STUDY, CONSTRUCTION AND ANALYSIS OF A VARIABLE DIRECTIVITY OMNIDIRECTIONAL SOUND SOURCE

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Summary

Study for the realization of the source ................................................................. 3
Construction choices ........................................................................................ 6
Electroacoustic transducers ............................................................................. 8
Innovative characteristics ................................................................................ 12
Practical realization .......................................................................................... 13
Geometric form ................................................................................................. 13
Support structure ............................................................................................ 16
Connector .......................................................................................................... 18
Transducer positions ....................................................................................... 19
The assembled unit ......................................................................................... 21
Testing the source's power handling .............................................................. 23
Electrical & acoustical characteristics of the assembled unit ................. 23
Experimental tests .......................................................................................... 24
Testing directivity characteristics ................................................................. 27
CONCLUSIONS ............................................................................................... 33
**Study for the realization of the source**

There are two main fields of use for an acoustically isotropic source:
1. Calculation of the reverberation time of rooms for their acoustical characterization, in order to be able to carry out the changes necessary for sound diffusion suited to the specific application;
2. Assessment of the acoustic insulation ensured by the building structures separating two adjacent rooms.

During the source’s design phase, it was therefore necessary to refer to the current principle international standard on this subject, grouped together in UNI EN ISO 140-X and ISO 3382.

**Current standard**

The starting point for the realization of the omnidirectional source must necessarily be the reference standard for environmental measurement (UNI EN ISO 140-1).

In some cases, a blank pistol shot can be fired to acoustically saturate a room. Although very practical and inexpensive, this method has several disadvantages:
- the sonic impulse of the pistol shot hasn’t enough energy at low frequencies and therefore doesn’t give sufficient measurement dynamics to calculate reverberation time in the lower frequency bands;
- it’s not a test signal that can be reproduced with sufficient accuracy; this means that the measurement of the reverberation time isn’t reproducible either, unless the average of a series of measurements is used;
- it doesn’t allow to establish a stationary condition of sonic energy in the room, but just its response to the impulse;
- it easily overloads the measurement system because of its high crest factor.

Figure 1 shows as an example a generic reverberation time measurement chain.

In order to measure the reverberation time of any given room, it’s necessary to have a source able to:
1. acoustically saturate the room.
   A saturated environment guarantees a good calculation of reverberation time ($T_{60}$). Sources that are not very powerful are only able to raise the sound pressure level by 60 dB in very small rooms; in the majority of the cases, it’s decided to extrapolate the $T_{60}$ by doubling the decay time of the sound pressure signal of 30 dB ($T_{30}$).
2. Make the direct field as limited as possible.
   This means that direct radiation mustn’t be dominant compared to that arriving from the room’s reflections. In order to guarantee this effect, it’s necessary that the source doesn’t have preferential sound emission directions.
The condition of omnidirectionality can be easily checked in situ: the measurement equipment must give an almost constant level for acquisition in various points, i.e. the microphone must be outside the zone characterized by direct field (6dB each time the distance is doubled). In this sense, the standard suggests the use of sources radiating in phase mounted in a single cabinet. It even suggests the polyhedral form to be used: a dodecahedron. This is because, according to the abovementioned standard, a good approximation of omnidirectional radiation of sound energy is achieved. Naturally, there’s nothing against analysing other forms that have an almost spherical shape, such as other polyhedrons (such as an icosahedron or a truncated icosahedron).

To test the source’s directivity, the standard suggests measuring the sound pressure level as follows:

- acoustic free field;
- measurement of sound pressure at a distance of 1.5m.;
- test signal: noise (white or pink);
- 1/3 octave (ISO 140) or octave (ISO 3382) acquisition procedure.

The directivity index values are obtained by taking the difference between the average energy level for the full 360° (L_{360}) and the average values obtained by slowly passing all the arcs of 30° (L_{30,i}).

The directivity indices will be:

\[ DI_i = L_{360} - L_{30,i} \]

For the ISO 140 standard, the radiation can be considered as omnidirectional if DI values are within the limits of ±2 dB in the range of frequencies from 100Hz to 630Hz. In the range of frequencies from 630Hz to 1000Hz, the limits increase in a linear manner from ±2 dB to ±8 dB, whereas from 1000Hz to 5000Hz they have values of ±8 dB. Figure 2 shows the tolerance mask.
Figure 2: Limits within which a source is considered as omnidirectional according to ISO 140

On the other hand, for ISO 3382, radiation can be considered as omnidirectional if DI values are within the limit of ±1 dB in a frequency range from 125Hz to 500Hz. In the range of frequencies from 1000Hz to 2000Hz, the limits increase in a linear manner from ±1dB to ±5dB, whereas for 4000Hz, they have values of ±6dB. Figure 3 shows the tolerance mask for this standard too.

The standards foresee that this test, in the case of sources with a single loudspeaker, must also be carried out on various planes to be sure that the chosen plane isn't the symmetrical one. A single measurement plane is sufficient for polyhedral sources.

Figure 3: Limits within which a source is considered as omnidirectional according to ISO 3382
**Construction choices**

According to what emerged in the study of the current standard, the sound source must have the following requisites:
- high power;
- compact dimensions;
- omnidirectionality.

These three characteristics are unfortunately independently maximizable. A compromise had to be reached between these, carefully assessing the advantages and disadvantages of each single design possibility.

**The source’s shape**

The choice of the ideal shape of the cabinet in which the loudspeakers were to be mounted was based on practical and theoretical considerations, confirmed by acoustic simulation. Starting from the idea of completely symmetric distribution, regular polyhedrons are considered. The regular solids are the tetrahedron (with four triangular faces), cube or hexahedron, octahedron (eight triangular faces), dodecahedron (twelve pentagonal faces) and icosahedron (twenty triangular faces). It’s impossible to build regular solids with a different number of faces from these five polyhedrons. Obviously, the most symmetrical solid is the sphere, which can be considered as a polyhedron with an infinite number of faces, but building a spherical cabinet doesn’t mean that a number of loudspeakers can be fitted, in a completely symmetrical manner, other than 4, 6, 8, 12 and 20. The two solids that ensure the largest number of faces were taken into consideration: the dodecahedron (Figure 4) and icosahedron (Figure 5).

![Figure 4: Dodecahedron](image4.png)  ![Figure 5: Icosahedron](image5.png)

The former shape seems to enable better loudspeaker mounting. With this aim, the dimensions of a pentagon and a triangle limited by a circumference with a certain diameter (Figure 6) are compared.

![Figure 6: Comparison between the dimensions of a pentagon and a triangle limited by a circumference.](image6.png)
In the first case, 86% of the total area of the single face is occupied by the loudspeaker, while this percentage drops to 60% in the case of the icosahedron. This shows how the shape of the dodecahedron manages to exploit the space better.

On the other hand, the icosahedron allows twenty loudspeakers to be fitted instead of twelve: this, theoretically speaking, means that the source can be fed with 60% more power than the dodecahedron, thus ensuring a theoretical increase in maximum sound pressure level of 2.2dB. This improvement in the performance however has a considerable negative effect on practical use: in fact, with loudspeakers of the same dimensions, the overall dimensions (in terms of diameter of the sphere circumscribed about the polyhedron) increases by over 60% and the weight by approximately 70%.

Let’s now consider another aspect that depends on the dimensions: the phenomenon of destructive interference. This problem doesn’t occur for low frequencies, as the wavelengths involved mean that all the sources can be considered as being physically positioned in the same point. The sound field generated at these frequencies is approximately spherical and the contributions of all the transducers are summed. As the frequency increases, the loudspeaker gradually becomes more directive and the distance between the speakers becomes increasingly less negligible.

Standing at a distance of 1.5m in front of loudspeakers mounted on a dodecahedron, the result is that the five loudspeakers mounted on the adjacent faces are at such a distance that they cause destructive interference above 2800Hz; for that frequency, the sound field is very directive and therefore there is less interference at the listening point. Standing at the same distance as in the previous case in front of a face of an icosahedron with the same 130mm. loudspeakers, the three adjacent loudspeakers do not give problems, as they cause destructive interference at around 3600Hz, whereas the six loudspeakers with a vertex in common with the central face are at such a distance from the speaker in question as to cause destructive interference at 1400Hz.

As well as these two shapes, a truncated icosahedron was also taken into consideration (shown in Figure 7). This is a polyhedron obtained by truncating the pyramids with pentagonal bases corresponding to the twelve vertices of the icosahedron.

The resulting semi-regular solid has twenty hexagonal faces and twelve pentagonal faces (the same geometry as soccer balls!). One advantage of this shape is the fact that no less than 32 loudspeakers can be fitted on surfaces with a perimeter that well circumscribes the circle of the transducer (hexagons and pentagons). However, to exploit this characteristic to the utmost, it would be necessary to use two types of loudspeakers with different diameters, which complicates prediction of the system’s behaviour. Moreover, this solution implies excessive overall weight and dimensions.

Figure 7: truncated icosahedron
**Electroacoustic transducers**

The decision to use electrodynamic moving coil loudspeakers was due to the following advantages:

1) **They’re easily found on the market**: in fact, dynamic loudspeakers of this type are used to reproduce sound in the majority of cases (Hi-Fi systems, cinemas, computers, radios, televisions, etc.).

2) **Standard characterization**: precisely for the fact they’re very widespread, it was necessary to find a common method for describing the behaviour of individual transducers. *Thiele-Small parameters*, along with the frequency emission curve and the polar plot make a loudspeaker easily identifiable. It’s therefore possible to approximately predict the behaviour of the loudspeaker itself, without having to purchase it and then test it.

3) **High acoustic power:weight ratio**: although having quite low weight and overall dimensions, moving coil loudspeakers ensure high acoustic output. The weight and dimensions can both be further reduced (by up to fifty percent) by using neodymium magnets instead of traditional sintered ferrite ones. The weight problem, which can be overlooked in the case of just one loudspeaker, is of primary importance if the project involves mounting twelve loudspeakers in a cabinet that must be easily transported.

4) **Wide frequency range**: there are specialized loudspeakers for frequency bands of various levels: starting at the bottom end, these are *sub-woofers, woofers, mid-range units and tweeters*. Audio reproduction systems normally use a combination of several types of loudspeakers to cover the entire range of audible frequencies. The signal, appropriately filtered by electronic units called *crossovers*, is fed to the loudspeaker most suited to reproducing that particular frequency. Mid-range transducers able to go down to lower frequencies are called *extended-range* and are those most suited to the reproduction of test signals used for the analysis of room acoustics.

5) **Compact dimensions**: At the same power supplied, loudspeakers’ dimensions vary according to the range of frequencies they reproduce. A good low frequency loudspeaker needs to move a larger amount of air than a tweeter, therefore requires a much larger diaphragm with a long excursion. It’s therefore terrible for reproducing treble frequencies: above a certain frequency, a large diaphragm gives auto-resonance effects that jeopardise correct audio reproduction. Treble frequencies on the other hand require a light, rigid diaphragm with a short excursion.

For the frequency range necessary for the source being designed, a transducer with a nominal diameter of 5" is suitable: the single sources therefore have an overall size of approximately 130 mm.

Summing up the points covered so far, it can be said that the design specifications of the transducers are as follows:

- Sufficiently wide frequency response to efficiently reproduce all the frequencies required by the standards (125-6300Hz);
- 5” cone to reduce the dimensions;
- A coil with a diameter of 32mm. and magnetic oil in the air gap in order to handle continuous electrical power of up to 150W (according to the AES standard in the field of frequencies from 125Hz to 1250Hz);
- A neodymium magnet to reduce weight.

The product used for the OUTLINE dodecahedron is a custom-built extended range speaker with proprietary characteristics and excellent performance: the OUTLINE SSWM16NPP (Figure 8). When operating as described in the AES standard for power handling tests, the oil ensures better coil cooling and enables the transducer to be driven with a maximum power almost double that of a normal transducer without magnetic oil, providing mechanical conditions permit this. Moreover, through time, better control of the heat status eliminates the effect of Power Compression, due to the rapid rise in the temperature of the coil of a loudspeaker driven at high power.

The following are the technical characteristics of the transducers used for this project.
General characteristics
Frequency range 90-12500 Hz
Nominal impedance 16 Ω
IEC power 75 W RMS
Program power 150 W RMS
Maximum power 300 W Peak
Sensitivity (1W - 1m) 92.5 dB
Magnetic induction 1.1 T
Coil Diameter 32 mm

Thiele-Small parameters
R_E 10.03 Ω
F_S 91.75 Hz
Z_MAX 24.99 Ω
Q_MS 1.20 -
Q_ES 0.48 -
Q_TS 0.34 -
V_AS 4.72 dm³
BL 8.29 T/m

Mechanical data
Cone material Treated and waterproof cellulose paste
Suspension material Rubber
Magnet material Double Neodymium
Magnet weight 43+43 gr.
Total weight 800 gr.
Maximum diameter 133 mm.
Maximum depth 75 mm.

Figure 8: Outline S5WM16NPP loudspeaker

By means of a Bode plot, figure 9 shows the impedance curve of a single Outline S5WM16NPP loudspeaker.
Figures 10 and 11 on the other hand show the acoustic frequency response curve of a single Outline S5WM16NPP loudspeaker (1W - 1m), with an analysis in narrow band and 1/3 of an octave respectively.
Figure 11: Response on axis on a baffle of a single Outline S5WM16NPP loudspeaker - 1W/1m (Standard CEI EN 61260 - FFT Energy Integration at 1/3 octave)

**Cabinet material**

When a loudspeaker's diaphragm moves under the effect of vibrating/driving power, two perturbations are generated in the surrounding air: compression on the side in which the cone moves forwards and rarefaction on the opposite side. These two waves tend to cancel themselves, as they are two waves with opposite phase and practically coinciding spatial origin; this effect is more pronounced at low frequencies, i.e. when the wavelength is much greater than the physical dimensions of the diaphragm.

To prevent the rear wave from interfering with the front wave, a screen insulating the two perturbations is placed round the loudspeaker. This is the fundamental reason why all audio reproduction systems' loudspeakers are mounted in a special wood or plastic cabinet. One might be led to think that this cabinet is a sound box, such as that of a violin or guitar; in fact, it doesn’t have to amplify the sound, but just deaden the rear sound wave. This “screen” must have some characteristics:

1) It must be of soundproofing material. There would be no point in having a screen that lets the sound pass, or the vibrations out of phase with the useful sound.
2) It must be rigid. If it wasn’t, the variations in the pressure inside would make the sides vibrate, generating sound waves outside the speaker cabinet in phase with the internal wave, so out of phase with the external wave.
3) It mustn’t resonate. In other words, it must be made from an amorphous material that resonates as little as possible. Chipboard or plywood definitely are much better than solid wood, as they’re more amorphous, with no veins that could cause resonance. Plastic material can also be taken into consideration, particularly for the low costs involved in large-scale manufacture. In our case, plywood was chosen: a material that ensures good resistance with a limited weight and cost, as well as being easily machined.

Nevertheless, it must be noted that in fact the cabinet in which the transducers are fitted behave like a Helmholtz resonator. See the next section regarding the assembled unit (page 21) for the description of the advantages of this choice.
Innovative characteristics

At present on the market, there are sound sources approximately similar to those in this project. Not wishing to copy these, we tried to build a product that could not only ensure compliance with standards regarding architectural acoustics, but also for use in developing some innovative aspects in this field.

High power

Having had the opportunity of testing some of the existing omnidirectional sources, it was seen that they’re often too weak to saturate large rooms or those whose walls are not very reflective. This defect leads to imprecise reverberation time measurement of: not managing to raise the equivalent level by more than 60dB, the $T_{60}$ is obtained by extrapolation from the decay curve and not direct measurement.

For this reason, it was decided to concentrate on a source able to handle high power: the loudspeakers with which the prototype is fitted can reproduce signals with a power of over 900 $W_{RMS}$. As has already been said, the idea was also to design and build a very efficient loudspeaker. In fact, in spite of the dimensions of the diaphragm, it has been designed with a neodymium magnetic circuit and the diaphragm’s mechanical characteristics are able to ensure a gain of up to 4/5dB higher than the average efficiency values of the majority of loudspeakers on the market with the same dimensions.

Variable directivity

This project’s most innovative aspect is definitely the possibility of driving groups of three speakers separately with each of the four amplification channels foreseen. Thanks to a multi-pole connector and a multi-channel amplification stage, it’s possible to decide which and how many loudspeakers to use, but above all what type of signal to send to each group of speakers. The version with 4 groups of three loudspeakers will be supplied with an amplification system with four independent channels completely separate from each other. In fact, in the sources currently available, the single transducers are connected in a series-parallel network to which a single-channel amplifier applies the load. Having four independent channels at one’s disposal means, as well as the immediate consequence of being able to vary the source’s impedance from outside (therefore easily), it’s also possible to vary its directivity.

To concentrate all the sound power in one point, for example, it’s necessary to dephase the single sources for each frequency in such a way as to have constructive interference in that point. Dephasing is easily obtained with delay units suitably set and connected at the input of the individual amplifiers.

Again with the aid of circuits able to delay and process the signal’s waveform (e.g. inverting it or amplifying it), it’s possible to control the form of the generated sound field, obtaining emission lobes typical, for example, of musical instruments. This can be useful for an accurate study of the action to be taken for the acoustical treatment of a concert hall. A version of the source is available on request with 12 separate channels. In this way, each of the 12 loudspeakers can be powered independently from the others and the source is able to reproduce higher directivity characteristics.

![Figure 12: Spatial positioning and sound field of two suitably out-of-phase sources](image-url)
Figure 12 shows an example of a directive sound field obtained with non-directive sources. To simplify the explanation, the case shown has just two single-point sources: A and B. These are placed 0.34m apart and the signal that reaches source B is delayed by 0.001 sec. and inverted compared to the signal fed to A. The sound field generated in the frequency range from 50 to 400 Hz is *cardioid*, a typical form for sound fields for directional sources.

Thanks to the two sources being out of phase and the distance between them, a phenomenon of constructive interference is created in front of the transducers in the positive direction on axis X, whereas towards the negative axis, the two sound waves are still out of phase. Parameters such as distance and delay depend on the frequency in question. In fact, as can be seen in Figure 13, for higher frequencies, this form is not so well defined, due to the occurrence of other interference phenomena.

![Figure 13: Sound fields for frequencies higher than the ideal one](image)

**Independent signals**

Constructive and destructive interference phenomena are inevitable when several single sources are positioned in different points and powered by the same signal. Introducing, as has just been described, delay or phase control units, these phenomena can be induced in the desired point, but cannot be eliminated in all the points in the space.

When an omnidirectional source is used for architectural acoustics, the distribution of test signal's frequency energy is important, not the type of signal. There are therefore endless random pink noises, all of which are the same spectrum in the frequency dominion. Giving each amplifier a different test signal with the same frequency characteristics, there will be constructive and destructive interference in all the points in the sound field in an absolutely random manner, resulting in more evenly distributed sound pressure.

**Practical realization**

The guidelines for this project were defined in the previous chapter:
- the smallest possible dimensions;
- dodecahedral layout;
- plywood cabinet;
- Outline SSWM16NPP extended range transducers;
- multi-pole connectors for the control of each channel.

Let’s now see how the source was designed and built.

**Geometric form**

A dodecahedron is a regular polyhedron with 20 vertices, 30 sides and 12 faces. In order to realize it, its geometry must be known precisely. According to the dimension of each single face (for example the side) all the measurements of the solid can be calculated:
Side: 1

Area: \[ S = 3\sqrt{5(5 + 2\sqrt{5})^2} \]

Volume: \[ V = \frac{1}{4}(15 + 7\sqrt{5})^3 \]

Radius of the inscribed sphere: \[ r = \frac{5 + 11}{8\sqrt{5}} \]

Radius of the circumscribed sphere: \[ R = \frac{9 + 3\sqrt{5}}{8} \]

Dihedral angle:
\[ \theta = \cos^{-1}\left(-\frac{1}{\sqrt{5}}\right) = 116.56505 \]

Being a regular solid, the twelve faces are all equal, unless modification work is carried out later to enable the components such as the multi-pole connector, stand receptacle and handle to be fitted.

The dihedral angle \( \Omega \) is the angle between two adjacent faces of the polyhedron (see Figure 14).

![Figure 14: Dihedral angle](image)

As shown in Figure 15, to bring two faces together with the correct angle, it’s necessary for them to be “flared” by \( \theta/2 \).

![Figure 15: The flare angle of \( \theta/2 \) enables the faces to be correctly matched](image)
This means that the individual panels must be sections of a pyramid with a pentagonal base with the (virtual) vertex coinciding with the centre of the solid. La formula

\[ R = \sqrt{\frac{9}{8} + \frac{3\sqrt{5}}{8}} \]

gives the value of the radius of the sphere circumscribed about the polyhedron (see Figure 16). This sphere is tangent to the solid in all the dodecahedron’s 20 vertices.

![Figure 16: Sphere circumscribed about the dodecahedron](image)

It’s therefore possible to design a single panel like a pyramid section with a convex greater base whose radius of curvature is identical to the radius of the circumscribed sphere (Figure 17). Bringing the twelve faces together, the sphere formed.

![Figure 17: Panel for the spherical structure](image)

This new shape, conceptually identical to the previous one as far as the position of the individual transducers is concerned, is advantageous for the following reasons:

- there are no flat surfaces: a flat surface round a loudspeaker cuts and or boosts the signal according to the frequency in question and the panel’s physical dimensions;
- apart from the receptacle for the stand, there are no sharp corners that could cause diffractive phenomena: in this way, homogeneous radiation is favoured;
- from a commercial point of view, thanks also to other small construction ideas that will be described later, the spherical source is decidedly more attractive.
Support structure
For a correct use of the sound source, some components must be added to the structure, such as:
1. A receptacle for mounting the dodecahedron on its stand;
2. A plate for fitting the handle.

Both the stand and the handle must oppose the weight of the structure, so it’s advisable to fit them on the same action line of the force they have to balance. Of the infinite number passing through the centre of gravity, the most suitable is the one that also passes through two opposite vertices of the solid. The points furthest from the transducer, for each individual face, are in fact the vertices, so adopting this solution, the dimensions of the two components interfere as little as possible with the loudspeakers.
The two are also connected with a threaded bar to ensure greater structural rigidity and more uniform stress distribution. The M8 threaded bar is screwed on to the two components and locked by means of a nut and spring lock washer.

Stand receptacle
The connection to the stand is by means of a cylindrical pin mounted on the stand and a receptacle with the same diameter in the dodecahedron. Figure 18 shows a diagram of the component in question, while Figure 19 shows the work to be done on the vertex to fit the receptacle.

![Figure 18: Receptacle for fitting the stand](image18.png)

![Figure 19: Position for the receptacle in the vertex of the structure](image19.png)
As can be seen from the illustrations, the dimensions of the receptacle on the surface of the sphere is that of a circumference with its centre on a vertex; this had to be borne in mind when designing the individual panels, to avoid the stand receptacle interfering with the loudspeaker positions.

To ensure sufficient stability, it was considered necessary for the pin on the stand to enter a few centimetres into the receptacle. This resulted in considerable space being occupied in the cabinet, also due to the fact that above the pin receptacle a threaded dead hole had to be drilled for connection to the bar. To prevent the receptacle itself from interfering with the loudspeakers, some material must be removed. The truncated cone shape ensures the threaded bar the necessary hold (the space used by useable threads remains unvaried) and at the same time allows sufficient reduction in dimensions, as can be seen in Figure 20.

![Figure 20: Optimization of dimensions and space in the cabinet](image)

**Handle mount**
The component shown in Figure 21 is on the top vertex, diametrically opposite the receptacle and connected to it by means of the threaded bar.

This component, which we call a cap, has a twin function:

![Figure 21: Cap](image)
1. on the outside, it offers the possibility of fitting a handle to the dodecahedron for carrying it, or an eye-bolt for suspending it;
2. inside, it’s connected to the receptacle by means of a bar to prevent it from breaking away from the structure when the source is suspended.
Both the cap and the receptacle are axial-symmetric solids connected to the structure in order to avoid the risk of them rotating.

**Connector**

The choice of connector began with the need to ensure at least four independent input lines, in order to power groups of three loudspeakers separately and thus easily control directivity characteristics.

For this reason, the connection with the amplification system was realized with a Speakon 8-pole connector. This choice obviously requires a minimum of preparation as far as the cable and the connector are concerned.

The transducers installed on the prototype can handle a maximum power of 75 W_{RMS}, which, on a minimum nominal load of 1 Ω (configuration with all the loudspeakers in parallel), implies an overall system electrical power of 900W.

In order to create an object that was as spherical as possible, i.e. without any components interrupting the form’s regularity, it was decide to recess the connector into the cabinet, as shown in Figure 22. Figure 23 shows the prototype with the system for connection to the amplification set-up.

![Figure 22: Connector recessed in the cabinet](image)
In order to further improve the control of the source’s directivity, an AMP 24-pole connector can be fitted on request (Figure 24).

**Figure 23: The connector**

**Transducer positions**

The dimension of each individual panel depends above all on the overall dimensions of the loudspeaker. Figure 25 shows the design of the die-cast aluminium basket (frame) of the Outline S5WM16NPP loudspeaker designed, to occupy less space, with mounting holes inside the perimeter of the basket itself, without the traditional fitting tabs. The necessity of applying protective grilles over the transducers must also be considered. The grilles are fitted by means of adhesive polyethylene foam gaskets applied to the rim. The slot necessary for hosting the grille and gasket is able to hold the grilles without any screws being required.

**Figure 24: AMP 24-pole connector**
Figure 25: the Basket of the Outline S5WM16NPP loudspeaker

Figure 26: View of a single face with the current grille

Figure 27: View of a single face with the grille with a spherical cover
The grilles mounted at present on the product can be seen in Figure 26. The choice of shape was taken since, from the practical point of view, this type of grille had the advantage of having flat surfaces for laying down the source. Nevertheless, to reproduce the spherical shape, grilles are available with the shape of a spherical cap with the same radius of curvature as that of the cabinet (Figure 27).

It’s worthwhile stressing that nevertheless, from the acoustical point of view, the shape of the grille doesn’t in any way influence the sound field produced by the source.

**The assembled unit**

Considering the overall dimensions of the transducers mounted in the centre of the faces and their relative grilles, as well as the components located on the vertices, it’s possible to define the ideal dimension of the pentagon that enables to exploit the space to the utmost without interference between components or excessive empty spaces. This means that the resulting sphere has a diameter of just 310 mm. Figures 28 and 29 show respectively the three-dimensional models of the assembled wooden structure and the cross-section highlighting the support system.

Lastly, Figure 30 shows the finished plywood prototype with open grain and walnut paint finish, whereas the standard product has a scratchproof water-resistant paint finish, as is normally the case with professional equipment.

![Figure 28: Cabinet](image)

![Figure 29: Cross-section](image)
From the photo shown in Figure 30, it’s possible to see a series of holes covered with material permeable to air, at the vertices of the sections from which the source is built. It was decided to machine these holes in the cabinet for various reasons:

1. The first advantage was greater power handling than a configuration in pneumatic suspension around the frequencies of the reflex tuning. In fact, at the tuning frequency, the loudspeakers are almost stationary and therefore mechanically in safety conditions precisely in the frequency band in which there’s a risk of mechanical breakages due to the amount of diaphragm movement. This advantage is achieved without set-offs, since frequencies below resonance do not need to be reproduced by the dodecahedron anyway and can therefore be filtered (even electronically) in order to improve the operation of the reflex load and avoid the eventuality of uncontrollable diaphragm movement below tuning frequency.

2. The second advantage consists in the increase in efficiency around the tuning frequency as, since the reflex system behaves like a high-pass system of a higher order than the equivalent system in pneumatic suspension, there’s an increase in the level of approximately 3 dB in the tuning zone, followed by a rapid drop-off compared with the case of a pneumatic suspension system, with the advantage already mentioned.

3. The third advantage consists in the fact that the reflex system also implies lower loudspeaker distortion, due to less diaphragm movement.

4. The fourth and last advantage consists in the exchange of the air between the inside and outside of the cabinet; as well as the higher thermal exchange characteristics of the loudspeakers used, this exchange enables to keep the temperature of the electromagnetic circuits of the speakers and the internal part of the source very low, even in the event of it being driven for long periods at high power.
Testing the source’s power handling

In order to characterize the source’s power output, tests were run on the power handling according to standard AES2-1984-(r1997) (ANSI S4.26-1984) – “AES Recommended Practice Specification of Loudspeaker Components Used in Professional Audio and Sound Reinforcement”. According to this standard, the sound source must be driven with pink noise that extends for ten bands above the bottom frequency limit in question (90-900Hz). This pink noise must be filtered with a Butterworth filter with a 12dB/octave slope. Moreover, the ratio between the peak voltage and the RMS value of the voltage must be equal to 2 (a maximum difference of 6dB between these two quantities is accepted).

The source must be driven with increasingly higher electrical powers, based on the formula:

\[ W = \frac{V_{\text{RMS}}^2}{Z_{\text{Min}}} \]

This power must be continually maintained for two hours. This test is considered as having been passed when the variation in the acoustic characteristics before and after the test is less than 10%. The temperature of the magnets must be monitored during the test.

Electrical and acoustical characteristics of the assembled unit

In order to carry out the power handling test, it was necessary to establish the electrical characteristics of the entire assembled unit. In particular, to establish the power supply to the dodecahedron, the definition of the source’s electrical impedance was of fundamental importance. The configuration of the electrical connections for this type of test involved the connection of three loudspeakers in parallel on each amplification channel. The valuation was carried out using a MLSSA/SPO acquisition system. The value of impedance measured, and therefore used in the following valuations, as per the provisions of the AES recommendations, is the minimum impedance value: 3.96 Ω at a frequency of 165Hz (Figure 31) for each group of three loudspeakers.

Following a series of tests, it was decided to feed each of the four channels (with the type of connection between the loudspeakers described above) with a prudential power voltage of 23.7 V. Each single channel was thus fed with a power of:

\[ W_{\text{RMS}} (3 \text{ altoparlanti}) = \frac{V_{\text{RMS}}^2}{Z_{\text{Min}}} = \frac{23.7^2}{3.96} = 141.8 \text{W} \]

Whereas to feed the entire source, it was necessary to feed out by means of two Outline EX 1500 amplifiers, a \( W_{\text{RMS}} \) power (12 loudspeakers) of approximately 567.2 W distributed over four 4 Ω rated channels.

The main characteristics of the EX 1500 Series of amplifiers are as follows:

- Power, both channels operating (8/4 Ohms) 450/ 730 WRMS
- Input impedance 100 kΩ (balanced)
- Bandwidth (Hz) 10 - 80000 +0dB /- 3dB
- 20- 20000 - 0.5dB
- Maximum output power: 730 W\text{RMS} per channel on 4 Ω
Figure 31: Impedance of a group of three loudspeakers of the source (values refer to the 90-900Hz portion of the spectrum)

**Experimental tests**

The experimental tests to establish the source’s power handling were run on 30\textsuperscript{th} October 2003 in the semi-anechoic chamber at the Outline premises in Flero (BS). The following instruments were used:

- Bruel & Kjaer PULSE 3560 Multi Analyzer System
- Bruel & Kjaer 3015 acquisition cards
- Bruel & Kjaer 4190 microphone
- B&K 4228 Calibrator
- Versatherm 707 L.C.D. versatile thermometer
- Model K thermocouple
- 2 Outline EX 1500 amplifiers
- Outline MX 405 mixer
- Outline ET2-ST2 turntable
- Cool Edit 2000 software
- AGEMA 590 PAL Thermocamera

The system of acquisition of the acoustic parameters was calibrated before and after each measurement session and no deviation of more than 0.01 dB noted.

A B&K 3015 acquisition card was used to check the mean sound pressure level on a rotation of 360° of the source round its vertical axis, measuring the spectrum at regular intervals in order to monitor the acoustic yield of the source.

The second B&K 3015 card was used to monitor the power voltage of the speakers in order to keep any problems of power compression under control. The internal temperature was monitored by gluing a model K thermocouple to the magnetic circuit of one of the speakers. The temperature during the various phases of the test, the power voltage of each group of loudspeakers and the mean sound pressure on one turn of the source are shown in Table 1.
### Table 1: Temperature, Voltage and Pressure Level behaviour during the test

<table>
<thead>
<tr>
<th>Time passed [min]</th>
<th>Temperature [°C]</th>
<th>Voltage [V]</th>
<th>Lp [dB(L)]</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>25.3</td>
<td>23.7</td>
<td>128.0</td>
</tr>
<tr>
<td>30</td>
<td>42.5</td>
<td>23.7</td>
<td>127.8</td>
</tr>
<tr>
<td>60</td>
<td>44.2</td>
<td>23.7</td>
<td>127.7</td>
</tr>
<tr>
<td>90</td>
<td>44.6</td>
<td>23.7</td>
<td>127.7</td>
</tr>
<tr>
<td>120</td>
<td>44.9</td>
<td>23.7</td>
<td>127.7</td>
</tr>
</tbody>
</table>

The sound pressure level was measured on the radius of the sphere that had as its centre the centre of gravity of the source and with a surface area of 1m². Figure 32 shows, as an example, a thermograph carried out on the sound source during the power handling test.

![Thermograph of the sound source at the end of the power handling test](image)

**Figure 32:** Thermograph of the sound source at the end of the power handling test

In compliance with the provisions of the criteria of the AES standard, after 4 hours of cooling of the reproduction system, when the source’s internal temperature was 19.0°C, a check was made to ensure that the source had not undergone a variation of over 10% in its performance. The sound pressure level measured under these conditions, with a power voltage of 23.7V, was 128.1 dB (Table 2).

### Table 2: Check on power handling after cooling

<table>
<thead>
<tr>
<th>Time passed [min]</th>
<th>Temperature [°C]</th>
<th>Voltage [V]</th>
<th>Lp [dB(L)]</th>
</tr>
</thead>
<tbody>
<tr>
<td>120 ON + 240 OFF</td>
<td>19.0</td>
<td>23.7</td>
<td>128.1</td>
</tr>
</tbody>
</table>

Two successive attempts at powering the source according to the AES standard with higher voltages over long periods led, in both cases, to the breakage of one of the speakers. Therefore, thanks to the tests carried out according to the AES standard, it was possible to state that the sound power the OUTLINE omnidirectional source is able to feed out continuously without loss in performance is 128 dB re 1pW. Nevertheless, in the case in which the power is diluted over a wider frequency band than that foreseen by the AES standard, the sound source can be driven, for shorter periods with higher electrical powers than those used for the power handling test.
To prove this, distributing the power on the 90-9000 Hz band (instead of the 90-900 Hz band used for the measurement according to the AES standard), a further test was run, taking the source’s power stages to the limit. The sound power fed out for a few minutes by the source was 130.3 dB re 1pW. Figure 33 shows the frequency distribution of the sound power generated.

![Figure 33: Frequency distribution of sound power (1/1 octave and 1/3 octave)](image)

Figure 34 on the other hand shows the sonogram measured on a complete rotation of the source on its axis.

![Figure 34: Sonogram of acoustic response (1/3 octave) over a complete rotation of the source - Maximum power @ 0.28m. The left-hand scale shows the longitude with 144 divisions per 360° (an equivalent level every 2,5°)](image)

With a view to improving the source’s frequency response, another test was carried out with the appropriate equalization of the electrical power fed into the source, obtaining the frequency distribution of the sound power level shown in Figure 35. In this case, the sound power achieved by fully exploiting the two Outline EX 1500 amplifiers was 127 dB re 1pW.
Figure 35: Equalized source sound power in 1/1 octave and 1/3 octave bands

For the final version of the system, more powerful integrated amplification stages will be used in order to have sufficient margin to avoid clipping of the output stage, which limits the supply of power in the event of equalization. Consequently, an increase in the sound pressure level will also be achieved in the case of research by means of the equalization of a more even frequency distribution of the sound power. For short operating periods, the loudspeakers will not be damaged because, as equalization will only be applied above 1000Hz, the magnetic oil in the loudspeakers limits the increase of the temperature of the coil due to the higher voltage, which in this case is mainly supplied at the high frequencies. Therefore, using a suitable amplification system, as is planned for the production version, a more uniform sound pressure level will be obtained over the whole reproduced spectrum, while maintaining the maximum dynamics allowed by the loudspeakers’ electromechanical characteristics.

Testing directivity characteristics

In compliance with international standards ISO 140, DIN 52210 and ISO 3382, a test was also carried out on the sound source’s directivity. This calculation was done by placing the source in OUTLINE’s semi-anechoic chamber and driving it with 1W. The microphone was then positioned at 1.5m. from the axis of the turntable used to rotate the source. The tests were carried out on December 22nd 2003. Room temperature was 21°C and the pressure 999 hPa. The measurement procedure used was the same as described previously for the power handling test.

Figure 36 shows the sonogram obtained by repeatedly measuring the equivalent sound pressure level on the medium plane of the source at 1.5 m. from the axis, with intervals of 2.5° of rotation of the source round its vertical axis. The corresponding tolerance diagrams according to norms ISO 3382 and ISO 140 are shown in Figures 37 and 38.
Figure 36: Sonogram of the acoustic response (1/3 octave) on a rotation of the source 1W @ 1.5m. The right-hand scale shows the longitude with 144 divisions for 360° (an equivalent level every 2.5°).

Figure 37: Test of frequency directivity index according to ISO 3382 standards

Figure 38: Test of frequency directivity index according to ISO 140 standard
To complete the description, the diagrams from Figure 39 to 46 show the polar representation of sound pressure level data measured at a distance of 1.5m. from the rotation axis of the source driven with an electrical power of 1W.

Figure 39: SPL polar plot - 1w @ 1.5m (octave bands 125-500Hz)

Figure 40: SPL polar plot - 1w @ 1.5m (octave bands 1000-4000Hz)
Figure 41: SPL polar plot - 1w @ 1.5m (1/3 octave bands 100 - 160Hz)

Figure 42: SPL polar plot - 1w @ 1.5m (1/3 octave bands 200 - 315Hz)
Figure 43: SPL polar plot - 1w @ 1.5m (1/3 octave bands 400 to 630Hz)

Figure 44: SPL polar plot - 1w @ 1.5m (1/3 octave bands 800 to 1250Hz)
Figure 45: SPL polar plot - 1w @ 1.5m (1/3 octave bands 1600 \( \pm \) 2500Hz)

Figure 46: SPL polar plot - 1W @ 1.5m (1/3 octave bands 3150 \( \pm \) 5000Hz)
CONCLUSIONS

On the basis of the studies and measurements carried out, it can be concluded that for its performance, reproducible band width and homogeneity of acoustic emission, the OUTLINE omnidirectional sound source that is the subject of this studio is a sound source whose characteristics represent the state of the art in the field of sound sources for professional acoustical use in the architectural acoustics field.

Brescia, 1st March 2004

Technical supervision

| Edoardo Piana (engineer) | Giuseppe Pedersini (engineer) |

Scientific supervision

Prof. Gian Paolo Beretta

Bibliography


STATEMENT OF CONFORMITY

It is hereby stated that the OUTLINE omnidirectional sound source model GSR Globe - Source - Wide Band Isotropic Radiator, S/N: FACSIMILE is in compliance with the characteristics defined according to the procedure and instruments indicated in the document "Study, realization and analysis of a variable directivity omnidirectional sound source" and satisfies the indicated Standards.

TECHNICAL SPECIFICATIONS

- Nominal impedance for each of 4 channels: 4 Ohms
- Operating frequency field: 90 – 9000Hz
- Power handling: 128 dB re 1pW 90 – 900 Hz (Standard AES2-1984-(r1997))
- Maximum sound power level: (with two Outline EX 1500 twin-channel amplifiers, signal: pink noise 90 – 9000 Hz).
- Broad band: 130dB re 1 pW
- Spectral: Min. 98 dB re 1 pW in each 1/3-octave band
- Connector: Speakon Neutrik NL8
- Diameter: 310 mm
- Weight: 12.5 kg.

Frequency directivity index according to ISO 3382 and ISO 140 standards:

<table>
<thead>
<tr>
<th>Frequency (Hz)</th>
<th>Directivity Index</th>
</tr>
</thead>
<tbody>
<tr>
<td>125</td>
<td>0.0</td>
</tr>
<tr>
<td>250</td>
<td>0.0</td>
</tr>
<tr>
<td>500</td>
<td>0.0</td>
</tr>
<tr>
<td>1000</td>
<td>0.0</td>
</tr>
<tr>
<td>2000</td>
<td>0.0</td>
</tr>
</tbody>
</table>

Maximum sound power levels at 1/1 octave and 1/3 octave (achievable with two Outline EX 1500 twin-channel amplifiers, or others with the same or better performance, fed pink noise equalized in the 90-9000 Hz band):

- [Graph 1](#)
- [Graph 2](#)

The source is in compliance with the following standards: ISO 140-3, ISO 3382, DIN 52210.

Brescia, 9th March 2004

Head of the Acoustic Laboratory  
Prof. Gian Paolo Beretta

Chief Technician  
Edoardo Piana