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"CONCISE PARAMETERS FOR ASSESSING A PRO AUDIO SYSTEM FOR SOUND REINFORCEMENT USE"

While the November issue with my last article is still being mailed out I've got to write this follow-up for the January 2001 issue, as the editor has to go to press earlier than usual, due to the Christmas vacations, to give the printers sufficient time to deliver the magazine for mailing to subscribers.

Although this positive fact emphasizes the editor's commitment, unfortunately it causes me a few small problems.

In fact, I won't be able to write this piece taking into consideration the opinions, comments and suggestions that readers give after reading the previous issue, on a topic which I think should be covered in depth by all concerned, precisely in order for the result of this relationship to be approved and shared.

In fact, this "rush" compels me to put forward my idea without having been able to compare it with any others, perhaps even very different, apart from those in a pair of "e-mails" I've already received, and those that may arrive while I'm writing this article.

Before suggesting a new yardstick, or rather a new parameter, which is simple, but sufficiently significant to enable to rapidly evaluate a professional sound system, in particular (but not only) for live concerts, it's firstly worthwhile listing and carefully analysing **its most important distinguishing features with a view to practical use**. This analysis will show which parameter or parameters are most suited to giving the reply to the most important question in this situation:

"Will our sound system be sufficient to enable us to adequately support the performance or show it's been chosen for?"

This parameter definitely can't be the "Watt", because (as I showed in the previous article) by itself it is meaningless in this circumstance. However, ***it will have to be a substitute with the same or similar conciseness***; which I think is the only reason this "unit of measurement" has spread so widely among less technically minded trade members, by the most traditional "word of mouth".

Another equally necessary preliminary aspect is the choice of a **"Standard"** by means of which these characteristics or performance must be quantified, in order that that the terms of comparison (the figures used to compare various sound systems) can have a foundation, as they're obtained according to methods and procedure described in the context of the measurement **"Standard"** itself.

My suggestion is predictable and I think all concerned will agree on it, because it's already endorsed by the approval of thousands audio engineers and designers worldwide, consists in the choice of the **AES Standard**, which is based on Americans' typical pragmatism and for this reason quite "quick" as well as coming complete with practical informative appendices.

However, since this Standard (drawn up in far-off 1984) essentially covers components, not loudspeaker enclosures or complete systems, it should be referred and adapted to the latter, while leaving the principles and measurement methods applied unchanged.

This need for adaptation emphasizes the fact that the Standard should be updated, adding specifications more suitable and adequate for describing more accurate techniques and methods for measuring loudspeaker systems, or even "arrays" of various types.

I'm therefore of the opinion (as are many "technicians" in this trade) that it would be worthwhile integrating this Standard (but striving to leave the original pragmatism unchanged) with some parts of another standard, the **IEC**, see Fig. 2, to obtain a more complete instrument for providing the necessary "elements" for the most objective and practical quantitative evaluation of professional sound reinforcement systems.

In fact, the **IEC Standard**, more widespread in Europe, for many aspects candidates itself as a complement where the nonetheless excellent **AES Standard** is lacking, while it definitely lacks the latter's immediacy.

Of the two, I've therefore decided to use (wherever they exist) the rules and recommendations of the latter, since it was explicitly drawn up for professional use, but I'll also draw on the former whenever I find it more suitable and objective.

However, I ensure that assessing sound systems according to one standard or the other, or even according to both, doesn't lead to any differences in the substance. This obviously remains unchanged, but there' simply a difference in the type of assessment parameters, even if in the less technically minded this fact can easily cause misunderstandings, and often leads to gross errors of evaluation.

In fact, if you forgive a simile that helps understand the reason for the confusion in those who have to evaluate specifications in two different Standards (or a combination of them), it's like reading a piece of text written in two languages which have contaminated each other through time (Italian and English for example); if readers don't understand both perfectly, they run the risk of easily making mistakes, or even interpretations with the opposite meaning.

Then there are other standards, such as the **EJA** (not used much) or the **DIN**, which has practically disappeared; but in any case, whatever is measured or stated according to these standards would not add to or detract in any way from the validity of the **AES Standard**, the most used and widespread.

I therefore hope that this Standard, which I follow as much as possible (I've been a member for years of the international association that promoted it, and Outline is a sustaining member), is improved and the missing parts completed soon.

Showing the covers below to ensure impartiality and clarity, I suggest all those who are interested in going into this matter in greater depth should buy them from the Italian branches of the international organizations that publish them.

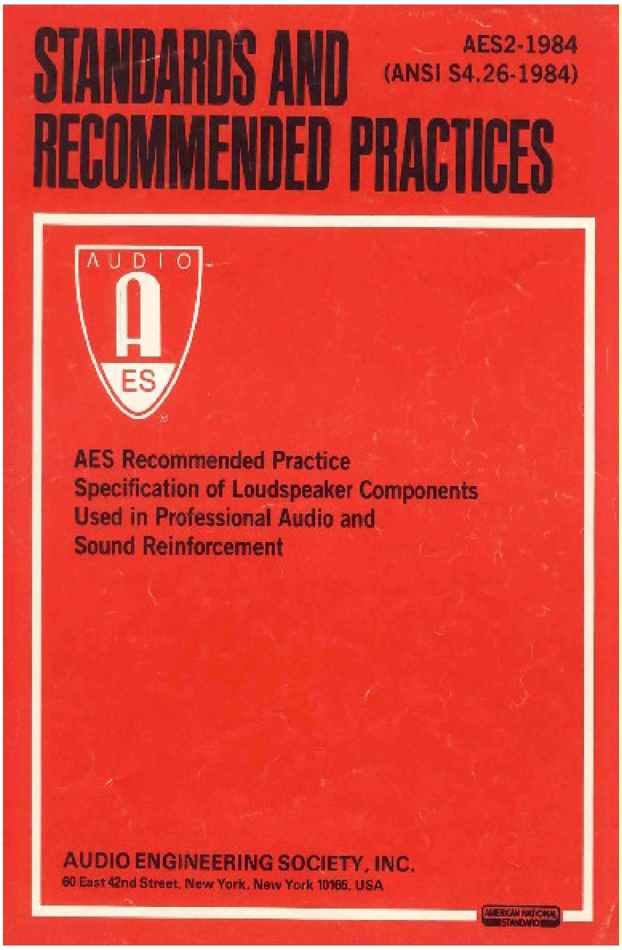

Fig.1 The cover of the AES Standard booklet

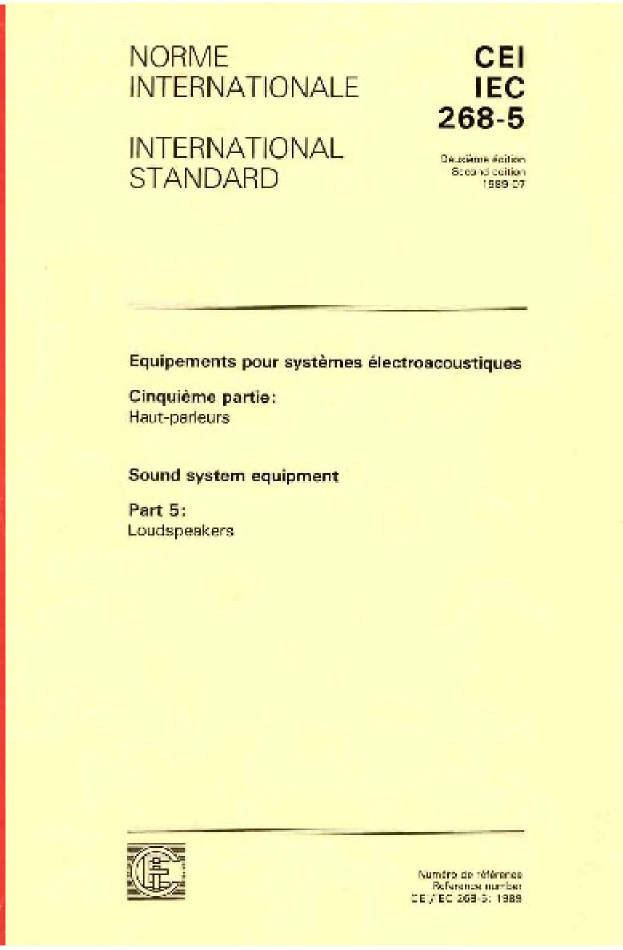

Fig.2 The cover of the IEC 268-5 booklet

Having said this and having therefore adopted an "integrated standard", or rather a combination of the two, as a "context" in which to develop a concise criterion of evaluation that replaces the "Watt", what are the most important characteristics of a sound system from which an objective response to the fundamental questions posed above can be obtained?

Running through those that suit our needs and ***only regard quantitative criteria*****,** there are:

1. **Impedance**
2. **Sensitivity**
3. **Efficiency**
4. **Axis Frequency Response**
5. **Effective Frequency Bandwidth**
6. **Power Handling**
7. **Max. SPL - Sound Pressure Level**
8. **Radiation Angle, Coverage**
9. **Power Compression**
10. **THD - Total Harmonic Distortion**

I don't intend describing the parameters listed in these pages in depth, as I'd run the risk of going well beyond their aim. I'll therefore briefly mention their meaning in the shortest possible manner and the "weight" they assume for the evaluation of a sound reinforcement system. I suggest those of you who want to go into greater depth read the many excellent books that cover this issue to a varying degree.

In any case, these are the parameters to consider carefully and which have to be "reckoned with" before "summing up" the performance of a sound reinforcement system, as will be seen. In fact, they're interdependent and therefore contribute to a varying degree in forming a criterion of quantitative evaluation.

To tell the truth, there isn't only a quantitative value, since their size also allows to highlight performance purely from the point of view of quality, such as for example tone balance or listening fatigue, etc. However, even if very important, this is another "story" I'll definitely have the opportunity of covering in the future along with other equally interesting matters.

IMPEDANCE

A complex electrical quantity that "describes" the opposition to the flow of alternating current in an electrical circuit in relation to the load.

The load of a loudspeaker enclosure changes with the frequency and definitely can't be considered as a mere resistance. In fact, impedance has a resistive and a reactive component.

Without going into greater detail, measuring the impedance is essential for the accurate choice of the power of the amplifier to be used and is necessary for the measurement of the real sensitivity of the enclosure, which should almost never be evaluated bearing in mind the *Nominal Impedance* indicated by the manufacturer, as this value is rarely true. In fact, the Nominal Impedance value, defined by the IEC Standard, mustn't exceed the Minimum Impedance value by more than 20%, whereas it usually does (for example, an enclosure with a nominal impedance of 8**Ω** must not have a minimum impedance of less than 6.4**Ω**, whereas values lower than 6**Ω**, or even 5**Ω**, are almost always seen). Bypassing the problem, the AES Standard suggests measuring the Impedance at its **minimum value** in the enclosure's operating band (**Zmin**).

On one hand, this seems to penalize the datum itself, which is the lowest possible for an enclosure, but on the other (in my opinion) this value, being used to calculate the power of the amplifier, becomes an advantage, as it "automatically" leads to greater caution being taken.

In fact, calculating the maximum power to apply to an enclosure at its minimum impedance in the operating band, means that, at the worst, when the signal fed to the enclosure excites precisely that frequency, by means of the amplifiers connected, the maximum power the latter would feed out will never exceed the power initially calculated after due consideration. The measurement of impedance is generally shown with graphs like Fig.3.

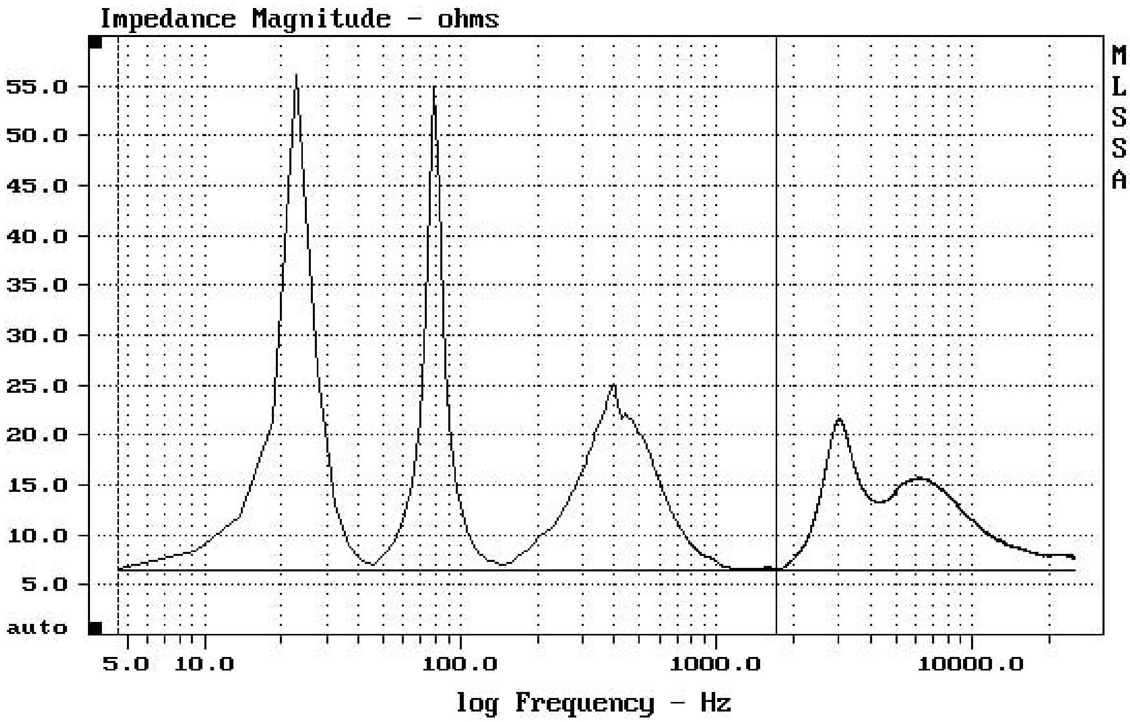

Fig. 3 – Graph of the impedance curve of a multi-way enclosure - 8Ω (nominal).
The marker shows the minimum impedance value, approximately 6.5Ω at 1720Hz.

SENSITIVITY

The ratio of input power to output power on a device, whether it's mechanical (a loudspeaker) or electronic (a preamplifier).

In the case of a loudspeaker enclosure, it indicates the power of the output signal for a standard input signal. The measurement unit is normally dB SPL and the input signal, according to the AES Standard, is classic pink noise, whose test level is 1 Watt at the minimum impedance, **filtered according to the frequency band for which the manufacturer declares the unit's sensitivity**.

Measurements are taken in Free Field (4**π**, anechoic etc.) or Half Space (2**π**, etc.) at a distance of at least four times the maximum dimension of the enclosures being tested, then the value there would be at 1 metre is calculated using the "Inverse Square" law.

This is also an important parameter because, along with the impedance value, it allows us to have a certain sound pressure figure (measured in set conditions, with a given reference voltage and distance), from which the enclosure's maximum SPL performance at maximum power can be calculated. See Fig.4

EFFICIENCY

This parameter could possibly be omitted from this list. I'll cover it briefly, to make certain readers don't confuse it with the previous one - SENSITIVITY.

Efficiency is measured as a percentage and shown with the Greek letter **η**% (eta).

It shows the percentage of the applied input energy that is converted into acoustic energy by the loudspeaker enclosure. This figure is an indication of the system's electroacoustic yield and varies for the various types of loading in which the loudspeaker is used – e.g. Bass Reflex or Horn). It has nothing to do with the sound pressure level emitted by the enclosure: this is found by referring to the methods used to measure SENSITIVITY, the parameter that interests us for practical purposes.

AXIAL FREQUENCY RESPONSE

Represents a graph of output level or sensitivity (therefore the same standard input signal is used).

Shows the difference in the loudspeaker enclosure's sensitivity in relation to frequency variation. This is also an important datum, above all in the case of a multi-way enclosure with each section amplified separately. It can also be said that it shows the range of frequencies to which the enclosure will "respond" and the relative amplitude with which they will be reproduced.

EFFECTIVE FREQUENCY BANDWIDTH

This parameter could in fact be included in the previous one, but I prefer to describe it separately, to emphasize its great importance for the evaluation of performance, particularly when manufacturers do not supply a frequency response graph, but just a numerical indication.

In fact, this is the enclosure's "passband" and is defined by an allowance range that shows its maximum deviation (at both low and high frequencies) compared to the average value of an octave, which includes the zone of maximum sensitivity, within the frequency response stated by manufacturers.

The AES Standard doesn't propose any range in which to evaluate the response.

The IEC Standard on the other hand does, and sets the figure at -10dB. In other words, all the frequencies, with a value within 10dB under the average value of the zone of the response that contains the highest significant peak, come within this range (<u>peaks and gaps with a width of less than 1/9 of an octave do not have to be taken into consideration</u>). The lowest and the highest of these frequencies therefore establish the enclosure's useable bandwidth.

The value of 10dB is not a casual choice, but corresponds in psychoacoustics (the area of acoustics that studies the "way" in which humans hear) to halving or doubling the <u>sound sensation</u>.

Someone might object that 6dB halves or doubles the **<u>sound pressure</u>** (SPL) measured and therefore suggest this value as a range for evaluation. I personally wouldn't object, if it wasn't for the fact that there's a Standard regarding this and I therefore consider it risky to change it, even if the change had a logic that could be described as an "improvement".

However, what counts in order to carry out comparisons that are as correct and objective as possible, is the fact that all the figures supplied by manufacturer or trade members must respect the methods laid down by the standards agreed on and adopted, whatever they are. See Fig.4.

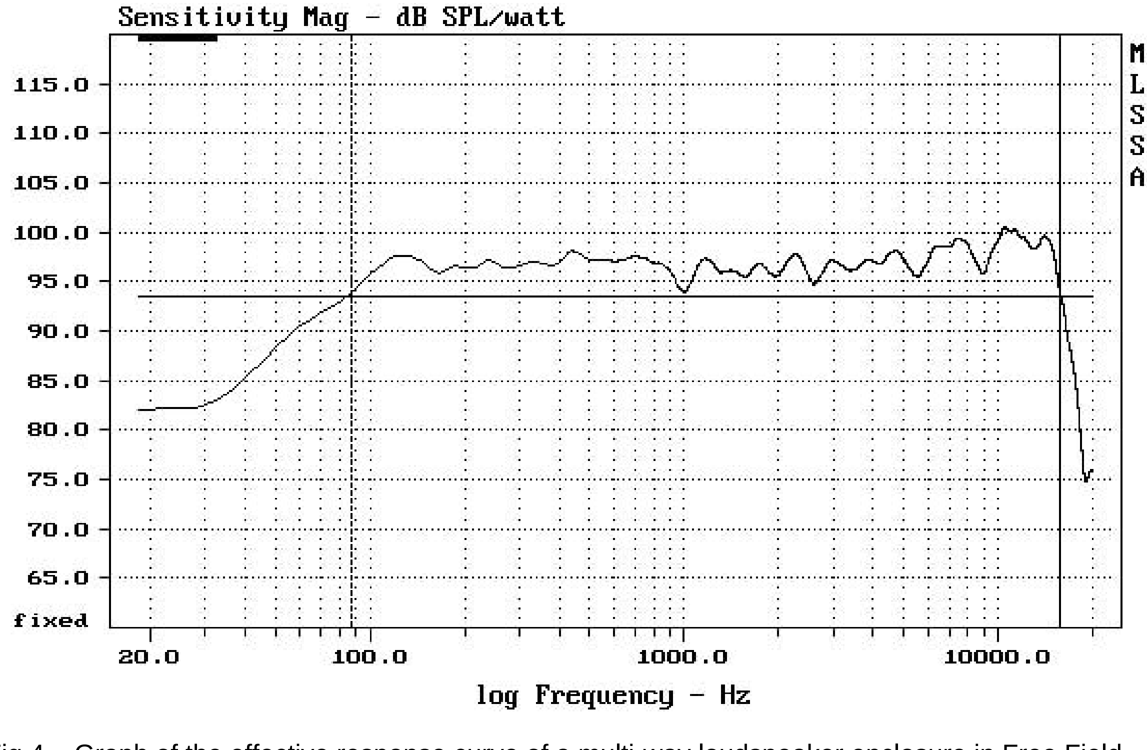

Fig 4 – Graph of the effective response curve of a multi-way loudspeaker enclosure in Free Field.

The two markers show the effective frequency band at approximately -6dB, which goes from 87Hz to 16,000Hz The level varies from 94 to 100 dB approx., whereas the RMS value is approximately 98 dB SPL, corresponding to the value of average wide-band sensitivity in Free Field (4π).

POWER HANDLING

The electrical energy, measured in Watts RMS, that the loudspeaker enclosure is able to receive from the amplifiers according to a precise method, established by the measurement standard adopted. In the AES Standard, the most realistic for use in professional applications, this is measured by applying pink noise filtered for the frequency band stated by the manufacturer.

This signal, the power of which is calculated on the minimum impedance, is gradually increased for as long as the loudspeaker enclosure shows it's able to handle it for two consecutive hours without a permanent alteration of over 10% in its acoustic, mechanical or electrical performance. **Under these conditions, the amount of power supplied is the loudspeaker enclosure's maximum power handling capacity**.

It seems to me needless to stress the importance of this parameter, on which the system's real maximum performance will depend. The datum is numerical and shown as **W AES**; i.e. "Watts as per AES standard". In fact, the Watts are still effective W RMS (average value), no matter what they're called; the abbreviation AES merely shows the method use to measure them.

MAXIMUM SOUND PRESSURE LEVEL

The sound pressure found in a given point, at a certain distance and at a certain frequency. This parameter, which depends on the power handling capacity, is also important for understanding up to what distance the loudspeaker will be able to usefully reinforce sound in the direction of its axis, above all outdoors, or at any rate in large areas.

Since it's extremely unlikely that manufacturers will give this figure after having effectively measured it (for problems of a practical nature which are often insuperable), it's normally obtained by calculation. It's therefore without doubt valid and comparable if calculated on the basis of Sensitivity and Power Handling values measured under the same conditions for all the products to be compared. This is also a numerical datum and is given in dB SPL, normally at the reference distance, 1 metre in Free Field, as with the Sensitivity's SPL value. I'd like to specify that the maximum pressure stated in this way is obtained from the calculation of the SENSITIVITY value; so, since the latter is an average datum in relation to the band taken into consideration, MAXIMUM SOUND PRESSURE will also be an average value considered in the same band and not taken at just one frequency.

For example, a loudspeaker enclosure, whose sensitivity response, **1w/1m/4π/ Zmin**, has an average value over a wide band of 98 dB SPL, in the same conditions, having assumed a power handling capacity of 100 W AES, it will reach a maximum sound pressure over a wide band of 118 dB SPL.

RADIATION ANGLE, SOUND COVERAGE

A parameter that indicates the capacity of loudspeaker enclosures to radiate sound in directions other than the propagation axis.

This parameter, which is indicated by a value in degrees (on both the horizontal and vertical plane), is generally shown with response polar plots at various frequencies and different resolutions. These are obtained from a large series of measurements, generally taken in Free Field, with the enclosure suspended at a discreet height from the ground (5/10 metres), according to the lowest frequency at which the manufacturer intends measuring the dispersion. The AES standard therefore foresees taking measurements with pink noise, showing the polar plots in various resolutions: octave, ½ octave and $^{1}/_{3}$ octave. The measurement is taken in so-called far field (or Fraunhofer region), at a distance of at least four times the largest dimension of the enclosure being tested. This avoids any influence due to the diffraction of the cabinet and minimizes the interference due to differences in the arrival of the signal at the measurement microphone, reducing so-called comb filtering to the utmost.

This is obviously the best method for finding the enclosure's directivity characteristics, which are extremely varied, according to the frequency: in fact, the IEC Standard also uses this method, suggesting polar plots with a resolution of $^{1}/_{3}$ octave.

In our case however, to **"find a synthesis of parameters that substitutes the Watt, at least from a practical point of view, if not in as far as "validity" is concerned**, it's necessary to obtain a "speedier" datum, in which the average of all these "polar" measurements is found and with this parameter simply define the sound coverage the enclosure is able to give.

Here again a suggestion is given by the **IEC Standard** which, as well as the polar plots, also foresees the specification of the radiation angle on both the vertical and horizontal planes. In the frequency band reproduced, this angle refers to the point in which the sound level drops by 10dB compared to that measured on-axis. Therefore, once the frequency at which (for a given off-axis angle) there's a drop of a maximum of 10 dB, this will become the frequency from which the enclosure's vertical or horizontal radiation angle will be indicated.

In other words, moving the measurement microphone through an arc, laterally in relation to the axis, we'll find the first point in which the level drops by 10dB at a given angle (e.g. 30°) and that point will correspond to a precise frequency (e.g. 1000 Hz). We can therefore say that, above 1000Hz, our enclosure has a radiation angle of 60° on the horizontal plane, if we've measured on this plane, or vertical if measured on the other. (There are two sides in relation to the enclosure's axis, therefore twice 30° = 60° total).

This interpretation of off-axis measurement, used to indicate dispersion, could be questionable, as I've already also mentioned for the Effective Frequency Bandwidth parameter. In fact, here again, the 10 dB limit according to which the radiation angle is stated seems to me to be rather "loose"; however, my previous line of reasoning holds valid, unless the change is decided on by a large group of technicians from this sector.

For example, as far as dispersion is concerned, as the designer with Outline, I personally adopted a slightly more complicated method for the calculations, which I believe better meets the need to achieve a good approximation of the area that can be covered by an enclosure with the required uniformity.

Ignoring the frequencies below 500 Hz, where it's improbable that there's a need to have greater coverage, as they are generally already highly "dispersed" by any enclosure, Outline's "data sheets" also give an **Average Dispersion** value, which is the arithmetic average of the frequency bands in the **500-1000-2000-4000 Hz** octaves. They also give another value, called **Nominal Dispersion**, obtained by calculating the average value for all the frequencies at $^1/_3$ octave from the frequency band **from 5000 to 16000 Hz**.

In this way, whoever has to design a sound reinforcement system for speech applications will make certain to consider the average dispersion figure, whereas, if the system is use for quality sound reinforcement of music, the system designer will also bear in mind nominal dispersion, which highlights the enclosure's behaviour at high frequencies. Instead of at -10dB, as foreseen by the aforementioned IEC Standard, the dispersion values are all taken at -6dB, in order to give a more restrictive datum for this parameter.

Taking this method as a starting point, one could simplify the declared datum, while nevertheless keeping the correct information necessary for system designers: <u>by calculating a further arithmetic average of the two averages described, calling it **Weighted Average Dispersion,** or even</u> **Effective Average Dispersion**, as is done with the Effective Frequency Bandwidth parameter. See Fig. 5.

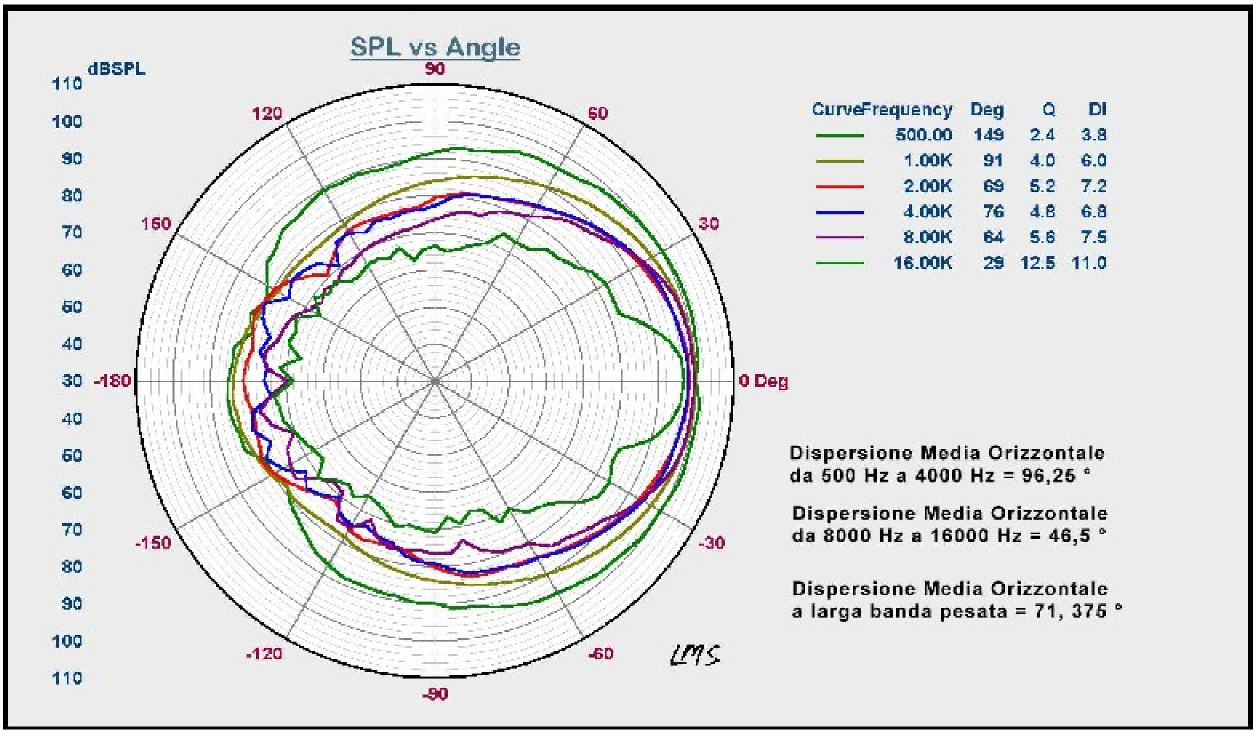

Fig. 5 – Graphs of the polar curves of an Outline enclosure's effective average wide-band dispersion.

Up until now, we've considered all the parameters from which the evaluation of a loudspeaker enclosure's **quantitative performance** could emerge, I've explained my proposals and the possible variations in "interpreting" the parameters measured according to the most widespread standards.

Before reaching the conclusions in the next issue, with the definition of a new method for the ***"quantitative evaluation of a sound reinforcement system"*,** I look forward to receiving readers' comments and suggestions.

For further information, I'll anticipate that I'll analyze another pair of parameters which in my opinion cannot be overlooked in relation to those already examined, POWER COMPRESSION and DISTORTION. I'll also analyze how, once it has been shown that the criterion of concise evaluation is valid for a single enclosure, it's also valid for systems or groups of enclosures, no matter what geometric configurations they're used in.