
- elementary volumes;
- distinctive resonance pattern;
- effective algorithm for real-time implementation;
- scalable in real time.

The following basic shapes have been selected and implemented:

- **Free Field:** when there are no walls, trees or other surfaces which may reflect sound, the sound will propagate further and further until its energy is dispersed. For our ears, there is no impression of a specific acoustical environment which would color the original sound (Fig 11).



Figure 4: The En Vau calanque.

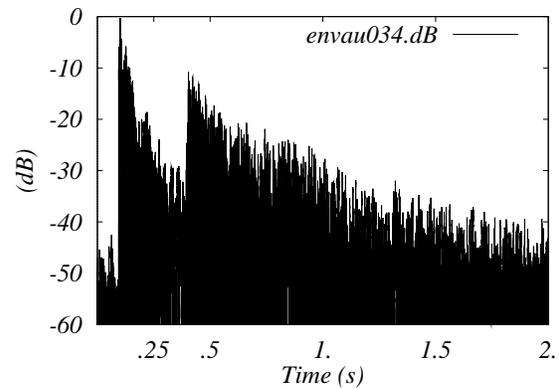


Figure 5: Echogram of the En Vau calanque.

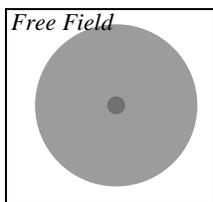


Figure 11:

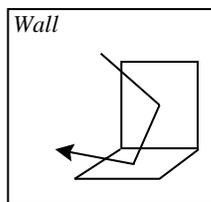


Figure 12:

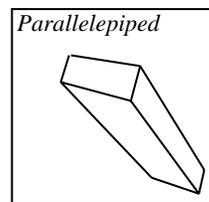


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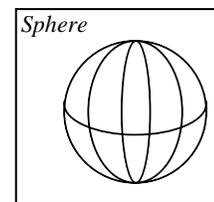


Figure 16:

- **Wall** in the middle of an open space: you shout towards the wall and a moment later you hear your voice coming back from the wall. The delay between your shouting and the echo depends on your distance to the wall (Fig 12).

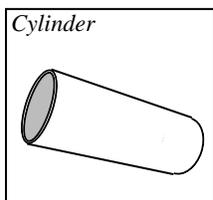


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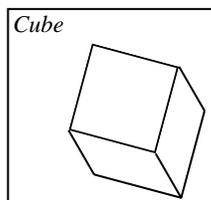


Figure 14:

- **Cylinder**: tubes, tunnels, or flutes have their own pitch which is determined by their length. If you listen to music through or in cylinders, those frequencies from within the music will be reinforced which coincide with the proper pitch of the cylindrical shape (Fig 13).
- **Cube**: all walls in a cube are in the same distance to each other. This results in an emphasized reinforcement of certain frequencies. The size of the cube plays an important role (Fig 14).
- **Parallelepiped**: in contrast to the cube, length, width and height are of different size. The ratio

of these three dimensions plus the size of the room determine the characteristic coloration a sound gets added by the room (Fig 15).

- **Sphere**: each sphere has a very characteristic sound which depends on its size. If you bounce a ball and listen closely to the resulting sound or if you put a balloon right in front of your ear, you can discover the characteristic property. It seldom happens to listen to sounds from within spheres but with multimedia projects such as the Extended Virtual Environment (EVE) from Jeffrey Shaw it could happen more often. The EVE is based on a large inflated lining that is used as projection screen. The volume of EVE is close to a sphere and several people can gather in it for a multimedia show [9]. The spherical form of this space poses severe acoustical problems for the musical performance. The spherical basic shape of the AML allows the museum visitors to make first experiments beforehand (Fig 16).

2. CHOOSING A SYSTEM FOR IMPLEMENTATION

Reverberation can be added to an acoustic signal by various means. The signal should be recorded almost anechoic. Several of the nowadays available techniques have been considered for the installation. The advantages and

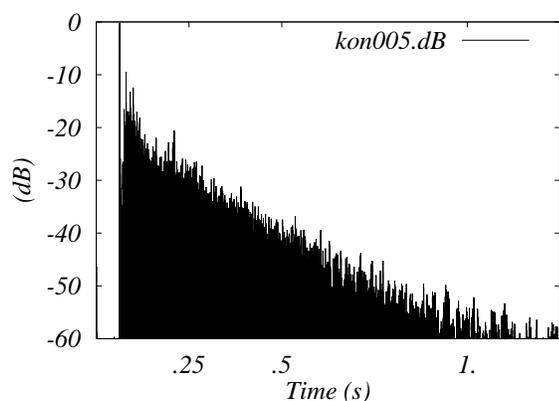


Figure 6: Echogram of a drama theater.

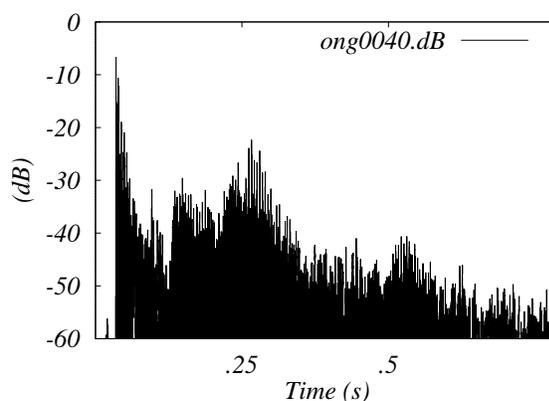


Figure 7: Echogram of the roman theater of Orange.

disadvantages of the approaches will be showed in a short description of a reverb system with recursive filters (IIR Infinite Impulse Response) and a system using filters with finite impulse response (FIR) involving the technique of fast convolution.

2.1. Feedback Delay Network

Digital recursive filter have been used for a long time to add reverberation to anechoic signals. The representation of such signal processing structures in computers can be done very efficiently. The computation load for such an elementary structure is very low, (one multiplication, one addition and a delay operation per sample). Sophisticated combinations of these elements can be built to produce a richer and natural reverberation sound [10, 11, 12, 13]. Usually the recursive filters will be modified and connected in matrices, the so-called Feedback Delay Networks, to smooth the artificial reverberation. Such a reverberation system has many parameters which must be adjusted to get a special reverberation.

2.2. Convolution

With non-recursive filters (FIR) another approach is used: the impulse response of a room has been measured, which means that the transfer function between a loudspeaker and a microphone is available as a set of FIR-coefficients. An alternative to the acoustic measurement of the room is to use the architectural design in a CAD program for room acoustics [14, 15]. The reverberation will be added to the input signal by convolving it (i.e. filtering it) with the impulse response of the room. This convolution is computationally expensive: for each signal sample the number of multiplications and additions that is required is equal to the number of samples of the room impulse response. The impulse response of a room is usually several seconds long, hence the corresponding FIR-filters are extremely long. With the technique of fast convolution,

some kind of multiplication in the frequency domain, it is possible to reduce the complexity of the computation to a large extent. The fast convolution algorithm makes use of the properties of the fast Fourier transformation (FFT) to achieve its efficiency. This advantage comes with a drawback: the latency of the system is proportional to the length of the FFT. A reverberation system using this technique for a room impulse response of 2 seconds would lead to a delay, from the input to the output, of 4 seconds. Real-time applications cannot cope with such a latency. With a refined technique it is possible to further exploit the peculiarities of the FFT algorithm and to reach a low latency [16, 17, 18, 19]. This technique balances the trade-off between the direct FIR implementation and the fast convolution. Different reverberation effects can be obtained by exchanging sets of coefficients.

2.3. Specifying and selecting a reverberation system

The requirements to the reverberation system are derived from the constraint that the exhibit must be interactive, that we need a development platform and from the overall costs. The fact that we want to allow a free combination of sounds and rooms implies that we need a real-time system. Since we also want to be able to process the voice of the visitor himself, the system must have a very low latency. The palette of rooms that are offered for selection ranges from churches to small living rooms and from complex shaped rooms to basic shapes. The reverberation system should be able to manage all these rooms. To achieve a tight integration and an easy-to-maintain exhibit, the whole system must use flexible components and rely on standard but versatile interfaces between the components

- real-time processing with low latency
- authenticity of the reverb for large and small rooms
- flexible development environment
- reasonable costs

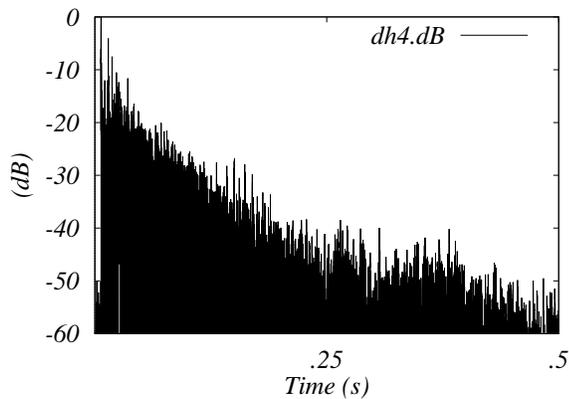


Figure 8: Echogram of a living room.

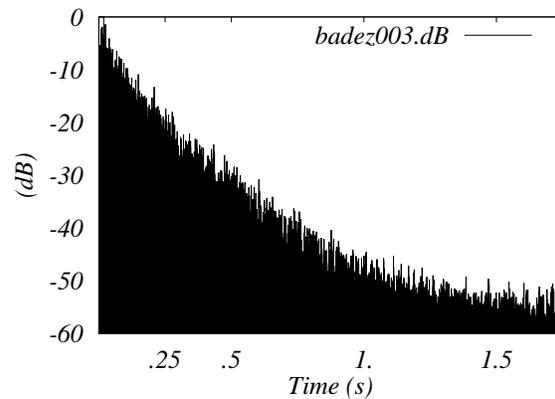


Figure 9: Echogram of a bathroom.

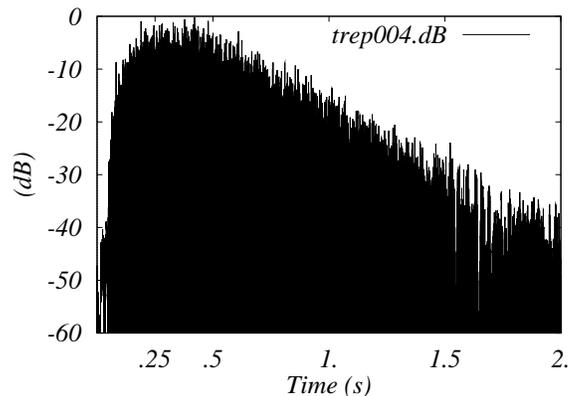


Figure 10: Echogram of a staircase.

2.4. Reverberation by convolution

An example of a system for reverberation by convolution is the Huron system from the Lake company. The system comprises a PC motherboard with an additional audio bus and a set of signal processing cards. With this system, room impulse responses up to a few seconds of length can be processed in real time with low latency. Such a system gives a maximum of naturalism. The production of a special room effect depends only on the quality of the measuring system or the simulation program for room acoustics. Several programs are available for room simulation [15, 20]. All of these programs have in common that they are designed for large rooms (concert halls, auditoriums, etc...) and they aim at predicting the acoustical quality of the planned room and to foresee the acoustic imperfections. Such predictive computations cannot be done in real time. These programs are also usually unsuitable for small rooms where modal acoustics plays a definitive role. The simulation programs usually fail at simulating acoustic effects like flutter echo. In order to get a room impulse response that shows a definitive flutter echo, it would be

necessary to do some kind of reverse engineering: sketch an artificial room in which the simulation program would exhibit a flutter echo.

Because of these facts and of its high cost, an auralisation system such as the Huron in combination with a room acoustics simulation program was not selected for our exhibit.

2.5. Reverberation by feedback delay networks

This type of reverberation system is often used in sound recording studios. These systems produce reverberated sounds that musicians and composers need in the process of recording or mixing music [21]. To simplify the use of the reverb device, the parameters of popular effects and room prototypes are stored in presets. The core of the system developed by Jean-Marc Jot, Espaces Nouveaux and IRCAM (Institut de Recherche et Coordination Acoustique/Musique) [12], comprises also a feedback delay network. The program package Spatialisateur works under Max on a NeXT-Cube with an ISPW-board [22, 23]. The entire system is controlled by two

sets of parameters. The so-called low-level-parameters allow to determine special time frames, energy distributions and frequency bands inside the impulse response. Through a kind of orthogonal transformation, the low-level-parameters are related to the high-level-parameters, which can be adjusted by the user to control subjective parameters such as distance, warmth, etc... The high-level-parameters were derived from studies about the assessment of room acoustics, from a perceptive point of view [24] (Table 1).

Table 1: Conceptual comparison of the approaches for reverberation algorithms.

Scientific	Perceptive
Simulation	Imitation
Off-line process	On-line process
Accurate	Approximative
Intellectual	Emotional

2.6. Choice of a system

For our installation we have decided to use the Spatialisateur running on a NeXT-Cube with ISPW. Such a system is more suitable to the purpose of the exhibit, because it relies on the habits and particularities of the auditory perception instead of dealing with the rigid simulation of a predefined room. The aim of the exhibit is actually to let the visitor hear distinct differences in the room acoustics instead of presenting simulations where only trained listeners can notice differences. Another reason for the decision is the real time processing capability of the system and the straightforward integration into the exhibit environment.

3. MEASUREMENT, SELECTION AND IMPLEMENTATION OF THE COMPLEX SHAPES

Most people have in mind some large rooms showing a lot of reverberation but few of them would be able to describe this reverberation precisely and identify the room only from its reverberation pattern. That even lay people in acoustics should be able to recognize a room solely from its reverberation is an important issue for the museum. We had to define a method to measure and evaluate rooms before implementing them into the exhibit.

3.1. MLS-Measurements

A practical way of measuring room impulse responses is the method known as MLS (Maximum Length Sequence) where a pseudo random noise is played in the room through a loudspeaker and the sound pressure in the room is picked

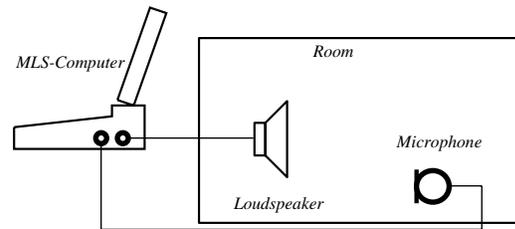


Figure 17: Experimental setup for room impulse measurements.

up by a microphone. A computer compares the signal that has been sent to the loudspeaker to the signal measured by the microphone using an algorithm called cross-correlation. The output of this operation is an impulse response that characterizes the room for the unique set of points where loudspeaker and microphone were placed. If these transducers are setup at other places then the impulse responses will be slightly different [25] (Fig 17).

In order to take equally into account all directions of propagation in the room, the transducers are supposed to be omnidirectional at any frequency. It is generally agreed that a measuring bandwidth of 5 to 10 kHz is sufficient for room acoustics measurements but we have stuck to the largest possible bandwidth, say up to 20 kHz, for several reasons:

- when dry signals are convolved with the impulse responses, a 5 kHz or 10 kHz lowpass filter effect would have been clearly audible;
- many subtle details of the reverberation pattern ask for a high time-resolution hence a large frequency-bandwidth;
- the analyses of the impulse responses in order to derive the parameters of the Spatialisateur are more reliable if the full audio bandwidth is available.

The transducers that we have chosen are a 1" studio-grade condenser microphone that features a low selfnoise and a flat free-field frequency response and a compact self powered tow-way studio monitor. Both devices are capable of reproducing clear transients with a bandwidth of up to 20 kHz. The directivity pattern of the loudspeaker is far from ideal, as we will notice it afterwards, but no information was available at the time when we performed the measurements. We are still looking for an alternative sound source for future measurements but the sources that are traditionally used for room acoustics measurements, such as dodecahedra, do not reproduce clear transients and have a reduced frequency bandwidth.

The MLS system that we have used allows us to reach a signal to noise ratio of about 50 to 60 dB, as can be es-

timated from the echograms, and it is enough for our application but it is too limited for studio applications were the convolution of a dry sound with a room impulse response should not add any perceivable background noise. The good performance of the MLS system could not be reached with a straightforward measurement because it leads to objectionable spurious echos. In order to remove these artifacts it is necessary to optimize the measurement by tuning the output level of the noise generator and by measuring twice, playing both the direct noise sequence and its reversed-time version.

Even though MLS measurements are feasible even under very poor signal to noise ratios, the higher the signal to noise ratio the better the results. Although this is usually not a problem in closed spaces it is a big issue for outdoor measurements such as for the roman theater or the En Vau creek. If there is no ceiling to reflect acoustic energy back to the ground, then it is useless to send energy in that direction. Hence in order to concentrate somewhat the available energy in the horizontal plane, we have used two identical loudspeakers above one another and upside down in order to keep the high-frequency transducers close to each other. The background noise is not the only annoyance that is found outdoors. Wind and temperature gradients play an important role to modify the propagation of sound waves. It is a good practice to average several MLS measurements in order to level out some of the periodical disturbances but the periods of wind and temperature variations are so large that it is impracticable for these perturbations [26, 27]. At both places Orange and En Vau we had a long trial and error period where we have been experimenting with various setups and positions until favorable conditions all of a sudden were gathered: no wind, small temperature gradients, low background noise and no waves in the creek. These rare favorable situations happened usually at dusk.

3.2. Selection of a measurement

Selecting a measurement is a crucial step before implementing a complex shape in the AML. The selection will be performed by listening to dry sounds that have been convolved by the measured impulse responses. Before listening to the impulse responses it is important to have a clear, although non traceable, mental representation of how the room sounds. This representation can only be gained by listening beforehand in the room to the same dry sounds. So the procedure for measuring is actually: first listen to dry sounds in the room, wander in the room to spot the audible particularities and to select the positions to be measured, then measure systematically.

As well as the quality of the measurements is dependant on the microphone and on the loudspeaker used, the selection of a measurement calls for a faithful headphone. We have chosen diffuse-field equalised dynamic

headphones, similar to those that would be used later on within the exhibit.

In order to compare several impulse responses a small set of short sounds is first selected. These sounds are chosen to reveal as many characters as possible (coloration, echos, clarity...). These sounds are convolved with the set of impulse responses and a (Sound x Impulse) catalogue is the result of this operation. Listening to this catalogue and comparing to the mental representation of the room reveals that only a few impulse responses are eligible out of the set of 20 to 40 measured impulse responses. A best-choice impulse response can eventually be selected that evokes at best the room. The person in charge of the selection had sometimes to ask for advice from colleagues, but the decision was most of the times astonishingly stable.

3.3. Adaptation to the installation

The selected impulse responses had to fulfil the following requirements:

- sound typical for the chosen type (such as church, bathroom etc...);
- be different enough from others so that lay people can differentiate them;
- be well balanced.

Once an impulse response has been retained for each room, a catalogue has been produced and each sound has been listened to carefully in order to judge whether all 3 criteria were met. If necessary corrections were made (select an alternative impulse, adjust the gain, adjust the fade out of the impulse before the noise becomes too disturbing...) and after a few evaluation cycles we could conclude this evaluation. The selected set of impulse responses has been sent to the acousticians of IRCAM (Jean-Marc Jot, Olivier Warusfel and Laurent Cerveau) who have computed for us the parameters for the Spatialisateur [28]. We have done some fine tuning of these parameters before integration in the exhibit but the initial values were pretty good. The room that is the most difficult to render is the staircase because the Spatialisateur cannot build up the diffuse field at the required speed.

4. DESIGN AND IMPLEMENTATION OF THE BASIC SHAPES

For the implementation of the basic shapes we rely on lighter models than for the complex shapes. The acoustic properties of small rooms are easier described by modal acoustics than by other methods such as geometrical acoustics. Therefore resonator implementations are better suited to illustrate these rooms. The models are also programmed in MAX-FTS and they use some elements from the Spatialisateur, but it turned out to be easier to keep them independent from the Spatialisateur and

switch from one system to the other according to the requirements of the visitor. Whereas models have been described in the literature for most of our basic shapes, we had to design an original model for the sphere.

- **Free Field:** is implemented by playing the dry-recorded sound through a module that implements the geometric attenuation and the air absorption (the `air~` module of the Spatialisateur [8] and Fig 11 and 18).

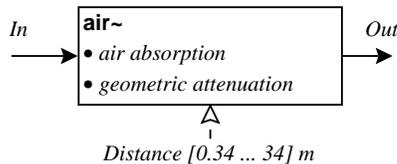


Figure 18: Implementation of the basic shape "Free Field".

- **Wall:** to implement this basic shape, we should add a delayed version to the dry signal, take into account the frequency dependent attenuation due to the air and to the reflection on the boundary as well as the geometric attenuation due to the distance. The latter effect can be very important for larger distances and prevent the visitor from perceiving the other subtler ones. Depending on the shape of the reflecting boundary, this geometric attenuation can also vary very much.

To simplify, we have decided not to implement the geometric attenuation. This way, the level of the echo is fairly independent of its duration and the visitor can clearly follow the spectral and temporal effects of the echo. For distances smaller than about 3.4 m, the visitor will hear the effect as a coloration of the dry sound whereas for distances larger than about 8.5 m he will clearly perceive the echo as a delayed version of the dry signal. For settings in between, he will hear a slapback effect where each impulsive sound is heard as a set of 2 impulses that are very close to each other. To make the distinction between dry sound and echoed sound more obvious, a frequency-dependant reflection coefficient is affected to the reflecting wall (Fig 12 and 19).

- **Cylinder:** we chose to present cylinders that all have the same shape factor which relates the diameter to the length. Therefore the slider of the AML control desk controls simultaneously the length and the diameter of the cylinder. The longer the cylinder, the larger the diameter. The delay-line duration is proportional to the length of the cylinder and the dissipation at high frequencies grows as

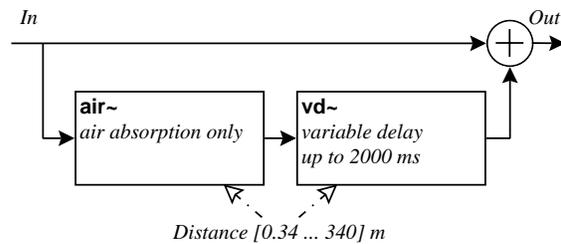


Figure 19: Implementation of the basic shape "Wall".

the length and the diameter are increased. This last feature ensures that the coloration due to the cylinder is balanced through the range of settings and the visitor does not get the impression that the cylinder is switching from one type of resonator to another one. The cylinder is implemented as a recursive comb filter with a first order lowpass filter in the feedback loop [10, 29]. The output to the headphones is picked up after the delay line. The feedback coefficient is fairly high so that the effect is obvious for the museum visitors (Fig 13 and 20). An interesting combination within the AML is with the guitar musical excerpt. The sounds of the guitar reveal the individual resonances of the cylinder and the cylinder can be used to build an imaginary guitar with, for example, very low-frequency resonances.

- **Cube:** is imitated using a structure similar to that of the cylinder but three delay lines are used instead of a single one. The first delay line is tuned to the size A of the cube, the second to the diagonal of a face of the cube $\sqrt{2}A$ and the third to the diagonal between two opposite corners of the cube $\sqrt{3}A$. This model might seem oversimplified but we estimated that it is accurate enough for the purpose of the AML. We could convince ourselves about its relevance by listening to the resonances of an empty cubical room of the ZKM-building as it still showed its bare concrete boundaries (Fig 14 and 22).
- **Parallelepiped:** this basic shape is imitated again with a similar signal processing structure but with six delay lines. As well as the cylinder the shape factor is fixed. We have chosen proportions according to the gold number and its square: Height x Width x Length are in the proportions of 1 x 1.618 x 2.618. Three delay lines implement the reflections corresponding to the sides of the parallelepiped, three other ones implement those corresponding to the diagonals. This elementary model could be much improved but it was selected because it sounds much different from other basic

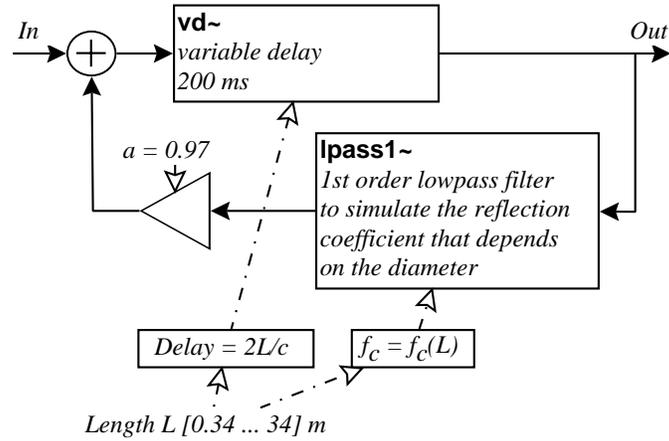


Figure 20: Implementation of the basic shape "Cylinder".

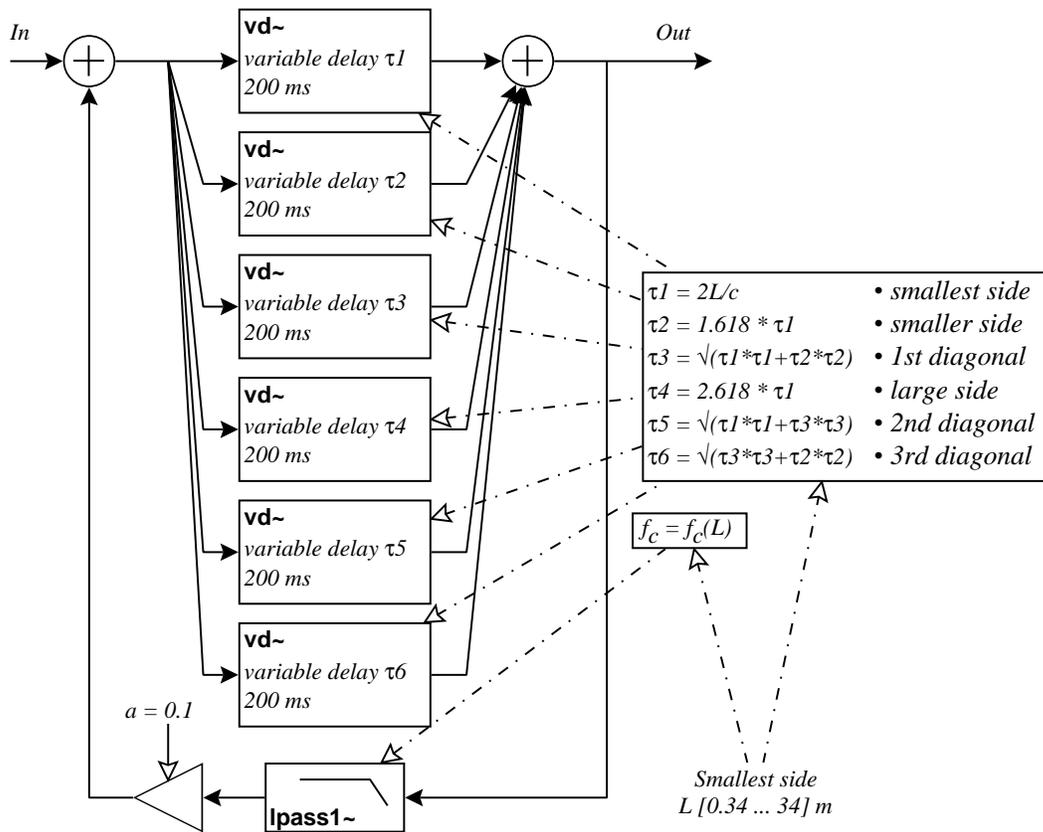


Figure 21: Implementation of the basic shape "Parallelepiped".

shapes of the AML and it seems to be realistic compared to the large wine tank that we had a chance to listen to. This tank made of concrete was a par-allelepiped and had a stainless-steel inside-coating (Fig 15 and 21).

- **Sphere:** this basic shape could not be implemented with algorithms similar to those of the previous shapes because the modes of a sphere are not harmonically spaced but related to the roots of the derivative of the spherical Bessel functions [30]. By considering the distribution of the roots we came to a trade-off where some subsets of roots are implemented within modified recursive comb filters and where the lower resonant frequencies are implemented using high-Q parametric equalisers (Fig 16 and 23). This model is again far from perfect but it is well differentiated from the other basic shapes and it reminds of the sound of a sphere. Our reference sound was that given by a plastic ball having a 0.65 m diameter. Listening to AML-spheres having larger diameters, e.g. a few meters, is not so convincing but it gives nevertheless the impression that the boundaries are curved, which is certainly not the case for the other basic shapes.

5. BINAURAL PRESENTATION

In the exhibition space of the MediaMuseum the AML is not the only exhibit and, from the standpoint of concept and design, a certain privacy should exist around each of the installations. The fact that the AML is an installation that focus on the sound media gave us a hint how to differentiate it from other exhibits. There is a dialectic here between sound and image. Whereas most exhibits in the museum are dealing with the image media and are enclosed in dark booths, the AML, dealing with the music media, could stand for itself in the exhibition space. As well as the exhibits with video are protected from intense ambient light and respect the visual privacy of the visitors, the AML with its sounds should be protected from ambient sound and respect the sonic privacy of the visitors. The main exhibition area of the museum is situated around a wide inner court with a glass roof. Due to the unfortunate reverberation characteristics of this place, we chose a headphone sound presentation. This way, the influence of the background noise is reduced as far as possible and no AML-sound interfere with those of the other exhibits.

One drawback of the headphone solution is that the AML-visitors are not immersed in a common sound field like those that can be created using a multichannel loud-speaker approach such as the CyberStage [31]. Nevertheless to enhance the binaural spatial sound presentation of the AML we have integrated a head tracking system into the ISPW sound processing environment.

The cable-based magnetic head-tracking system consists of a transmitter mounted on the corpus of the exhibit and a receiver mounted on one of the headphones. The position of the transmitter defines the origine of a cartesian space where 6 degrees of freedom (x,y,z, azimuth, elevation and roll) are provided. The spatial sound processor Spatialisateur [28] can process this information to create a virtual binaural auditory scene, i.e. the movements of the listener's head are compensated in order to keep the sound source at a fixed position within the virtual scene. The position and orientation of the visitor's head is measured every 20 msec and is sent by way of a serial computer connection to the signal processor. This moderate update rate is a trade-off between lively interactivity and manageable sound processing complexity.

6. CONCLUSION

The exhibit has been in operation for more than a year now and many users are very fond of it because they enjoy listening to music in different acoustical environments and they can experience the relation between sounds and performance space. The room selection is such, that most people can recognize or identify a space after listening to a few sounds. Visitors, musicians and even acousticians can learn intuitively how a piece of music sounds in a place or in another. Music students are fascinated to hear how a room can modify the sound of their instrument. The basic shapes allow to demonstrate some modal acoustics.

The Spatialisateur has proven to be very effective at rendering most complex shapes but it fails at providing a convincing imitation of the staircase with its challenging diffuse sound field. People who did not visit the exhibit ask sometimes why we did not use an architectural and auralisation CAD program. One of the reasons lies in the limited audio quality of these systems because computer synthesized room impulse responses do not sound as convincing as measured ones, even after adaptation to the Spatialisateur. Things might improve in the future but at the time when we made our decision, the difference of quality was quite perceivable. Other people ask whether we could not have done our work with of-the-shelf audio reverberators. These devices are usually designed to enhance a piece of music and not to add objectionable echos or coloration. In our project on the contrary, we wanted to underline the acoustical peculiarities of the rooms even if sound engineers would have preferred to get rid of them. Whereas it is cumbersome to modify usual audio reverberators, the Spatialisateur could be trimmed to fit our special requirements.

The installation is unfortunately not yet complete. We are still planning to measure and integrate the Strasbourg cathedral and the Vienna Burgtheater. We would like also to extend the catalogue of complex shapes with the

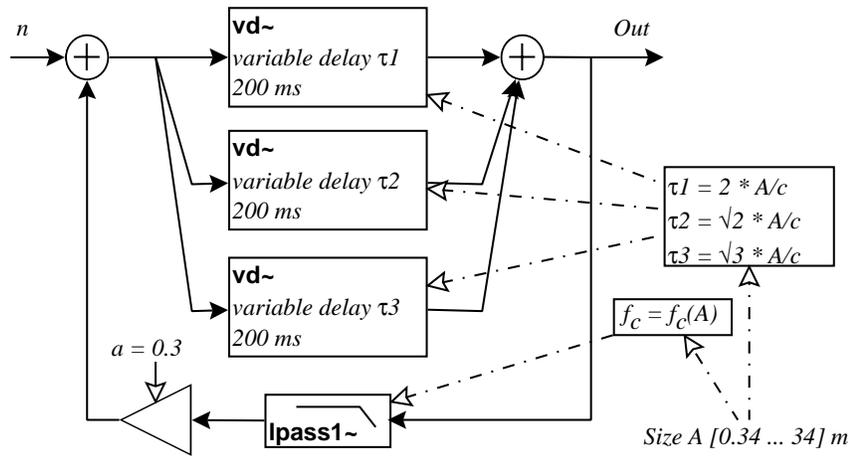


Figure 22: Implementation of the basic shape "Cube".

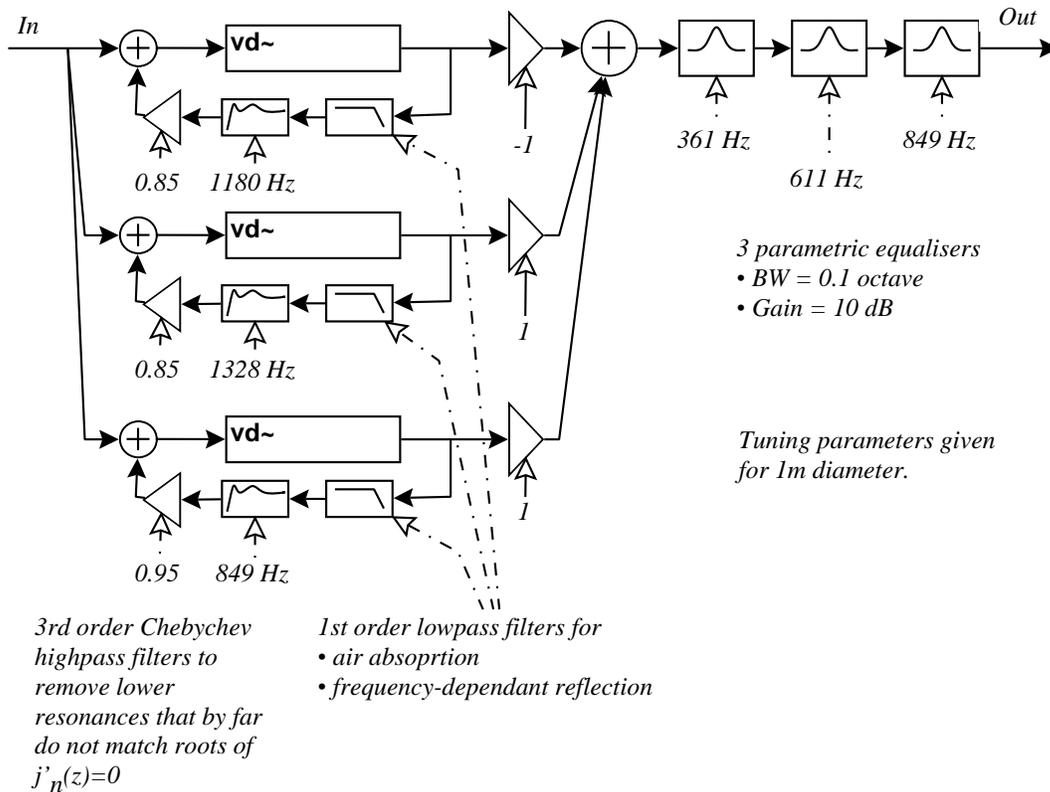


Figure 23: Implementation of the basic shape "Sphere".

EVE, the inside of a car and a birch forest. These two last acoustical places are interesting because people listen to music more often in their car than anywhere else and because a birch forest has a unique reverberation pattern where each tree sends a small echo back to the sound source.

We would like to find a more accurate implementation for the sphere as well as to refine the implementations of the cube and of the parallelepiped.

From the detailed analysis of the measurements made at the Saint-Stephan church and of additional measurements made in an anechoic room we know that the sound source that we have used for our measurements is far from perfect. It is too much directive at higher frequencies. We have investigated a dodecahedra having 12 woofers and 20 tweeters. The preliminary conclusion is that this dodecahedra acts more like a diffusor than like a punctual sound source. We are still looking for a better system.

John Chowning has shown that controlling the ratio between the direct and the reverberated sounds allows to give a cue of the distance between the source and the listener [32, 28]. We have not yet tuned this feature for the complex shapes but when it is operating, the museum visitors will be able to control this cue using the slider of the AML-control desk.

Programs for low-latency convolutions have been developed during this project and have been used intensively for the evaluation and the selection of the room impulse responses before integration in the exhibit [33]. These programs find further applications by composers who wish to integrate natural sounding reverberation within their computer music compositions [34].

Although all measurements and evaluations have been performed monophonically, the mono sounds give back an astonishing impression of space and depth. Measuring and processing binaurally would allow us to reach another degree of accuracy and liveliness. Our strongest impression during this project might have been to listen to dry sounds in the calanque of En Vau. We heard a strong and clear echo coming perpendicularly from the cliff and afterwards the many other echoes decreasing in amplitude originating from the many rifts and wrinkles of the cliff. The cliff gave from each impulse an answer that was initially compact but that was spreading to both sides as the time went forth. It would have been necessary to perform binaural measurements in order to capture such an effect.

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