Impulse source versus dodecahedral loudspeaker for measuring parameters derived from the impulse response in room acoustics

Ricardo San Martín,[a)] Miguel Arana, Jorge Machín, and Abel Arregui
Acoustics Laboratory, Physics Department, Public University of Navarre, Campus de Arrosadía, 31006 Pamplona, Spain

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This study investigates the performance of dodecahedral and impulse sources when measuring acoustic parameters in enclosures according to ISO 3382-1 [*Acoustics–Measurement of room acoustic parameters. Part 1: Performance spaces* (International Organization for Standardization, Geneva, Switzerland, 2009)]. In general, methods using speakers as a sound source are limited by their frequency response and directivity. On the other hand, getting impulse responses from impulse sources typically involves a lack of repeatability, and it is usually necessary to average several measurements for each position. Through experiments in different auditoriums that recreate typical situations in which the measurement standard is applied, it is found that using impulse sources leads to greater variation in the results, especially at low frequencies. However, this prevents subsequent dispersions due to variables that this technique does not require, such as the orientation of the emitting source. These dispersions may be relevant at high frequencies exceeding the established tolerance criteria for certain parameters. Finally, a new descriptor for dodecahedral sources reflecting the influence their lack of omnidirectionality produces on measuring acoustic parameters is proposed.
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I. IMPULSE RESPONSE (IR) MEASUREMENT

From today's perspective, the apparatus Sabine[1] used for his famous research on reverberation is quite modest. The room was excited with multiple organ pipes and he used his ear as a measuring device in conjunction with a simple clock. With the development of the power amplifier in the 1920s, most measurement techniques became electrical. In acoustics, the ear was replaced by a microphone and sound level decay was analyzed with the help of electromechanical level recorders. Later, electric filters and speakers became an alternative to mechanical sound sources. More recently, the advent of the digital age has brought a new revolution. All kinds of signal processing (filtering, storage, evaluation, and presentation of results) can now be done using digital devices, generally more powerful, accurate, flexible, and especially more affordable than traditional equipment. However, the main difficulty of performing impulse response (IR) measurements in rooms remains the inability to create and emit real Dirac delta functions. The different alternatives listed below are intended to overcome this limitation, each with its own advantages and disadvantages.

A. Impulse sources—Pseudo-impulse signals

The simplest method for obtaining the IR of a system is to use sound sources that generate very short and powerful transient sounds. Consequently, they can provide sufficiently good approximations of ideal impulse excitation. The so-called impulse sources can distinguish several types, depending on how the impulse is generated: explosive sources, impulse sources with electrical origin, compressed air-based sources, and sources whose impulse generation is based on mechanical impacts.

The main advantage is their low cost and the simplicity of implementing the measurement system. They are truly appropriate when measuring equipment transport is limited, either due to the total weight or the inaccessibility of the room to be tested.

However, the common difficulty for these sources is to satisfy a suitable dynamic range for low frequencies. Pulse duration, by definition, is very short and it is very difficult to deliver enough energy to overcome the background noise. The source must also be reasonably omnidirectional, i.e., radiating equally in all directions. This feature is quantified by the requirement that variations in sound level in all directions should be less than ± 1 dB for frequencies up to 500 Hz and less than ± 6 dB at 4 kHz (see ISO 3382[2] Sec. 4.2.1, Sound Source). This approach, which ensures the possibility of comparing measurements made with different equipment, is essential due to the impossibility of artificially correcting responses obtained using directive sources.

Among explosive sources, pistol shots[3] have the longest history in room acoustics. Although still explicitly specified by the standard[2] as a possible alternative, their use has been in steady decline since the 1980s. Most of the energy emitted by such sources focuses on the region between 1 and 2 kHz. At lower frequencies, the level decreases about 13 dB with each third-octave band[4] as opposed to the desired spectrum to compensate for the presence of background noise whose higher value spectral components belong to the low frequencies. It is usually necessary to increase the powder charge for each shot manually, introducing a new variable that will affect

[a)]Author to whom correspondence should be addressed. Electronic mail: ricardo.sanmartin@unavarra.es

measurement repeatability, or use certain caliber pistols. On the other hand, they are not completely omnidirectional sources as small design differences between seemingly similar guns generate significant changes in the resulting directivity.[5] Due to its small diameter and because the angular distribution of particle velocity at the opening of a gun is not uniform, differences in levels have been found up to 10 dB, depending on the direction, for frequencies around 100 Hz.

On the contrary, if the powder charge is exploded directly—little cardboard cylinders with a small mass of powder and ignition fuse ("firecrackers")—the generated impulse is nearly omnidirectional. Its directivity index is, on average, around 1 dB for the octave bands between 125 Hz and 16 kHz.[6] In addition, both its time curve and spectral power are highly repetitive, resulting in levels above 115 dB (reference 1 pW) within the aforementioned range.

Meanwhile, sources that generate the impulse electrically (commonly called "spark generators") can, with proper electrode geometry, be omnidirectional for all relevant frequencies. However, their low power at low frequencies[7] has relegated them to laboratory scale models, where this limitation is not as important. Nowadays, they are rarely used in field measurements.

Of the sources operating with compressed air, balloon explosions are the most popular in the acoustic measurement field. Compared with the previous case, a 12 or 15 in. balloon (regular sizes) is a relatively bulky source. As a result, its explosion contains more energy at low frequencies, increasing the pulse duration. However, its main drawback is the low repeatability of the generated signals because the actual tearing of the balloon seems to be dominated by stochastic type processes, therefore impossible to predict. Some studies[4] have found level differences of up to 20 dB in explosions of balloons with similar characteristics. And when balloon directivity patterns are stable over repetitions, they do not radiate omnidirectionally. The degree of omnidirectionality improves with balloon size and at midrange frequencies.[8]

Finally, there are studies based on the performance of acoustic measurements using sound impulses generated by mechanical impact. Undoubtedly, the simplest source within this category is a slap.[9] Compared with other types of impulse sources, it has the lowest repeatability because the spectral characteristics of the generated impulse vary depending on how the slaps are generated. Common to all of them is their low power at low frequencies and sharp directivity. However, due to their obvious accessibility and immediacy, they can be used to detect unwanted acoustic phenomena such as echoes and flutter-echoes. In an attempt to eliminate the human factor, specifically designed sound sources following the same mechanical principle can be found.[10] Usually made of wood ("wooden clappers"), their characteristics vary depending on size and design of the device. However, problems still remain concerning both energy and radiation uniformity for low frequencies.

In an attempt to improve pulse generation repeatability, different researchers[11,12] have conducted studies using computer-generated pulses as the excitation signal. These signals, perfectly suitable for measuring speaker characteristics, may also be used for room IR measurements. However, the problem arises in the absence of powerful sources to deliver the signal. When all the emitted energy is concentrated over a very short period of time, there is risk of speaker damage trying to get an appropriate signal-to-noise ratio (SNR).

B. Deterministic signal processing

To overcome the difficulties described in Sec. I A, it is possible to distribute the energy supplied to the room over time and use digital processing techniques to calculate the IR signal. That way, the total radiated energy and dynamic range can be increased while reducing the influence of background noise. In these methods, prior knowledge of the excitation signal is needed to apply deconvolution techniques. Control over the excitation requires using loudspeakers as sound sources, but greatly increases the measurement technique repeatability allowing successive averaging to increase effective SNR.

Excitation signals with white spectrum allow the use of cross-correlation to obtain the IR of the system. Among them, the binary pulse sequences stand out for their ease of creation and processing in digital systems. The best known are the maximum length signals (MLS). They were introduced in the late 1970 s[13] and achieved great popularity due to the possibility of creating sequences using hardware and can be processed without leaving the time domain[14] using the fast Hadamard transform, saving both memory requirements and computational power, respectively. It is now clear that these advantages have faded. However, they have some disadvantages. Along with its vulnerability to distortion and temporal variation,[15] the purely white spectrum is considered a limitation nowadays. Often, to compensate for the increased presence of low frequency noise, a colored spectrum may be desirable for IR measurements in enclosures.[16,17]

Today, computers have the capacity to perform fast Fourier transforms, divisions in the spectral domain and inverse fast Fourier transform with great efficiency, which enables virtually any broadband signal—with enough energy for the entire acoustic frequency range—to act as a stimulus in conducting acoustic measurements.

Sweep signals, commonly known as "chirps," have been used historically in room acoustics. Back in 1967, Heyser[18] presented the development of a new technique for measuring transfer functions with the help of sweeps now recognized as time delay spectrometry. Designed initially to measure loudspeakers in reverberant environments, the technique can be applied to acoustic measurements of rooms or any other linear time-invariant system in general. Other researchers[4,19,20] have listed the advantages of using sweeps as excitation signals due to the high SNR obtained—the use of sweeps with instantaneous frequency varying exponentially over time (*logarithmic sweeps*) is recommended—and greater immunity from loudspeaker nonlinearities compared with other methods of measurement. The sweep synthesis from the frequency domain[21] has also been proposed, developing methods to generate specific excitation signals and increase the dynamic measurement range. Thus, modifying the speed with which the sine wave "sweeps" through, an excitation signal with a specific spectrum can be obtained, which could be used to compensate colorations introduced by the speaker or increase the SNR at certain frequencies where background noise is more present.

These methods, requiring the use of loudspeakers to emit signals, are limited by the frequency response and directivity of the speaker. For all sources discussed here, the loudspeaker is the only source where the frequency response can be corrected, but variations in direction cannot be removed and may become important at high frequencies.[22,23]

II. TEST ROOMS AND EXPERIMENTAL PROCEDURE

The results shown in Secs. III–V are from a series of experiments carried out by means of both measurements and simulations in different rooms. With the aim of selecting an appropriate recreation of typical conditions in enclosures for speech and music, where the acoustic parameter measurement standard ISO 3382[2] is usually applied, five different halls—all of them located in northern Spanish cities—were chosen. At the same time, a well-known hall among acousticians was included in the assessment, the Elmia Hall in Jönköping, Sweden, test room at the Second International Round Robin on Room Acoustical Simulation.[24]

The theaters and concert halls selected are listed in Table I along with basic information on each. They have reverberation times within the range of 1.0–2.9 s at mid frequencies, their volumes ranging from 4000 m^3 to 25 800 m^3 and three different shapes, i.e., rectangular, slightly fan-shaped, or semi-surrounded.

All measurements were made in accordance with the ISO 3382[2] standard. Three types of excitation signals were used: MLS signals (Sec. III) and logarithmic sweeps (Secs. III and IV) as deterministic signals, and firecrackers as impulse source (Secs. III and IV). Deterministic signals were emitted from two omnidirectional dodecahedron loudspeakers: the 01 dB's (Limonest, France) DO12 source (SA) and the Norsonic's (Lierskogen, Norway) type 223 source (SB), 450 mm and 500 mm in diameter, respectively. Regarding impulse sources, small explosive charges closed in cylindrical cartridges are activated by a wick placed at the cylinder base (see Fig. 1). Each cylinder is 37 mm high and the base diameter is 5 mm. The detonation wick (12 mm long) is fixed in one of the bases. The matter inside is a kind of powder, mainly a mixture of sulfur, charcoal, and potassium nitrate. 0.3 g of explosive mass is mixed with a less dense explosive and this is all enclosed by strongly pressed cylindrical cardboard.

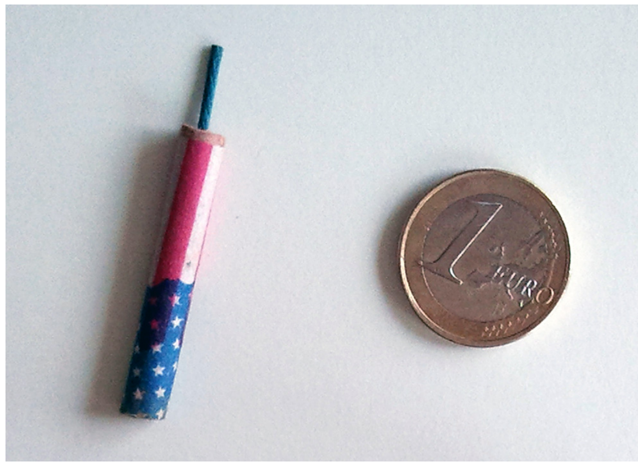

FIG. 1. (Color online) Photograph of a firecracker as used in the study next to a 1 Euro coin for size comparison.

Finally, the cylinder bases are covered with inert material strongly pressed in order to prevent a faulty explosion. They were fastened by thin threads[6] in order to avoid any influence of the placement device.

The parameters derived from monaural IR measurement, T_{30}, early decay time (EDT), D_{50}, C_{80}, T_S, G, were obtained through an omnidirectional microphone GRAS (Holte, Denmark) type 40AC. Different microphones were used in order to measure parameters related to spatial impression such as lateral energy fractions and interaural cross-correlations. The "Audio-Technica" (Tokyo, Japan) AT4050/CM5 condenser microphone with switchable omni and figure-of-eight polar pattern was used to measure J_{LF}. Binaural parameter, $IACC_E$, was calculated from binaural IRs measured with a "HEAD acoustics" (Herzogenrath, Germany) HSU III dummy head. WinMLS[25] sound card based software was used to generate, record, and analyze the signals.

As for the simulations (Sec. V), four different *omnidirectional* types of sources were surveyed and tested. Three of them were different commercial dodecahedron sources: the "omnipower" sound source B&K (Brüel & Kjaer, Naerum, Denmark) 4296 (S1), 350 mm in diameter, and two Norsonic sources, the 500 mm diameter type 223 (S2) and the 450 mm diameter type 270 (S3). The remaining source (S4) was developed at the Institute of Technical Acoustics in Aachen, Germany.[26] It consists of a three-way measurement loudspeaker in which a subwoofer is used to achieve the required sound power at low frequencies and two specially designed

TABLE I. Basic information of selected enclosures.

Hall, Location	Volume (m^3)	Seats	T_{30mid}[a] (s)	Shape
Sarasate Theatre, Pamplona, Spain	4000	480	1.0	Rectangular
Bretón Theatre, Logroño, Spain	6300	988	1.3	Rectangular
Elmia Hall, Jönköping,[b] Sweden	11 000	1100	2.2	Fan
Baluarte Concert Hall, Pamplona, Spain	20 000	1568	1.9	Rectangular
Kursaal Concert Hall,[c] San Sebastián, Spain	18 000	1806	1.9	Rectangular
Mozart Concert Hall, Zaragoza, Spain	25 800	1992	2.9	Semi-surround

[a]According to ISO 3382 (Ref. 2).
[b]Simulated only.
[c]Measured only.

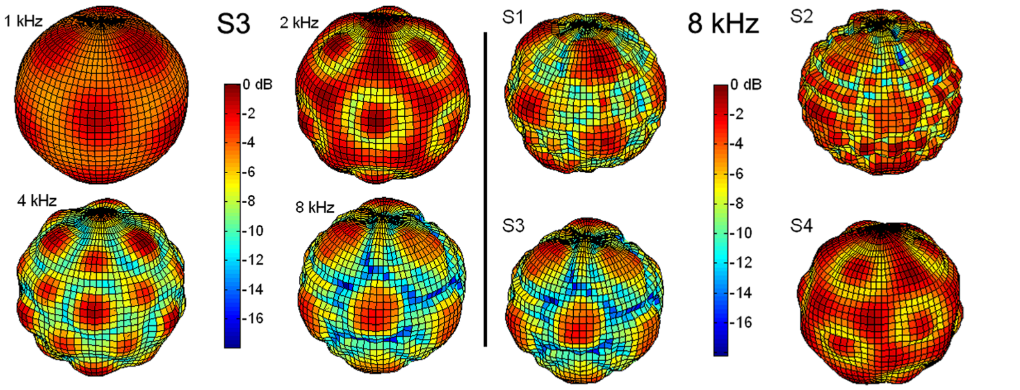

FIG. 2. (Color online) Octave-band directivity balloon plots for S3 at 1, 2, 4, and 8 kHz (left) and S1, S2, S3, and S4 at 8 kHz (right). Normalized levels at 0 dB with maximum for each source and frequency band as reference.

dodecahedron speakers with different diameters—260 mm and 90 mm for measuring mid and high frequencies, respectively—are used to improve the omnidirectional sound radiation in comparison with what is obtained using conventional singular dodecahedron measurement devices.

So as to simulate the directional pattern of the four sources, the radiated fields were measured in an anechoic chamber. A computer-controlled measurement procedure was used to obtain a half-sphere of measured data in 5° angular increments between adjacent points. The full sphere was built by applying symmetry rules. Graphical representation of directivity patterns obtained for S3 at 1, 2, 4, and 8 kHz octave-bands along with 8 kHz octave-band directivity pattern for all the sources are shown in Fig. 2. For S3, the progressive loss of omnidirectionality can be clearly seen as the frequency increases. Also, for the same frequency band, notable differences arise between the sources used. It might even be advisable to increase the spatial sampling density for three of the sources (S1, S2, and S3). However, this resolution—data every 5 deg—is the maximum currently supported by the simulation software such as the one used here (Odeon room acoustic software[27]) and is sufficient to compare the differences between the sources and to verify the improved better performance of source S4.

III. GENERAL REMARKS: SNR AND IR(*t*)

The first experiment analyzed the SNR obtained with impulse sources, MLS signals, and logarithmic sweeps. It was evaluated by the effective decay range (EDR). This parameter describes the available decay curve range to obtain the reverberation time. The crosspoint between the decaying response and the background noise is detected using the iterative algorithm proposed by Lundeby.[28] Next, the truncation point is determined using a safety margin above the crosspoint, typically 5 dB. The final slope of the truncated response is calculated by linear adjustment and a compensation for the energy lost by truncation is estimated assuming exponential decay to infinity.[25] This indicator is more reliable than pure SNR obtained from the actual IR, because it normally considers the final part (typically 10% of the signal) to estimate the background noise, and its accuracy depends on this portion actually coinciding with the portion of the signal containing noise.

The value of this indicator obtained in each octave band for ten emissions of all three types of signal in a source-receiver position (distance 20 m) is shown in Fig. 3, together with their associated standard deviation in Fig. 4. Special attention was paid to the specifications assigned to the two types of deterministic signals, sweeps and MLS, so that the time that each frequency band was excited was equal for both cases. In each measurement, 7 averages were made with a cycle length of 5.46 s in the case of the sweeps (from 16 Hz to 24 kHz) and 14 averages of 2.73 s cycle length in the case of MLS ($n = 17$, with sampling frequency of 48 kHz), where 40 s is the total duration for each measure. The output level was optimized by trial and error for use with MLS sequences looking for minimum IR contamination. The emitted sweep could have been raised by at

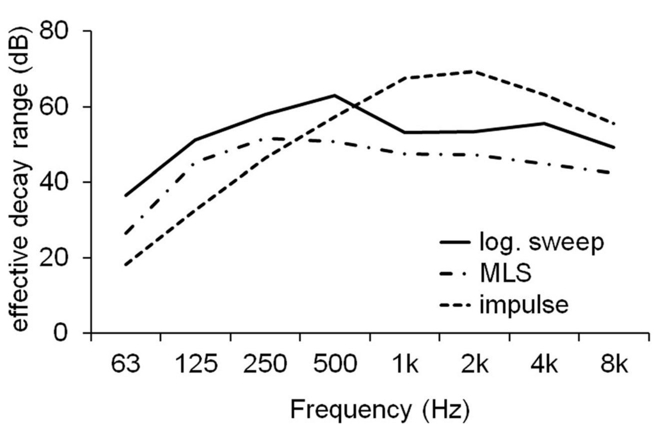

FIG. 3. Average EDR (dB) values for the three techniques discussed.

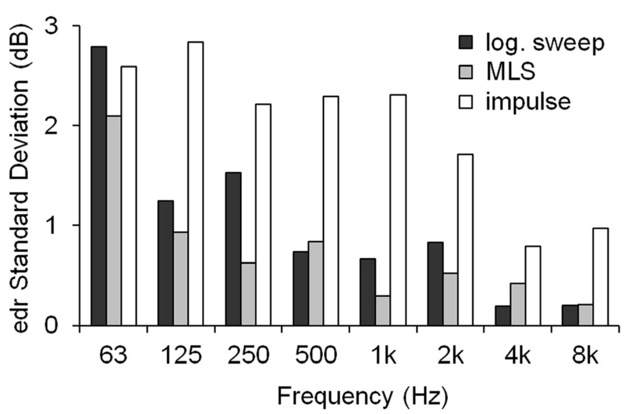

FIG. 4. Standard deviation of EDR (dB) for the three techniques discussed and a total of ten emissions with each technique.

least 10 dB without causing amplifier clipping, thereby increasing the SNR.

First, the higher SNR achieved by using logarithmic sweeps with respect to the corresponding MLS signals should be noted, although emission time was the same. In all frequency bands it exceeds the figure obtained with MLS signals between 6 and 12 dB. Furthermore, the SNR between 1 and 8 kHz for pseudo-impulse signals significantly improves the two other techniques. However, their low power at low frequencies is reflected in the EDR obtained. This is disadvantageous when it comes to calculating the acoustic parameters of a room, especially reverberation time, T_{30}. In addition, techniques based on deterministic signals allow the SNR to rise by increasing both the source sound power and/or the duration of the excitation signal. This ability to improve the quality of an IR measurement, involving the use of electroacoustic equipment for signal generation, disappears in the case of using impulse sources such as those used in this study, where just the source-receiver distance can be an insurmountable obstacle.

Moreover, the standard deviation of both types of deterministic signal is very low, unlike the pseudo-impulse signals, as shown in Fig. 4. Due to the different sound power from each explosion, firecrackers have greater dispersion. This effect is evident at low and medium frequencies. Nevertheless, this greater deviation is not a crucial aspect, provided enough SNR are obtained.

Analysis below looks at the degree of repeatability in the IR curves, IR(t), for the different emissions of the three types of excitation signal. In the case of sweeps and MLS signals, the curves obtained are indistinguishable, something expected as a deterministic type emission. On the contrary, for the case of pseudo-impulse signals, nondeterministic signals, certain dispersion in the results is obtained. Figure 5 shows the ten normalized energy decay curves[29] obtained by the impulse sources for the same source-receiver position. As can be seen, there is a certain variability resulting in dispersion in the parameter values derived from these curves. It is noted that these curves have been normalized to the unit at the y axis, so that the different sound power from different firecrackers is not represented, only the different shape of the IR. Similar curves, with respect to the lack of repeatability, are obtained when octave band filtering.

Figure 6 shows the average values for ten measurements with sweeps, MLS, and pseudo-impulse signals. The average

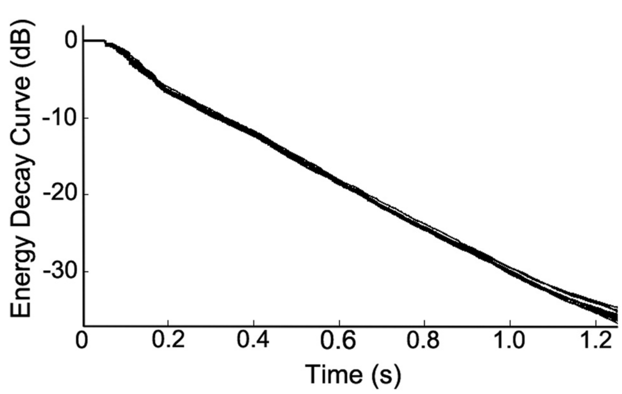

FIG. 5. Normalized energy decay curves for ten IRs measured by impulse source in the same source-receiver position.

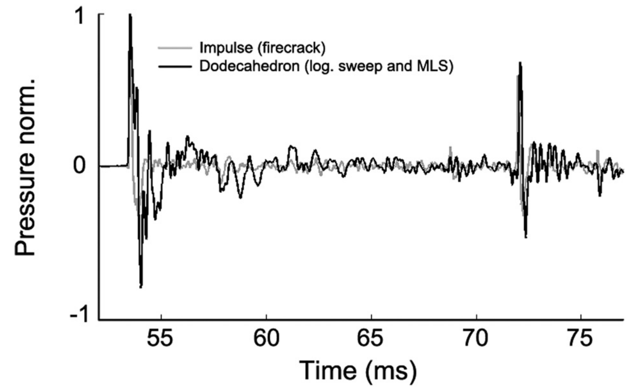

FIG. 6. Average IR obtained by the two types of source (detail of the first 20 ms).

curves of logarithmic sweeps and MLS signals are completely matched, so that just one legend groups them, referring to the electroacoustic source used to obtain them.

The fact that the IR obtained from pseudo-impulse signals are not as extremely repetitive as any obtained from deterministic signals leads us to conclude that these techniques are preferable. We must remember, however, that the deterministic signals emission technique requires the use of amplifiers and loudspeakers, implying added variables which involve other dispersion factors in the results, as shown in Sec. IV.

A remarkable circumstance can be highlighted concerning the IR(t) shape as follows. If we observe a zoom of the IR covering the arrival of the direct sound and the first reflection, different "fluctuations" around direct sound are perceived in the case of deterministic signals. Given the small time interval between them—tens of milliseconds, corresponding to centimeters of path differences—such fluctuations can only be attributed to delays between the emissions of the different speakers making up the dodecahedral source used. This implies certain "artificiality" of the IR for deterministic signal techniques, attributable to the experimental device, something from which impulse sources technique is exempt. Ultimately, the final results, including all the variables that cause results dispersion, will allow us to compare the reliability or solidity of the different techniques.

IV. FIRECRACKERS VS DODECAHEDRAL SOURCES

The next objective was to quantify the dispersion of the acoustic parameters in the results for both impulse sources and dodecahedral sources. In the former, this deviation around a mean value is caused by the lack of repeatability of the device used as described in Sec. III. In the case of using deterministic signals emitted by dodecahedral sources for obtaining the IR, this problem disappears. However, its potential lack of omnidirectionality adds a new variable to the measurement chain that was not present in the previous method: source orientation.

In a central and representative point of the audience in selected enclosures, the average values and dispersions of acoustic parameters were compared obtained from ten explosions and two commercial dodecahedral sources (SA and SB). Given the particular symmetry of the dodecahedron, 24

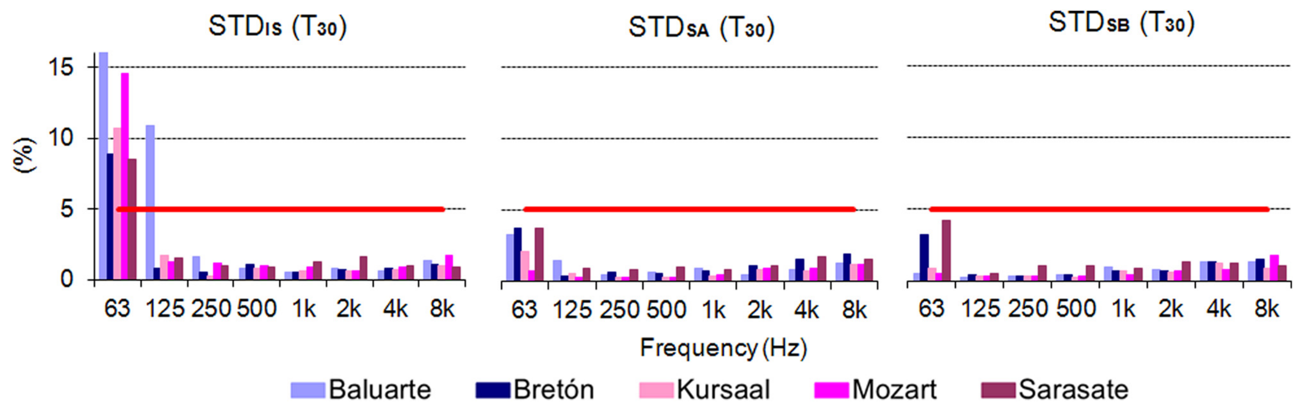

FIG. 7. (Color online) STD of T_{30} parameter in the five enclosures (% relative to the mean value) for the impulse source (IS) and the two commercial dodecahedral sources (SA and SB).

orientations covering a rotation of 120° in 5° steps were considered sufficient. The average values of the acoustic parameters analyzed were, with few exceptions, similar or within the range set by its own deviation.

Since the object of our interest in this section is to compare the parameter deviations in the five rooms for the three types of sources, Figs. 7–9 show the dispersions obtained for three of the parameters where the most significant results were obtained: T_{30}, C_{80}, and G. Dispersion caused by impulse sources has been designated as $\mathrm{STD}_{\mathrm{IS}}$. In turn, the dispersion caused by the orientation of the commercial dodecahedral sources is denoted $\mathrm{STD}_{\mathrm{SA}}$ and $\mathrm{STD}_{\mathrm{SB}}$. The graphs also highlight the just noticeable difference (jnd)[30,31] value associated with each parameter. The jnd, minimal variation in the acoustic parameter value that humans can perceive, is currently accepted by the scientific community as the best indicator to quantify the subjective perception caused by a variation in that parameter. The jnd therefore offers guidance on the quality of a measurement, determining the accuracy with which the acoustic parameters should be obtained.

With respect to T_{30} (Fig. 7), it should be noted that the deviations obtained for most of the frequency bands were very low. For dodecahedral sources, they are around those associated with the technique and always less than 1 jnd. It is clear that the parameter is not particularly sensitive to variations in the measurement chain by the high integration time required to derive it. Its biggest challenge is to achieve sufficient SNR at low frequencies. This is what happens in some rooms at the 63 Hz frequency band with dodecahedral sources and in all rooms at the same band (even at 125 Hz for the Baluarte auditorium) when using impulse sources. The main problem of the explosive source used is thereby reflected again: Its low power at low frequencies in some cases may be insufficient.

The C_{80} parameter behaves very similarly to other parameters studied (EDT,D_{50},T_S,…) and is therefore represented in Fig. 8. The deviation caused by the impulse source is clearly superior to the use of dodecahedral sources in the low frequency area, while inferior, in most cases, to half the jnd value of the corresponding parameter. In the 1 kHz band, both techniques' dispersions are comparable. However, with deterministic signals the dispersion is remarkably greater than that of firecrackers from the 2 kHz band, especially for source SA, which exceeds the value of jnd in several of the rooms analyzed.

The results obtained for the G parameter (Fig. 9) are particularly interesting. Although similar conclusions to the other parameters could be drawn concerning the comparison between dispersions, it is evident that their magnitude is higher for all frequency bands and halls, nearly three times the value of the jnd for source SA at very high frequencies. Its particular derivation, reference-free field sound pressure level at 10 m distance is estimated by windowing the first

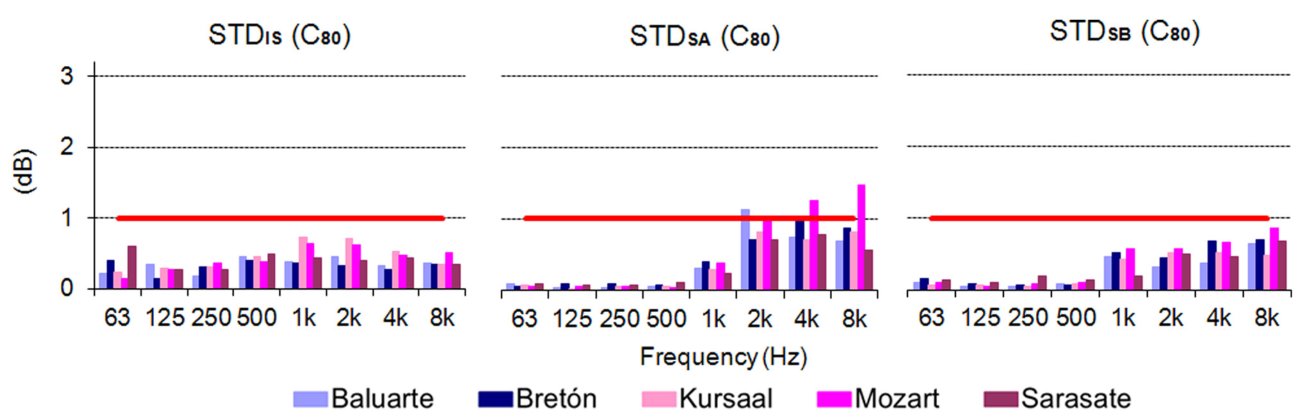

FIG. 8. (Color online) STD of C_{80} parameter in the five enclosures (dB) for the IS and the two commercial dodecahedral sources (SA and SB).

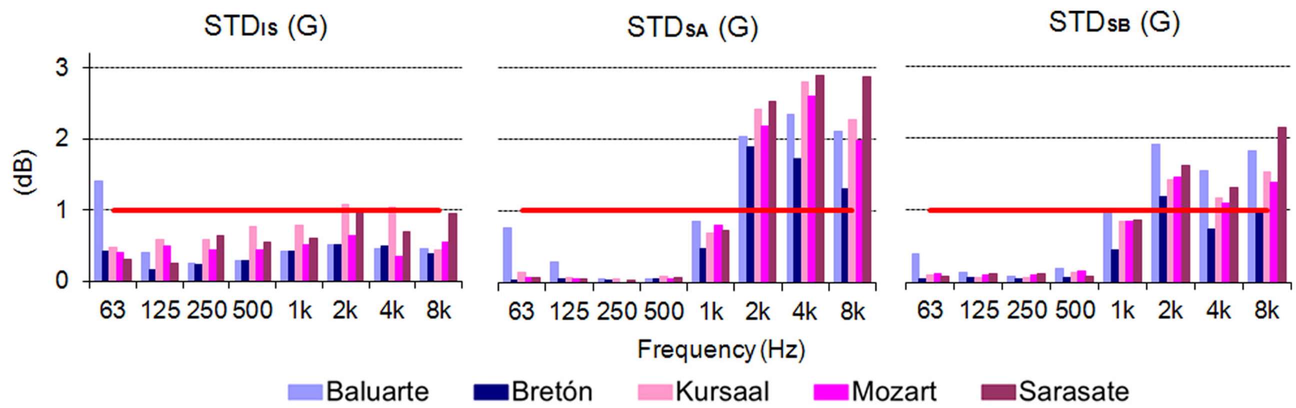

FIG. 9. (Color online) STD of *G* parameter in the five enclosures (dB) for the IS and the two commercial dodecahedral sources (SA and SB).

5 ms of the actual IR we want to compute from—makes the parameter extremely sensitive to possible variations in the measurement chain, especially any causing changes in the direct sound levels from which information is extracted to obtain the reference-free field level at 10 m from the source. This method to obtain *G* is not specified in the standard. In fact, comparison with properly calibrated measurements have demonstrated that this method will normally provide *G* values that are too high at low frequencies due to insufficient window length, fairly good *G* values at mid frequencies, and too low *G* values at high frequencies due to the influence of the transducer's immediate surroundings.[25] However, it is widely used to estimate the range of the parameter between seats located close to and far from the stage.

If we group the results for each enclosure at low (average of 125 and 250 Hz), mid (500 Hz–1 kHz), and high (2–4 kHz) frequencies and study their deviation for the other parameters relative to the jnd, the conclusions are similar.

Table II summarizes all the results in the dispersion comparison. Blank cells indicate that the dispersion is less than half a jnd. Otherwise, a single measure would provide "*a reliable result for the parameter value*" in identifying the standard deviation (STD) as the parameter that characterizes the dispersion of the results of a measurement that can be attributed to the measurand and stating the related 95% confidence interval, the maximum permitted difference threshold should be twice the STD of the measurement. So, the STD should not be larger than half the jnd of the parameter under discussion. Cells with a value in italics indicate that the dispersion is between 0.5 and 1 jnd. Any with a value in bold demonstrate that deviation exceeds 1 jnd.

This way of representing the results leads to a very clear conclusion. It clearly shows worse impulse source behavior for low and mid frequencies, but also their best behavior at high frequencies. For all parameters at low and mid frequencies, the pseudo-impulse signal technique in 24 cases exceeds the value of half a jnd (although only once—T_{30} in Baluarte at low frequencies—is the limit of 1 jnd exceeded) for only five times the dodecahedral source SB, while source SA maintains its dispersions below this tolerance criterion in all cases.

However, at high frequencies, while the impulse source STD_{IS} exceeds the value of half a jnd in 18 cases, 2 of them above a jnd, this figure increases to 23 cases, 5 above a jnd, for the STD_{SB} of source SB, very high performance

TABLE II. Relative values of STD, grouped in low (125–250 Hz), mid (500 Hz–1 kHz), and high (2–4 kHz) frequencies for all enclosures and parameters analyzed. Reference: jnd.

STD ref: jnd		Impulse source			SA			SB		
		Low	Mid	High	Low	Mid	High	Low	Mid	High
Baluarte	T_{30}	**1.26**								
	EDT						*0.58*			
	D_{50}						**1.09**			
	C_{80}						*0.93*			
	T_S						**1.17**			
	G						**2.19**		0.59	**1.73**
	J_{LF}		*0.66*	*0.64*			*0.91*			0.57
	$IACC_E$	*0.98*					*0.93*			0.67
Bretón	T_{30}									
	EDT		*0.55*	*0.97*			*0.80*			
	D_{50}	*0.54*	*0.54*	*0.58*			**1.46**			0.75
	C_{80}						*0.87*			0.56
	T_S						*0.89*			
	G						**2.19**		0.59	**1.73**
	J_{LF}		*0.53*				*0.85*			*0.62*
	$IACC_E$	*0.98*					*0.93*			*0.67*
Kursaal	T_{30}									
	EDT	*0.61*	*0.64*	*0.70*			*0.89*			*0.77*
	D_{50}		*0.73*	**1.00**			*0.87*			
	C_{80}		*0.59*	*0.62*			*0.76*			*0.52*
	T_S		*0.64*	*0.50*			*0.75*			
	G	*0.58*	*0.78*	**1.05**			**2.61**			**1.30**
	J_{LF}		*0.69*	*0.63*			**1.00**		*0.51*	*0.59*
	$IACC_E$		*0.52*	*0.52*			**1.06**			*0.85*
Mozart	T_{30}									
	EDT	*0.61*					*0.50*			*0.62*
	D_{50}			*0.52*			**1.61**			*0.86*
	C_{80}		*0.51*	*0.55*			**1.14**			*0.61*
	T_S		*0.65*	*0.59*			**1.52**			*0.77*
	G						**2.39**		*0.50*	**1.28**
	J_{LF}			*0.63*			*0.80*			
	$IACC_E$						**1.81**		*0.53*	*0.87*
Sarasate	T_{30}									
	EDT			*0.62*			**1.01**			*0.77*
	D_{50}		*0.75*	*0.60*			**1.27**			*0.60*
	C_{80}						*0.74*			
	T_S						*0.69*			
	G		*0.58*	*0.84*			**2.71**			**1.46**
	J_{LF}	*0.71*	*0.56*	*0.66*			*0.59*			
	$IACC_E$									

loudspeaker, and 34 for the source SA, 16 above a jnd, which also meets ISO requirements concerning directivity. For the latter, the dispersions are above the established tolerance criteria for all parameters and rooms except T_{30}, which always has a very low value with the three techniques, and the $IACC_E$ from the Sarasate Theater whose value (0.47) is touching the limit.

V. PROPOSED NEW DESCRIPTOR FOR DODECAHEDRAL SOUND SOURCES

Given the analysis so far, source rotations through three angular positions and the posterior averaging of subsequent measurement results—ISO 3382[2] requirement if the source directivity is found to have a significant effect on measured acoustic parameters—should not be discarded even in the case of dodecahedron loudspeakers having the ISO standard qualification. If the aim is to make it possible to compare measurements among different teams and pieces of equipment more accurately, the standard probably relaxes omnidirectional requirements at higher frequencies. A far-reaching aim would be to consider alternatives for assessing source omnidirectionality[32] since authors suspect that the ISO classification method may lead to misguided conclusions about omnidirectional performance owing to its implementation by using single-plane measurement arcs.[23] In this section, we propose a new descriptor based on the classical definition of *directivity*. It considers the sound radiation in a three-dimensional framework, in contrast to the traditional method which only takes into account the differences found in a single plane of measurement, usually one that divides the sphere through the middle and, due to the particular symmetry of dodecahedral sources, does not cross any of the geometric centers of the 12 speakers that form it.

The usual procedure for measuring the radiated field of a sound source in an anechoic chamber typically includes mounting the source on a rotating platform and taking data with a microphone array or a moving microphone around a quarter-circle.

If the increase for both the rotation angle of the platform and for each microphone position is $\Delta\theta = \Delta\Phi = \pi/36$, a set of 2664 sound levels—$L_{\theta,\Phi}$ (f), θ (elevation), and Φ (azimuth) as microphone and source position, respectively—is obtained for each frequency range f measured. Thereby, θ varies from $-\pi/2$ to $\pi/2$ radians (elevation) in 37 steps by Φ from 0 to $(2\pi - \pi/36)$ (azimuth) in 72 steps. For values of θ from $-\pi/2$ to 0, the source in the measurement device is reversed or, depending on the symmetry, the second half of the sphere is directly reconstructed.

In order to find an average energy level, all the $L_{\theta,\Phi}(f)$ values are weighted by a factor $w_{\theta,\Phi}$ determined by the surface integration of each (θ,Φ) section sampled by a microphone position normalized with respect to the area of the full sphere,

$$\begin{aligned} w_{\theta,\phi} &= \frac{1}{4\pi r^2}\int_{\phi-\Delta\phi/2}^{\phi+\Delta\phi/2}\int_{\theta-\Delta\theta/2}^{\theta+\Delta\theta/2} r^2\cos(\theta)\,d\theta\,d\phi \\ &= \frac{\Delta\phi}{2\pi}\sin\left(\frac{\Delta\theta}{2}\right)\cos\theta \end{aligned} \tag{1}$$

for

$$-\frac{\pi}{2} < \theta < \frac{\pi}{2}.$$

This definition is consistent for all values of Φ. However, the peculiarity of the areas sampled at both poles leads us to change the integration limits for the first integral from $-\pi/2$ to $(-\pi/2 + \Delta\theta/2)$ and $(\pi/2 - \Delta\theta/2)$ to $\pi/2$, respectively,

$$w_{-\pi/2,\phi} = w_{\pi/2,\phi} = \frac{\Delta\phi}{2\pi}\sin^2\left(\frac{\Delta\theta}{4}\right). \tag{2}$$

Thus, the average energy level weighted by area, $\overline{L_w(f)}$, is given by

$$\overline{L_w(f)} = 10\log\left(\sum_{\theta=-\pi/2}^{\pi/2}\sum_{\phi=0}^{2\pi-\pi/36} w_{\theta,\varphi}10^{L_{\theta,\phi}(f)/10}\right). \tag{3}$$

The new proposed descriptor denoted $DI_{L\max}$—*directivity index in the direction of maximum level*, generally coinciding with the geometric center of one of the speakers that form the dodecahedron—is expressed in dB and defined for each frequency band f as follows:

$$DI_{L\max}(f) = \max[L_{\theta,\phi}(f)] - \overline{L_w(f)}. \tag{4}$$

Figure 10 shows the new descriptor defined plus the descriptor based on ISO 3382[2] requirements which, remember, is calculated by averaging over 30° sliding arcs.

Although the trends are similar, close inspection reveals differences for the deviations obtained by each of the descriptors. For example, if we look at the 8 kHz octave band, in a classification to order the four sources from best to worst performance regarding omnidirectional radiation,

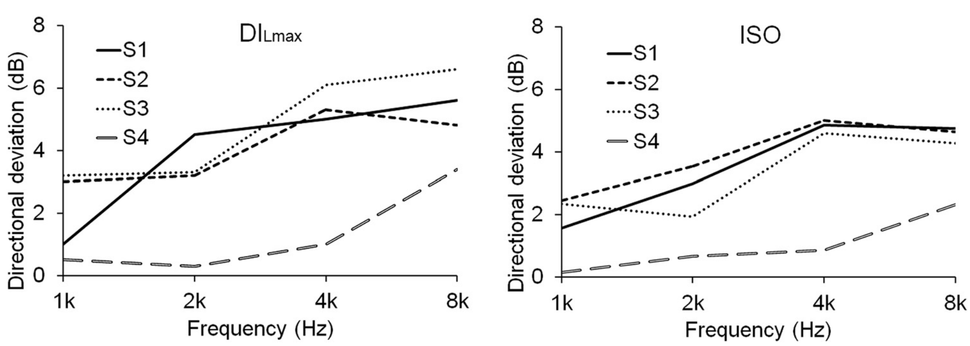

FIG. 10. Directional deviations (dB) based on the two descriptors, ISO 3382 (Ref. 2) (left) and $DI_{L\max}$ (right), for the frequency bands from 1 to 8 kHz and the four sound sources considered.

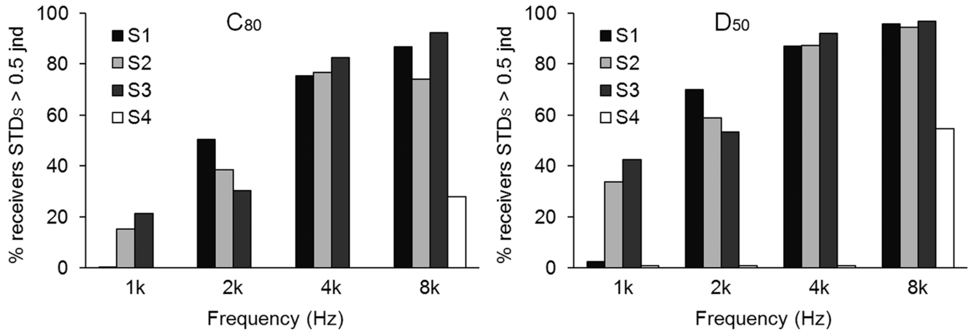

FIG. 11. Percentage of receivers with $STD_S > 0.5$ jnd for C_{80} (left) and D_{50} (right) depending on the type of source used.

S4 would always come out best. However, the organization of the other sources would be different depending on the criteria used: S3-S2-S1 if we use the ISO definition and S2-S1-S3 applying the new descriptor. Similar conclusions, with regard to possibly managing sources differently, can be drawn when analyzing the rest of the bands.

The purpose of descriptors should be to faithfully reflect, within the measured acoustic parameters, the impact of directional deviations they represent. With the help of grid simulations, it is possible to determine the number of receivers affected in the analyzed auditoriums[23] with uncertainties due to the orientation of the source exceeding a specified value. Considering the percentage of receivers with STD_S more than half jnd, as established tolerance criteria, representation of the impact for each type of source would result in parameters D_{50} and C_{80}, for instance, can be seen in Fig. 11. For the three commercial dodecahedral sources, the percentage of receivers whose measurement would create an uncertainty greater than variations noticeable by the listener exceeds 30% for both parameters from 2 kHz octave band. It is also evident, as could be predicted, that D_{50} parameter is more sensitive to the "source orientation" variable, as the percentage of receivers affected is always greater than in the case of C_{80}.

Moreover, this representation should match one obtained in Fig. 10. While at first glance it is not clear which descriptor best reflects the influence that the lack of source omnidirectionality produces when measuring acoustic parameters derived from the IR in rooms, the representation in Fig. 12 clarifies this better. It uses a scatter plot to depict the

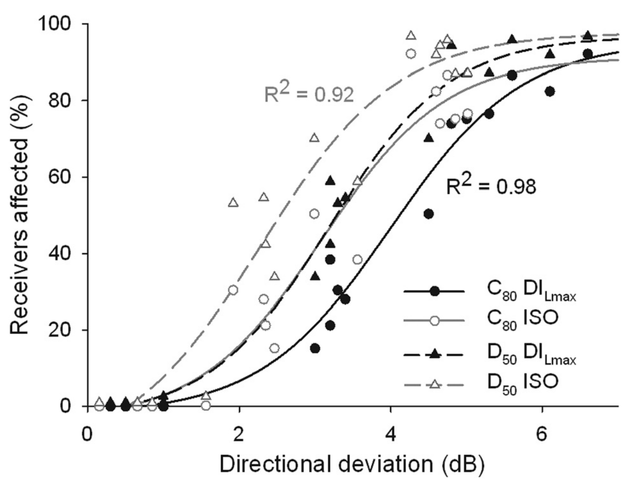

FIG. 12. Percentage of receivers with $STD_S > 0.5$ jnd for C_{80} and D_{50} against deviations based on each of the descriptors: ISO 3382 (Ref. 2) and $DI_{L\max}$.

percentage of receivers affected, for all sources and frequency bands analyzed, with respect to the directional deviation in dB that each descriptor reveals.

It can be established that a directional deviation less than 2 dB, regardless of the descriptor used to define it, will practically not affect any measured receiver. However, from 4 dB, the source directivity causes uncertainties above criterion in over 70% of receivers. It should be noted that in previous studies,[22,23] the central audience areas were the most affected, namely, receivers with a high probability of being chosen when applying ISO 3382[2] to evaluate the rooms' acoustical quality.

The main difference between the two descriptors analyzed lies in their ability to predict the uncertainty associated with the lack of omnidirectionality of dodecahedral sources. The trend lines show a correlation of 0.92 between directional deviation and impact in the case of the ISO descriptor. The best correlations are obtained with $DI_{L\max}$, reaching values of 0.98 in both cases (D_{50} and C_{80}).

VI. CONCLUSIONS

In this paper we have analyzed and quantified the influence of the type of sound source (impulse source and dodecahedral speaker) when measuring acoustic parameters in enclosures. The type of emitting source involves the use of both different signals for the excitation of the room and various methods to obtain the IR. All alternatives analyzed had advantages and disadvantages. On one hand, the firecrackers do not guarantee the extraordinary repeatability of techniques based on deterministic signals. On the other hand, the orientation of dodecahedral loudspeakers required to radiate sweeps and get the IR through signal processing may be an important factor that significantly increases the measurement uncertainty. However, if both techniques are applied correctly, similar values are obtained for the acoustic parameters derived from IR. Therefore, choosing between the two depends on the desired precision or the ability to solve problems like the presence of background noise.

The biggest drawback of firecrackers is the low SNR obtained for low frequencies and long distances. Moreover, the SNR cannot be improved by averaging techniques, as in the case of using dodecahedral loudspeakers and deterministic signals as excitation signals. This is disadvantageous when calculating room acoustic parameters, especially the reverberation time and, particularly, in the 63 Hz octave band. Nevertheless, at usual distances and typical excitation signal durations, both its best SNR and its omnidirectionality for bands from 1 to 8 kHz—frequencies where most commercial sound sources begin to lose that property—are remarkable and this technique avoids subsequent dispersion results due to variables such as the orientation of the source. Furthermore, detailed analysis of the IR obtained with a dodecahedral source reveals certain "echoes"—repetitions of lesser intensity at time intervals of tens of milliseconds—of both the direct sound and the early reflections, attributable to delays between emissions from its different speakers, circumstances that are not observed in the IR obtained from pseudo-impulse signals.

A direct comparison in the five enclosures between the two excitation signals analyzed in detail, firecrackers and sweeps, certainly reflected satisfactory results for the impulse source. Except at very low frequencies, the associated deviation is relatively constant in frequency and rarely exceeds the limit of 1 jnd. However, if we consider the effect of the orientation of the dodecahedral source in the dispersion of the results obtained using sweeps, STD_S values above this limit are common at frequency bands equal to or higher than 2 kHz, especially for one of the analyzed sources.

Finally, it was shown that the classification method proposed by ISO 3382[2] for describing source directivity—deviations in decibels of 30° sliding arcs averaged levels referenced on the average energy in the measurement plane—can lead to erroneous conclusions about their behavior and a new descriptor based on a three-dimensional approach is proposed. The denoted $DI_{L\max}$ showed a correlation of 0.98 between directional deviation and impact, considerably improving the value of 0.92 obtained for the case of the ISO 3382[2] descriptor.

ACKNOWLEDGMENT

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