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Advanced techniques for measuring and reproducing spatial sound properties of auditoria

Angelo Farina[†], Lamberto Tronchin[††]

† Industrial Engineering Dept. University of Parma,
via delle Scienze 181/A, 43100 Parma, Italy
Email address: angelo.farina@unipr.it

†† Dept. of Energetic, Nuclear and Environmental Control Engineering, University of Bologna,
Viale Risorgimento 2, 40136 Bologna, Italy
Email address: lamberto.tronchin@unibo.it

ABSTRACT

The definition and measurement of sound spatialisation have been strongly enhanced in last years, as nowadays spatialisation is considered quite important during design of auditoria and virtual audio reproduction of sound quality in dedicated listening rooms for 3D reproduction purposes. Even though international standards like ISO 3382 require measuring some spatial parameters (i.e. LE, LF, IACC), usually only binaural measurements are performed, by means of a dummy head, and rarely 3D impulse responses are measured and utilised for sound reproduction.

In this paper, an innovative procedure of measuring spatial sound characteristics is presented. The application of this new technique in virtual 3D sound reconstruction is presented. Furthermore, the methodology is compared with other techniques of 3D sound reproduction. Moreover, the results of a wide campaign of measurements of spatial parameters among different auditoria all over the world, ranging from Italy to Japan and Australia, and conducted with the novel methodology, are compared with the results of standard binaural and 3D measurements. The possibility to enhance the spatial reproduction of sound quality in real spaces and the comprehensibility of spatial parameters is finally considered and presented in different cases.

KEYWORDS : Measurements, Auralization

INTRODUCTION

When the famous and renowned Gran Teatro La Fenice in Venice burned during the night of 29 January 1996, one of the best sounding opera houses in the world suddenly disappeared. Its sonic behaviour, however, was at least partially saved, because several acoustical measurements had been performed just two months before, employing the binaural impulse response technique [1].

M.Gerzon [2] first proposed to start a systematic collection of 3D impulse responses measured in ancient theatres and concert halls, to assess their acoustical behaviour and preserve them for posterity. His proposal found sympathetic response only very recently, with the publication of the "Charta of Ferrara" [3] and the birth of an international group of researchers who agreed on the experimental methodology for collecting these measurements [4].

The focus of this paper is devoted to the usage of these advanced three-dimensional impulse responses for creating truly realistic audible reconstructions (auralization).

A new measurement method is described [5], which incorporates all the previously known measurement techniques in a single, coherent approach: three different microphones are mounted on a rotating boom (a binaural dummy head, a pair of cardioids in ORTF configuration, and a Soundfield microphone), and a set of impulse responses are measured at each angular position. Fig. 1 shows a schematic of this microphone setup.

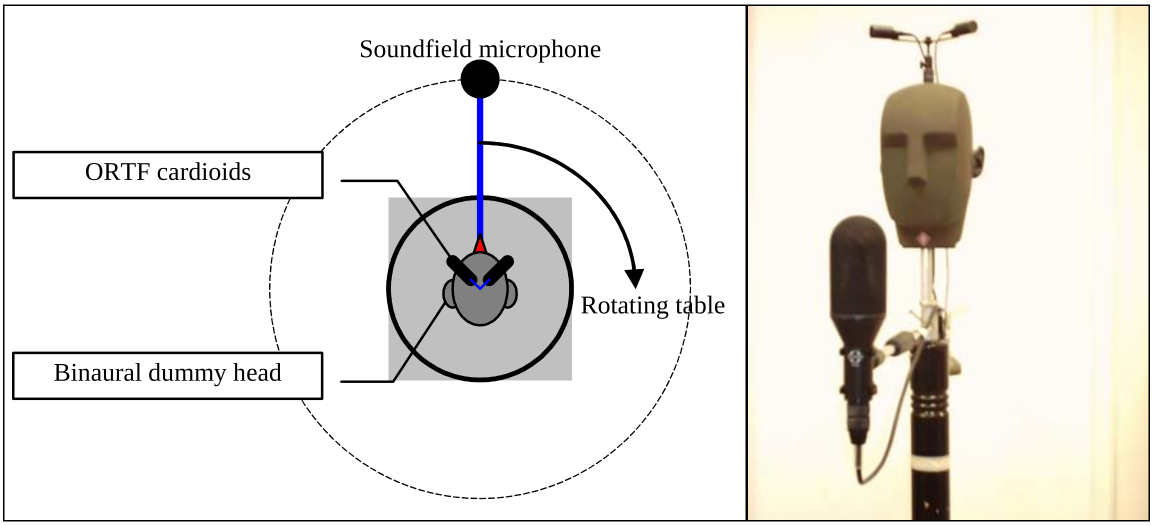


Figure 1: Scheme of microphones.

The results of this set of measurements are compatible with the already proposed methods for measurements in concert halls (binaural, B-format and WFS), but add the possibility to derive "standard surround" formats such as OCT and INA, and open the possibility to employ even the Binaural Room Scanning method [6], the Poletti high-order circular microphones [7], the Wave Field Synthesis Approach [8] and the Ambiophonics hybrid approach [9].

The paper describes briefly the measurement results, and more in detail the process of employing the experimental impulse responses for advanced method of auralization, aimed to the realistic reproduction of the spatial properties of the original soundfield.

MEASUREMENT OF MULTICHANNEL IMPULSE RESPONSES

This chapter describes the details of the measurement method, the equipment (hardware and software), and the procedure.
Although most of these items are not inherently new, the combination of them in a coherent approach provides a general method from which all known multichannel formats can be derived.

Test signal and deconvolution

The excitation-deconvolution technique employed for the measurement of the impulse response is the log sine sweep method, as initially suggested by one of the authors [10].
A good compromise between measured frequency range, length of the sweep and signal-to-noise ratio has been reached, by choosing the following parameters:

Start frequency	22 Hz
End frequency	22 kHz
Length of the sweep	15 s
Silence between sweeps	10 s
Sweep type	Exponential

The “unusual” length of the silence between sweeps is due to the traveling time of the rotating table. The rotation is triggered by a proper pulsive signal, automatically generated in the middle of the silence gap on the second channel of the sound card.

The choice of the above parameters allows the measurement of impulse responses, which have wide frequency span, good dynamic range (approximately 90 dB) and are substantially immune from background noise eventually present during the measurements.

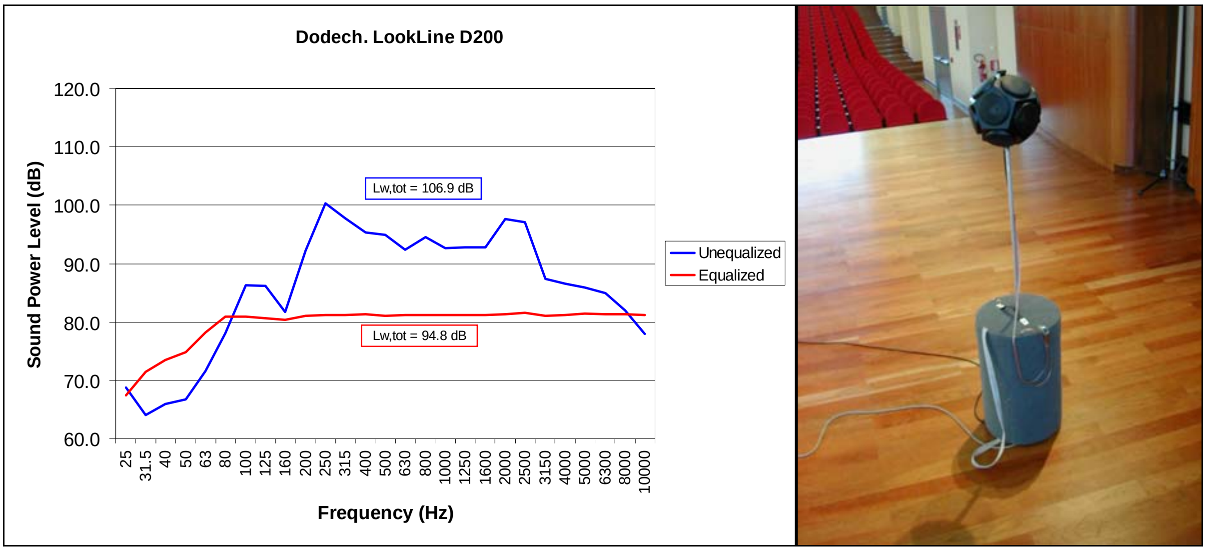


Figure 2: Dodechaedron loudspeaker and subwoofer.

The sound source

An omnidirectional sound source is usually preferred for measurements of room impulse responses. Albeit this does not correspond to the effective directivity pattern of real-world sound sources (such as musical instruments or human talkers and singers), the usage of an omnidirectional sound source is predicated by current standards (ISO3382, for example), and avoids exploiting strange room effects, as can happen employing highly directive loudspeakers (abnormal energization of echoes and focalizations for selected orientations of the source).

A special, ultra-compact dodechaedron loudspeaker was built specifically for the purpose of this research, employing 12 full-range drivers installed on a small size enclosure (approx. diameter is 200 mm). This unit, of course, is not capable of producing significant acoustical power under 120 Hz; for extending the low frequency range a subwoofer was added, incorporating it inside the cylindrical transportation case, which also contains the power amplifier (300 W RMS) and serves as supporting base for the dodechaedron. A small Italian company, LookLine, specialized in high quality, custom-built loudspeaker systems, built the doedechaedron.

The microphones

Three different microphonic probes were employed:

- a pair of high quality cardioids in ORTF configuration (Neumann K-140, spaced 180mm and diverging by 110°);
- a binaural dummy head (Neumann KU-100);
- a B-format 4-channels pressure-velocity probe (Soundfield ST-250).

All these microphones were installed over a rotating table, in such a way that the rotation center passed through the center of the dummy head, and through the point at the intersection of the axes of the two cardioids (which were mounted just above the dummy head). Alternatively the Soundfield microphone was displaced exactly 1m from the rotation axis, in front of the dummy head. The angular step was 10° (36 measurements for the complete rotation)

Measured data

At the time of writing, 12 famous theaters were measured with the previously described method, as reported in the following table.

N.	Theatre	N. sources/receivers
1	Uhara Hall, Kobe, Japan	2/2
2	Noh Drama Theater, Kobe, Japan	2/2
3	Kirishima Concert Hall, Kirishima, Japan	3/3
4	Greek Theater in Siracusa, Italy	2/1
5	Greek-Roman Theater in Taormina, Italy	3/2
6	Auditorium of Parma, Italy	3/3
7	Auditorium of Rome (Sala 700), Italy	3/2
8	Auditorium of Rome (Sala 1200), Italy	3/3
9	Auditorium of Rome (Sala 2700), Italy	3/5
10	Sydney Opera House (opera theater), Australia	4/3
11	Sydney Opera House (concert hall), Australia	3/3
12	Sydney Opera House (the studio), Australia	3/1

However the number of rooms being measured is increasing quickly, and it is planned to reach at least 30 different rooms in less than 6 months.

The goal of this paper is not to present a comprehensive comparative study of the measured data, which will follow when the collection of impulse response responses is complete, and all the results are fully analyzed. However, the following picture shows a quick comparison of the reverberation times of the above theatres.

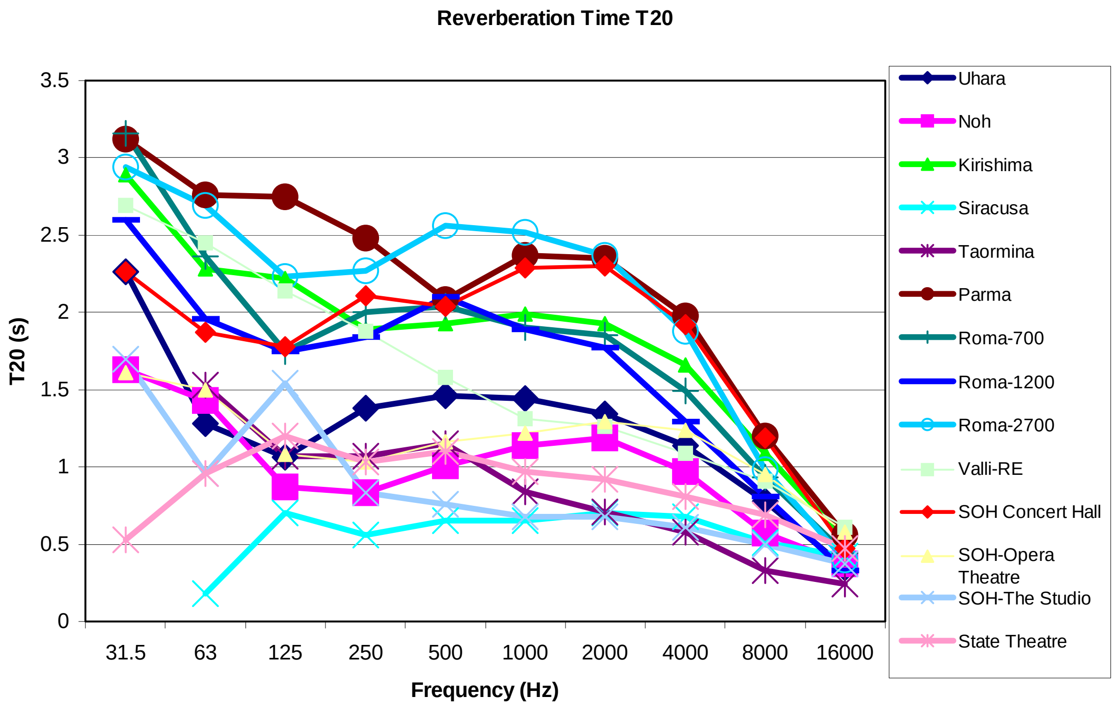


Figure 3: Reverberation times of the measured rooms.

METHODS FOR AURALIZATION

The basic method for auralization is convolution: the impulse responses are employed as very long FIR filters, applied to dry (anechoic) recordings of music or speech. Convolution is a very efficient filtering technique, particularly if implemented with proper (old) algorithms on fast (new) processors: as clearly demonstrated in [11].

In the following it is described how to create sets of impulse responses suitable for being employed by these software convolvers, creating the results in any of the currently available formats suitable for multichannel reproduction, and attempting to recreate as faithfully as possible the spatial attributes of the original soundfield.

ORTF-stereo impulse responses

This is the most basic processing, aimed at the creation of a "standard" stereo presentation of the results of the auralization. The process is based on the availability of a number of dry mono recordings, one for each section of the orchestra or for each singer.

Each mono recording has to be convolved with a specific stereo impulse response, obtained by the pair of cardioid microphones in ORTF configuration. In principle, each of these impulse responses should be measured with the proper position of the sound source.

Finally, the results of the convolution of all the dry recordings are summed in a single stereo output file, which is suitable for reproduction in a normal stereo system (2-loudspeakers).

Binaural impulse responses (binaural room scanning)

The basic binaural approach is substantially the same as for the previous ORTF-based method, but employing the binaural IRs. This way, the result of the convolution is a 2-channels file, suitable for headphone reproduction.

However, two methods can be employed for substantially improving the surround effect obtained: for loudspeaker reproduction a proper cross-talk cancellation must be added, and for headphone reproduction an head-tracking sensor can drive a real-time convolver, switching the impulse responses being convolved as the listener rotates his head.

B-format impulse responses (Ambisonics)

In this case, each dry mono source is convolved with the proper B-format impulse response. So, after the mixing of all these convolutions, a 4-channels B-format output is obtained.

The reproduction of a B-format signal over a suitable array of loudspeakers requires an Ambisonics decoder, for computing the proper feed for each speaker.

The creation of a software-based decoder has been pioneered by one of the authors [10], and has been further perfected by colleagues at the University of York, who recently released for free a suite of VST plugins [11], allowing for manipulation and decoding of B-format signals over various loudspeaker rigs.

ITU 5.1 surround (from selected B-format impulse responses)

The basic approach for ITU 5.1 rendering is to first select a configuration of microphones to be employed, for driving the 5 main loudspeakers [12]. Many of these microphone arrangements have been proposed, and in a recent round-robin project, called the Verdi project, most of them were comparatively evaluated [13].

Here we consider just two of them, which got good results in the aforementioned comparative test: Williams MMA [14] and OCT [12].

The following pictures show the microphone configurations for these two setups:

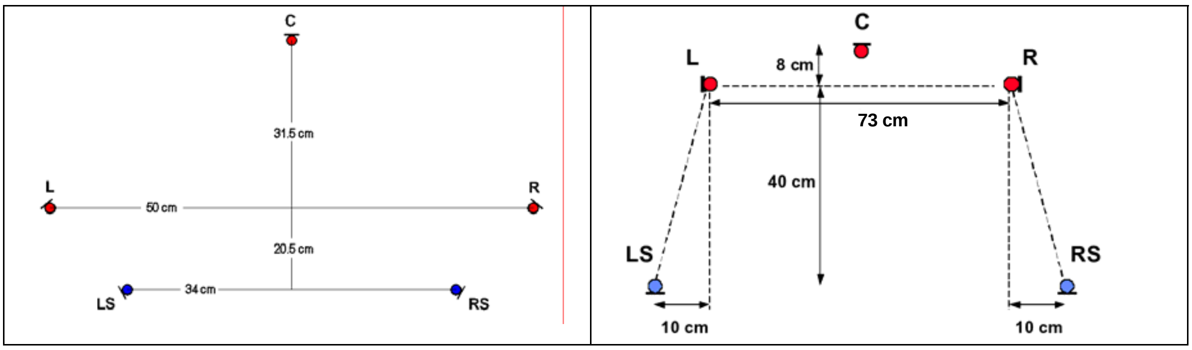


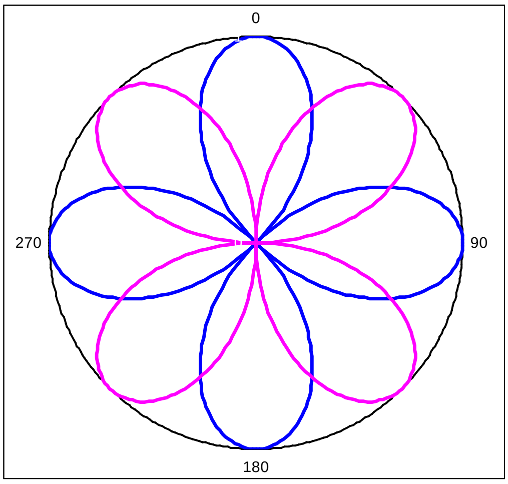
Figure 4: Layout of 5 microphones (Williams MMA, OCT)

For each of the above setups, it is possible to select a subset of 5 of the 36 positions where the Soundfield microphone was displaced, corresponding as close as possible to the intended positions of the chosen setup. Then, from the B.format impulse response measured in each of these 5 selected positions, a single (mono) impulse response is extracted, thanks to the program Visual Virtual Microphone, developed by David McGriffy and freely available on

the Internet [15].
Finally, each mono dry source is convolved with the 5-channels impulse response derived from the corresponding sound source position over the stage, and the results of all these convolutions are mixed in a single final 5-channels track, which is suitable for reproduction over a standard ITU loudspeaker rig.

Mark Poletti's high-directivity virtual microphones

During the rotation of the microphonic assembly, the two cardioids employed for ORTF recordings also describe a small circumference, with a radius of approximately 110 mm,
Looking for simplicity to just one of the two microphones, it samples 36 impulse responses during its complete rotation. From this set of data, it is possible to derive the responses of a set of various-orders coincident microphones, ideally placed in the center of rotation, making use of a modified version of the Poletti's theory [7].
The basis of this method is to define a class of multileaf-shaped horizontal directivity patterns of various orders. The order 0 is an omnidirectional, order 1 are two crossed figure-of-eight microphones (as in horizontal-only Ambisonics); then order 2 and 3 are added, with directivity patterns corresponding respectively to the cosine of twice and three times the angle:



$$D_0 = 1$$

$$D_{1,n} = \cos\left(\vartheta + n \cdot \frac{\pi}{2}\right) \qquad n = 0,1$$

$$D_{2,n} = \cos\left(2 \cdot \vartheta + n \cdot \frac{\pi}{2}\right) \qquad n = 0,1$$

$$D_{3,n} = \cos\left(3 \cdot \vartheta + n \cdot \frac{\pi}{2}\right) \qquad n = 0,1$$

Figure 5: Poletti's theory: concept (left, 2nd order) and formulas (right)

Form these, and advanced high-order Ambisonics decoder (horizontal-only) can be fed.
A second possible way of employing these high-order signals is to drive a standard 5.1 ITU array, by synthesizing 5 proper asymmetrical directivity patterns, as suggested in [27].

Circular WFS approach

The 36 B-format measurements made along the 1m-radius circumference are exactly the set of data required for employing the WFS method described in [6].
The basis of this method is the Huygens principle. The WFS is a 2D reduction of this general theory, where the microphones are placed along a closed curve around the listening area, and consequently the expansion/shrinking can only be done in the horizontal plane. Starting with a 1m-radius array, it is quite easy to derive the feeds for a loudspeaker array suitable for a medium-sized listening room, and to "stretch" the array so that the loudspeakers are arranged in

4 linear arrays instead of in a circular array. The next figure (partially taken from [8]) shows a schematic of this process.

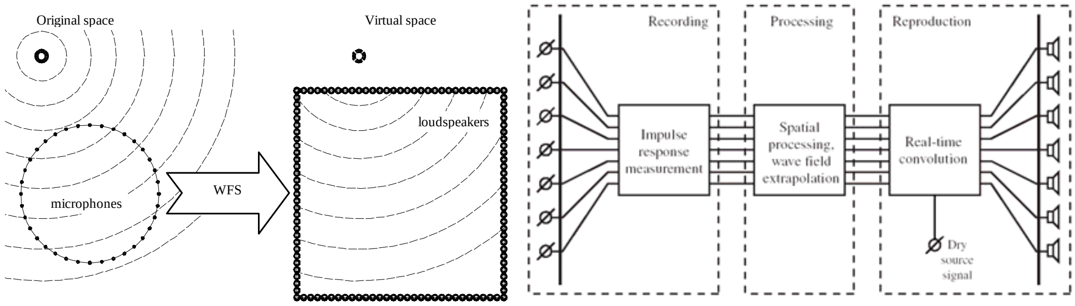


Figure 6: WFS processing scheme: concept (left) and implementation (right)

Hybrid methods (Ambiophonics)

The Ambiophonics method is an hybrid solution, aimed to mask the defects of two basic systems: cross-talk cancelled reproduction of binaural material over closely-spaced loudspeakers (Stereo Dipole) and 3D surround driven by convolution of corresponding oriented virtual microphones.

The following figure shows a typical Ambiophonics array, (frontal stereo dipole, plus 8-loudspeakers surround rig).

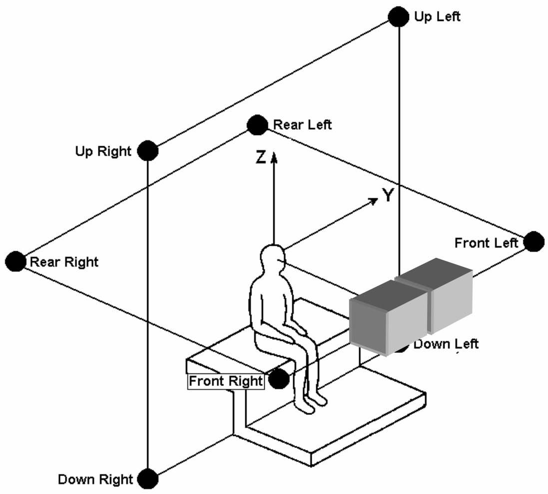


Figure 7: Ambiophonics array

The theory for deriving the signals for these loudspeakers has been already presented in the previous chapters, and the assembly of the whole system has been thoroughly described in [5].

LISTENING ROOMS

Two listening rooms suitable for three-dimensional auralization have been designed and built: the ASK room in Reggio Emilia and the Arlecchino room in Bologna [16].

Both rooms are equipped with an 8-loudspeaker Ambisonics array, plus a frontal Stereo-Dipole pair. The ASK room is also equipped with a subwoofer and a rear Stereo-Dipole pair.

Figures 8 and 9 show photographs taken in these two rooms.

Figure 8: the *ASK* listening room

Figure 9: the *Arlecchino* listening room

CONCLUSIONS

This paper has described how it is possible to derive subsets of impulse responses suitable for the reproduction of virtual acoustic spaces, following the currently available reproduction technologies. Referring in particular to the reproduction of the spatial properties of the sound field, it is noticeable that the data measured with advanced techniques allow for the auralization of the results employing:

- Standard stereo reproduction over a pair of loudspeakers;
- Binaural reproduction over headphones, with head tracking;

- Reproduction over closely-spaced loudspeakers (by means of cross-talk cancelling filters);
- Ambisonics reproduction over a 2D or 3D regular array of loudspeakers
- ITU 5.1 “surround” reproduction conforming to “standard” microphonic setups (OCT, INA, etc.)
- High directivity, multichannel reproduction by means of Mark Poletti’s circular-array method.
- Wide-area auralization by means of the Wave Field Synthesis approach (WFS)
- Any combination of the above methods, resulting in hybrid, higher-level surround methods (Ambiophonics, Panorambiophonics and derivations).

Consequently, this method provides the best available approach for storing the acoustical properties of famous and valuable rooms, such as concert halls and theatres, and preserving them for the posterity. The resulting data can be used for audible reconstructions (auralization) by means of today’s surround systems, without limiting the future usage by sticking to the limited reproduction technology currently available.

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