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ON THE ACCURACY OF THE ASSESSMENT OF ROOM ACOUSTICS PARAMETERS USING MLS TECHNIQUE AND NUMERICAL SIMULATION

Márcio H. A. Gomes
Samir N. Y. Gerges
Laboratório de Vibrações e Acústica, Departamento de Engenharia Mecânica, Universidade Federal de Santa Catarina, Florianópolis, S.C. - Brazil, Cx. Postal 476 - 88049-000

Abstract. *Advances in psychoacoustics through the years have made it possible to evaluate the acoustical quality of a room, based on several numerical parameters that have been developed. Even though there is not a total agreement about which parameters are truly important, some of them are accepted by most of the acousticians. When designing a concert hall, ray-tracing softwares can be used to calculate some of these parameters. However,some input data carry some uncertainties. Available data for absorption coefficients have to be used carefully and, beyond that, there is not a standardized recommendation for measuring diffusion coefficients, nor a data base for this parameter. This paper presents a comparison of a room simulated with one of the commercial softwares available (RAYNOISE), and the measured parameters in this same room, using Maximum Length Sequence (MLS. Discrepancies over the results are investigated and analyzed.*

Key-words: *MaximumLength Sequence, Room acoustics, ray-tracing*

1. INTRODUCTION

Room Acoustics has turned to be a scientific field of potential development, and research. It has been almost a hundred years, since Wallace Clement Sabine started to study the subject from a formal and practical point of view. However, only from 25 years or so, it is possible to note some substantial knowledge of the process, due, mainly, to extensive research in the field of psycho-acoustics, and to the development of computational tools for measuring and simulating room acoustical impulse responses.

Nowadays, even though there is not a total agreement over the parameters that are really important for the evaluation of the acoustical quality of a room, it is common sense that all the important parameters can be calculated from the impulse response between a pair source-receiver in the room. Today, there are several ways to measure, and calculate the impulse response of a room.

Impulse responses can be measured with a microphone standing on a fixed position, capturing, in that position, the response to a sound burst produced by a source on another point. This sound burst must approximate a Dirac Delta in the time domain. For this purpose, shotguns, electrical sparks or hand claps, for instance, can be used to produce the desired excitation. Most of these sources, however, do not have an uniform frequency response, are not exactly repeatable, and can lead the sensor (the microphone) to a nonlinear range. All these problems contribute to lower the dynamic range of the final result, and the signal-to-noise ratio, as well. For these reasons, scientists were encouraged to look for alternative ways of measuring impulse responses.

In the 70's, two techniques were developed for measuring impulse responses of several kind of systems: the Time Delay Spectrometry, and the one that uses a "Maximum Length Sequence"(MLS) as the driving signal (Schroeder, 1965; Kuttruff, 1979; Borish and Angell, 1983; Schroeder, 1979; Rife and Vanderkooy, 1989; Chu, 1990). The MLS, classified as a pseudo-random signal, takes advantage of its auto-correlation function, constituted by a periodic pulse. If the system considered is time invariant, it can be shown that the cross correlation of the driving signal and the system's response is the impulse response of the system. Because the driving signal is known, only a single channel is necessary in the measurement system, and, due to the cross-correlation operation, the non-correlated noise is "rejected". This fact leads to a high signal-to-noise ratio, one of the main characteristics of the technique. One important advantage that this signal was supposed to present, is that, as the uncorrelated noise is mostly "rejected", it would be possible to measure impulse responses during the regular activities of a room, with the room occupied; a concert or a speech, to say. This is not exactly true, apparently, because the environmental conditions of a room occupied are not steady, and the system can no longer be considered as time invariant. There is a controversy about this, among some acousticians (Griesinger, 1996), but the fact is that MLS technique has become more and more popular as a measuring tool of room impulse responses. Measurements of occupied halls have been done right before the events.

There are three theories developed for the acoustics of enclosed spaces. Two of them are useful for the prediction of room impulse responses; the wave acoustics theory, and the geometrical acoustics theory. The former is mathematically, and physically, more rigorous, and can be considered as a particular case of a more general vibration theory. Thus, only recently this theory has been used in some practical applications. Traditionally the referred theory has been used to understand some qualitative aspects about the behavior of sound within an enclosure. A recent work, presented by Terai & Kawai (1991), uses boundary elements to find impulse responses of a concert hall. The development of new numerical techniques, and computational resources has made it a very promising field of research, specially for low frequency investigations.

The geometrical acoustics theory has been more popular for predicting the acoustical characteristics of an enclosed space, and the algorithms used have become more and more sophisticated with the computational developments. The basic principle of geometrical acoustics is that the sound is considered to propagate just like light rays, in geometrical optics. The effects of diffraction and refraction are not taken into account, though. Only the laws of reflection, absorption by the walls, and air absorption are considered (Kuttruff, 1979).

Two basic algorithms were developed over the years. One, known as the ray tracing algorithm, and other, known as the image source algorithm. The ray tracing algorithm "follows" a sound ray emitted by a source, verifying if, between two reflections, some receptor (usually a sphere) is hit (Vorländer, 1989; Kuttruff, 1993). When it occurs, the energy of the ray and the time of arrival are stored. After each reflection the absorption of the boundary is taken into account, and the new direction of propagation is determined. The diffusion phenomena can be modeled in this kind of algorithm, and the computation time is proportional to the reflection order (number of reflections suffered by a ray before it reaches the receptor).

The image source algorithm is based on the idea that a sound ray "can be thought of as originating from an image source which is the mirror image of the original sound source formed by the wall" (Vorländer, 1989). After constructing the image sources it is necessary to sum the contribution of each source, taking into account the distance of the source to the receiver point, and the wall and air absorption. This kind of algorithm leads to an impulse response with an impressive high time resolution, which is very important when one wishes to perform convolution with other sampled signals. There are some limitations, specially concerning to the computational time, because the process of determining sources relative to reflections of higher order (rays reflected more than once, before reaching the receptor) is driven by an exponential law. Also, some of the image sources may not be visible to the receptor, and a "visibility" test must be performed. Therefore, the impulse response to be calculated, usually, has to be truncated, as the computational time required increases exponentially with its length. The effects of diffuse reflections cannot be modeled in the image source algorithm.

To overcome some of the problems just mentioned, and to combine the advantages of both methods, some hybrid algorithms were developed (Kuttruff, 1993). The first hybrid algorithms proposed did not take diffusion effects into consideration, but a Round Robin test run in 1994-95, conducted by Vorlaender (1995), indicated that these effects might be of crucial importance for the accuracy of the results. There are a number of commercial softwares, based on the principles of geometrical acoustics, and many of them have included some way of modeling diffusion effects. Consequently, the importance of developing ways to measure diffusion (or scattering) coefficients has arisen, and some groups are already working on this subject, aiming to propose measuring techniques suitable for standardization, and also to found a data base for this parameter.

This paper concerns the accuracy of commercial measurement and simulation tools. For the simulation software, the sensibility of the results due to variations on input parameters (absorption and diffusion coefficients) were investigated. A small auditorium was modeled, using RAYNOISE commercial software, and the results for three room acoustical parameters were compared to those evaluated from the measured impulse responses of the room (using the MLS technique, through the commercial system MLSSA). The small room was simulated with and without diffusion effects.

2. MODELING THE HALL

The room modeled is the auditorium of the Mechanical Engineering Department, at the Federal University of Santa Catarina. The hall is shown in figure 1.

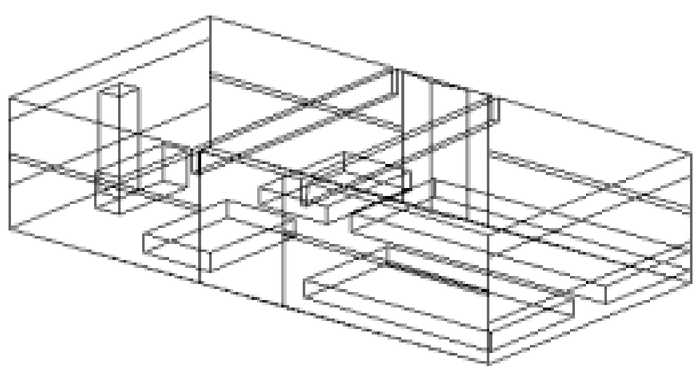

Figure 1: The Mechanical Engineering Department auditorium, at the Federal University of Santa Catarina.

Some details of the room were simplified. In the auditorium, the shape of the seats was the most meaningful simplification. Also, two metallic grids, in front of the main windows, inside the hall, were not represented, and the intention was to attribute absorption and diffusion coefficients to the correspondent element, in order to represent reality. The walls of the model are not exactly parallel.

When using RAYNOISE, one can choose the kind of algorithm to be performed between two hybrid methods (conical beam method and triangular beam method). The software allows calculations following up to 1000 reflections, for each ray, and the receiver can be represented as a mesh. For the presented calculations the triangular beam method was used.

3. SOFTWARE'S SENSIBILITY TO THE INPUT PARAMETERS

The input parameters are, basically, the absorption and diffusion coefficients, the environmental conditions (temperature and humidity), number of rays (N) and reflection order (R). Even though there are some uncertainties concerning absorption coefficients, for each material, they have to be chosen within an appropriate interval. The environmental conditions are not very difficult to be determined. Therefore, one have to deal with the right combination of N and R, since, depending on the average absorption coefficient of the room, and on its geometry as well, the final results may be very sensitive to the choice of these two parameters[7]. To minimize these effects, for each kind of room, the number of rays (N) chosen must be "high enough". The residual energy of the impulse response can be higher or lower, depending on the mean absorption coefficient of the room and, consequently, the number of reflections suffered by a ray. Therefore, the reflection order (R) also has to be "high enough".

To check the sensitivity of the results to the choice of N and R, some preliminary simulations were performed, first varying the number of rays, keeping R constant, and then varying the reflection order, keeping N constant. These results (1000 Hz, 1/1 octave band center frequency), for EDT, are shown in Figures 2 and 3.

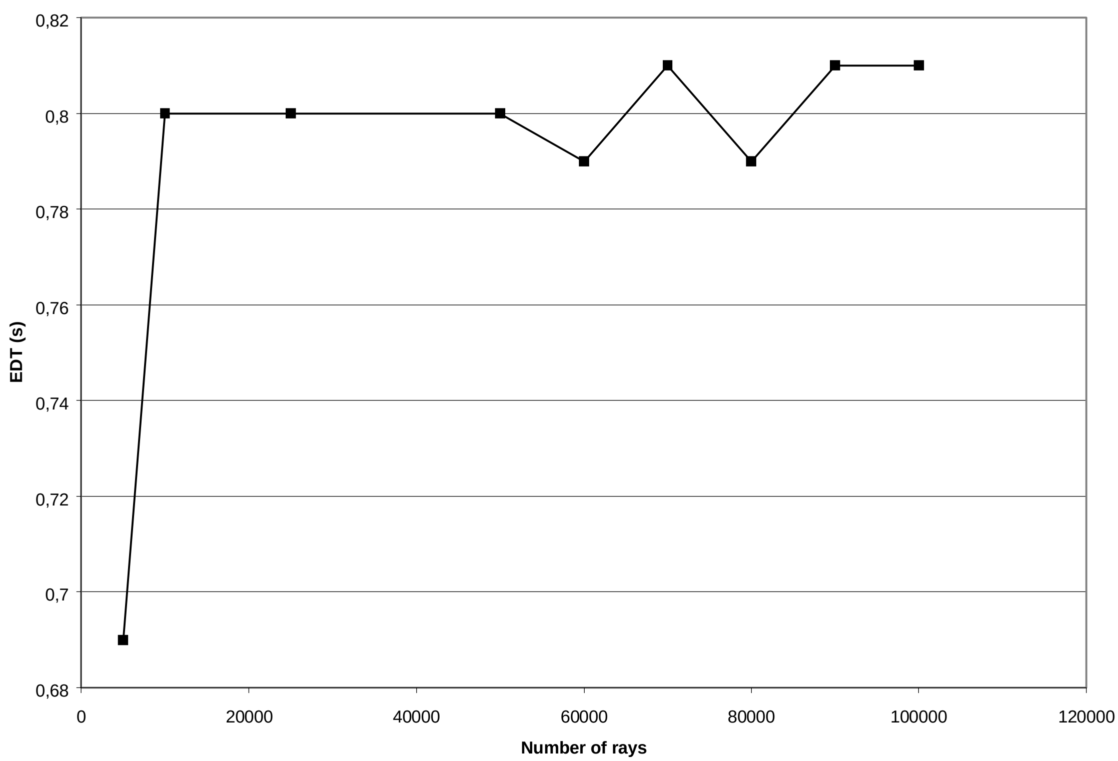

Figure 2: EDT x Number of rays (for reflection order 50, 1000 Hz);

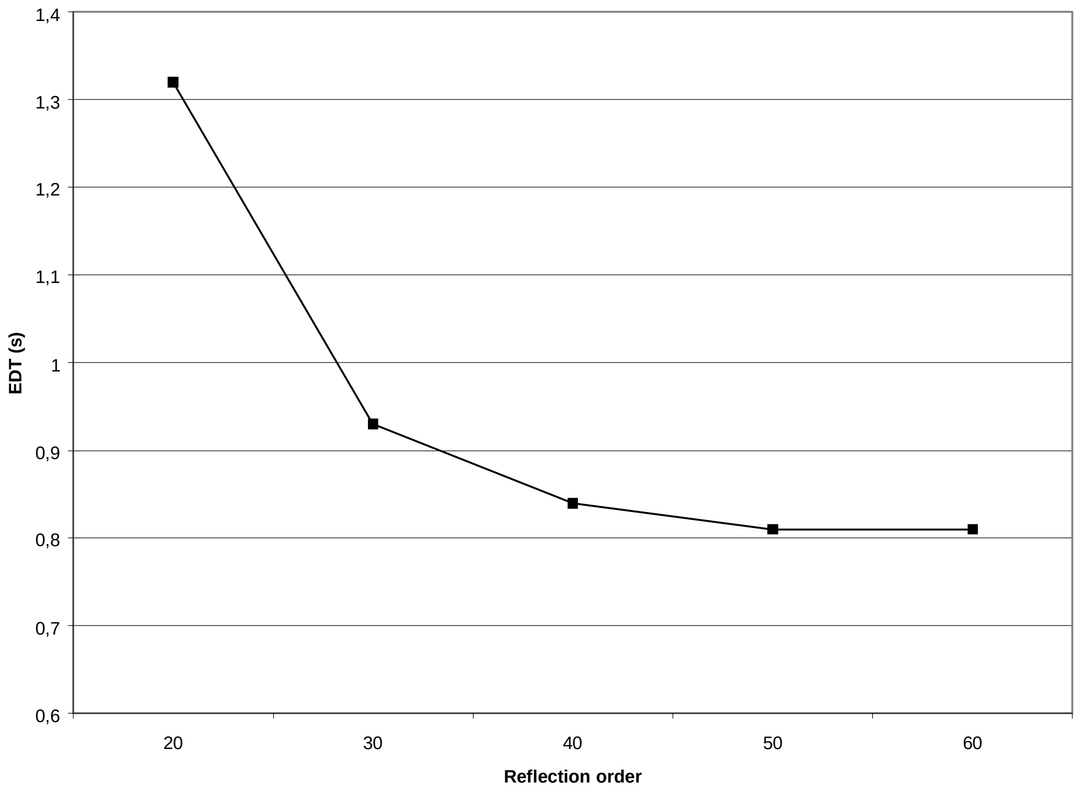

Figure 3: EDT x Reflection order (1000 Hz and 100000 rays).

From Figures 2, and 3 one can say that 50 reflections are "high" enough for EDT and Clarity to converge. Also, from Figure 2, it is possible to say that 25000 rays are enough for EDT to converge.

Nowadays, one of the greatest difficulties concerning the use of geometrical acoustics softwares is to attribute values to diffusion coefficients of the surfaces. And this seems to be a critical point, because the results can differ a lot, whether a simulation is run considering specular reflections or diffuse reflections (as shown later). For most of the cases it is always advisable to consider diffuse reflections for all surfaces of the model. But then the question arises: how to choose diffusion coefficients, since there is not a data base nor a standardized method for measuring them? Vorländer and the research group at Aachen University have been working on this subject, and some results regarding the problem are to be published (Vorländer, 1998).

For the model shown on Figure 1, all surfaces were considered to scatter the incident rays with a diffusion coefficient equal to 0.1 for all center frequencies of the octave bands, except the surfaces representing the seats and the front and rear windows. For these surfaces (seats and windows), several simulations were run, changing the coefficient of only one type of surface each time. Table 1 shows the results for EDT, 1000 Hz octave band center frequency, for shifts with steps of 0.1 in diffusion coefficient for the surfaces which represent the seats. Table 2 is similar to Table 1, for shifts in absorption coefficient.

Table 1: Results for EDT, shifting the diffusion coefficient of the seats in steps of 0.1 (1000 Hz).

Diffusion coefficient	0	0.1	0.2	0.3	0.4	0.5	0.6	0.7	0.8	0.9	1
EDT (s)	0.51	0.48	0.45	0.45	0.45	0.42	0.42	0.39	0.39	0.39	0.39

Table 2: Results for EDT, shifting the absorption coefficient of the seats in steps of 0.1 (1000 Hz).

Absorption coefficient	0	0.1	0.2	0.3	0.4	0.5	0.6	0.7	0.8	0.9	1
EDT (s)	0.54	0.51	0.48	0.48	0.45	0.42	0.42	0.39	0.39	0.36	0.36

From Table 1 one can notice that errors greater than 20 % in EDT can occur if the diffusion coefficients of the seats are not correctly attributed to the surfaces. Errors greater than 30 % can occur, in this case, if absorption coefficients are not chosen correctly. However, there is a large data base for absorption coefficients available in the literature, and the same is not true for diffusion coefficients, as remarked previously.
When using RAYNOISE, it is also necessary to pay attention to the input data that determine the time length of the histogram used to calculate room acoustics parameters, such as EDT, Clarity and Definition. These input data are “Histogram Interval” and “Histogram Length”. The former concerns the resolution of the histogram (5 ms, for example), while the latter concerns the number of discrete points along the time axis. It is recommended that the histogram time length should be about the average reverberation time of the room, long enough for the energy in the room decrease to a very low level. In the present case, the “Histogram Interval” was set to 5 ms, and the “Histogram Length” to 80, leading to a histogram time length of 0.4 s, slightly smaller than the measured reverberation time.

4. EXPERIMENTAL SET-UP

The MLS is generated by a commercial computer board plus the software for acquiring the data (MLSSA). Figure 4 shows a representation of the whole system.

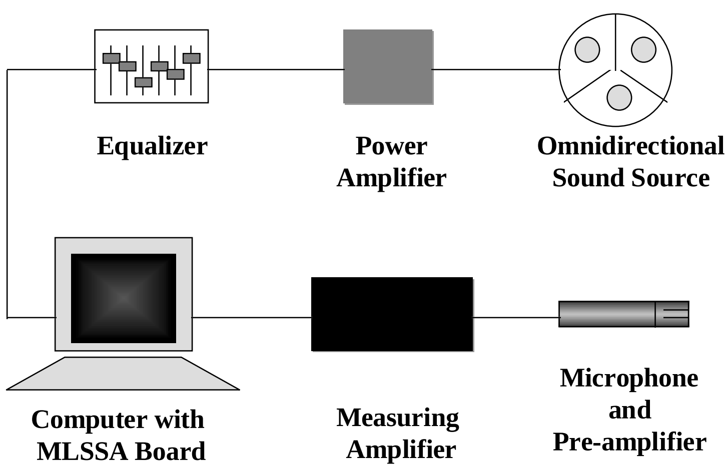

Figure 4: MLS measurement system.

The omnidirectional sound source used in the set was Brüel & Kjaer dodecahedron type 4241, and the microphone Brüel & Kjaer 4166 for random incidence.

As the frequency response of the loudspeaker is not uniform, usually, an equalizer is added to the system. For measurements of impulse responses, the source (loudspeaker) is recommended to be omnidirectional within a given tolerance (Lundeby et al., 1995). The directivity of the sound source was checked through measurements of reverberation time, turning the loudspeaker in steps of 20 degrees.

The use of the loudspeaker, and also the equalizer, can lead to another problem, depending on the signal level. The loudspeaker or the equalizer can be saturated, and, consequently, non-linearities are added, lowering the dynamic range of the results. Bradley (1996) has proposed a step-by-step procedure to optimize the decay range in MLS systems, depending on the background noise. In some cases it can be more advantageous to use lower levels for the driving signals, specially in quiet environments.

To avoid what is usually known as “time aliasing”, it is required to choose a sequence of a reliable length, in such a way that the test signal lasts approximately the reverberation time of the room (Borish and Angell, 1983).

5. RESULTS

The measurements were taken at the points indicated in figure 5. The results obtained with RAYNOISE compared with the measurements at a given position of the auditorium are shown in figure 6, 7, and 8, as a function of frequency. In figures 6, 7, and 8, the results of measurements and simulations at position 1 of the microphone, with and without diffusion are presented. The bars centered in the measured parameters (EDT, Definition and Clarity) indicate the acceptable deviation from measured results, in order to consider the simulation good, according to Vorländer (1995). Of course it is necessary to take into account the uncertainties related to the measurements, and at low frequencies they tend to be greater than at high frequencies. Even though, the results of most of the three parameters investigated seem to be reasonable for the frequency range analyzed, if diffuse reflections are considered.

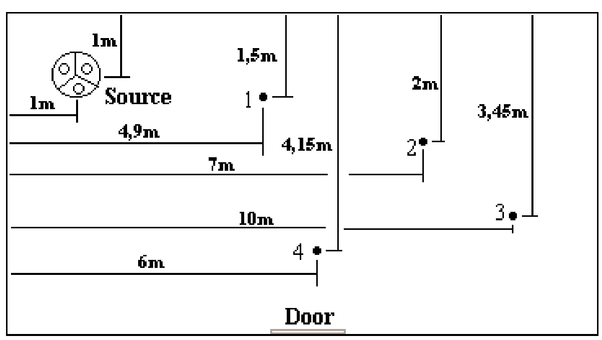

Figure 5: Representation of the positions of the sound source and the microphones where the measurements were taken.

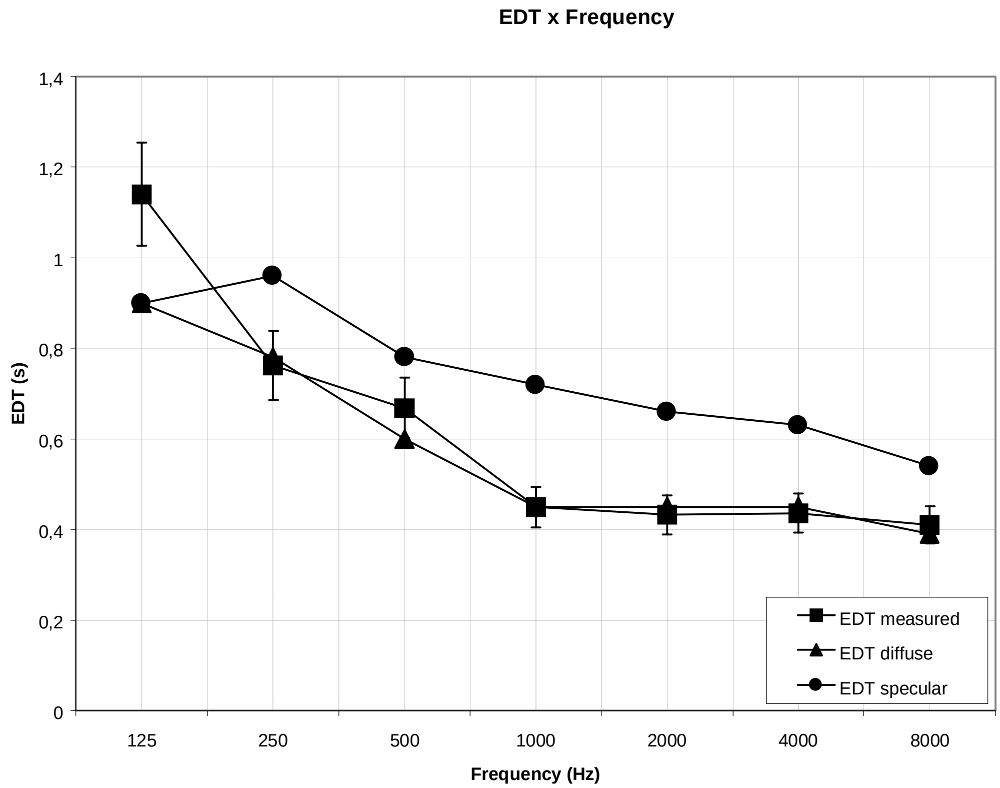

Figure 6: EDT calculated by RAYNOISE with and without diffusion, and measured at a given position of the room (position 1), as a function of frequency.

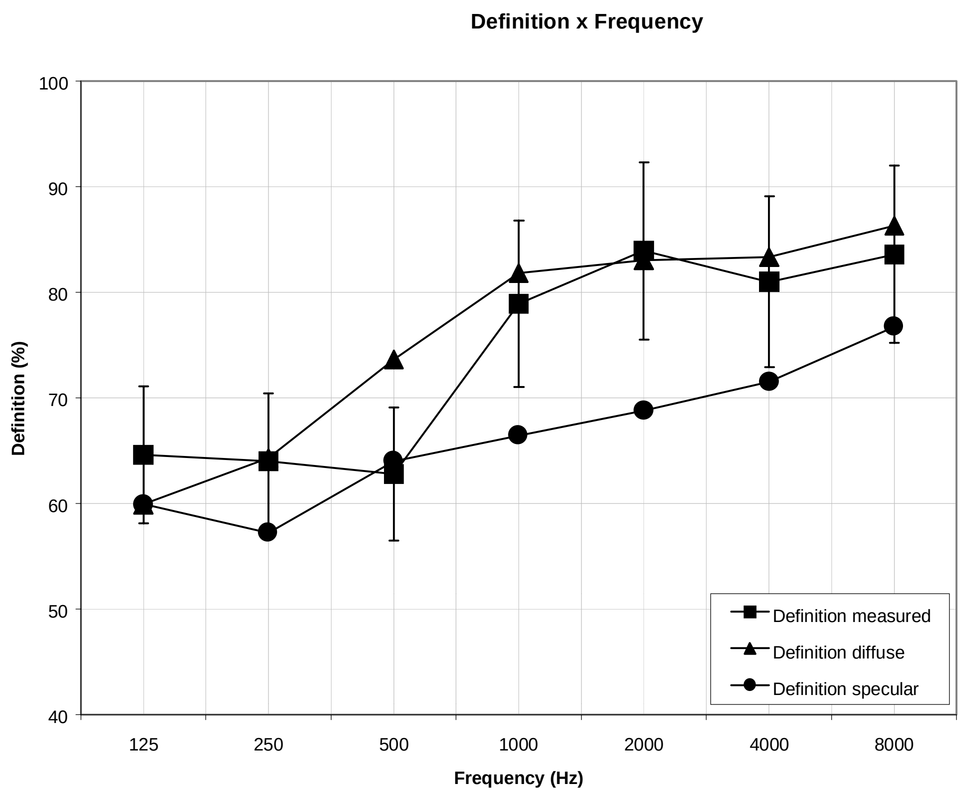

Figure 7: Definition calculated by RAYNOISE with and without diffusion, and measured at a given position of the room (position 1), as a function of frequency.

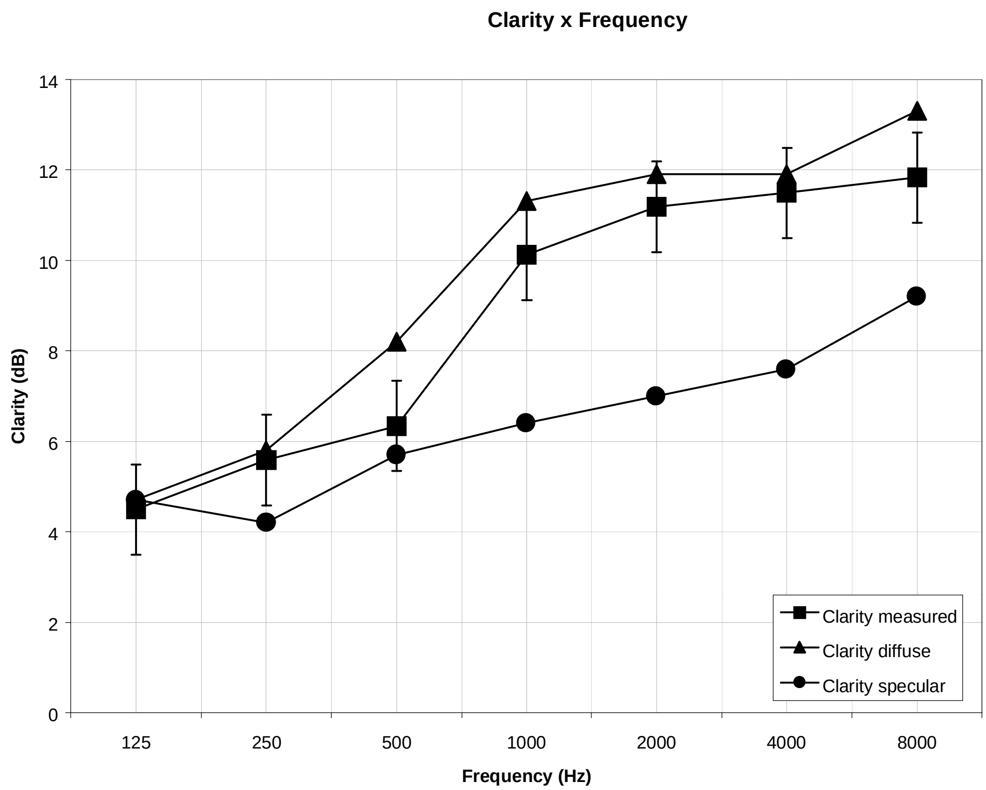

Figure 8: Clarity calculated by RAYNOISE with and without diffusion, and measured at a given position of the room (position 1), as a function of frequency.

If the effects of diffusion are not considered, the results tend to be very different from what is measured. The result for EDT, in 1000 Hz, is almost twice the measured value in that position if diffusion is not considered. For the specular case it would be impossible to bring the results for EDT closer to the measured values, within an acceptable interval, adjusting only absorption coefficients.

The results of the simulation presented here were achieved for diffusion coefficients set to 0, in 125 Hz, 0.02, in 250 Hz, 0.05 in 500 Hz, and 0.1 above 1000 Hz for all surfaces but the seats, where diffusion coefficients were set to 0.2 above 1000 Hz. The results for other positions in the room were also investigated and are similar to those presented here.

6. CONCLUSIONS

The calculated results, for EDT, Definition and Clarity, are very close to the measurements, when diffuse reflections are considered, and the differences tend to lie within the interval considered acceptable for this kind of algorithm (10% for EDT, 10% for Definition, and 1 dB for Clarity). There were no difficulties to attribute values to absorption coefficients, but attributing diffusion coefficients to the surfaces was almost a trial-and-error procedure. Some rules for choosing these coefficients, as proposed by Vorländer (1998), were partially followed, but doubts concerning their magnitude would still remain.

It is important to say that diffusion coefficients were attributed to all the surfaces of the model. In previous simulations this was not done and the simulated results for EDT did not match the measured values.

The differences verified between the results of the simulations, run when considering and not considering diffusion, call for the necessity of the development of techniques for measuring diffusion coefficients, and the foundation of a data base for this characteristic feature of the surfaces.

The results presented here are for a single source-receiver position in the room, even though the results for a different position were also investigated. The differences, for these other positions, are of the same order than those just described.

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