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Acoustical Parameters Used for the Subjective Assessment of Musical Performance Spaces

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Abstract

The problem of evaluating the subjective acoustical quality of a performance space through studying the value of measurable acoustical parameters is yet to be fully solved. As a contribution to its solution this paper aims to determine whether any relationships exist between the various acoustic parameters most often considered. The acoustical parameters that are topical for this study are in the number of three. The first of these parameters is the Inter-Aural Cross-Correlation Coefficient (IACC), which is a measure of the diffuseness of the sound field in the room. Then the Early Lateral Energy Fraction (ELEF), or a related measure, the Spaciousness (S), is used for estimating the relative amount of early sound energy reaching the listener from lateral incidence. And lastly the Initial Time Delay Gap (ITDG), which is a temporal factor and an indicator of the rapidity of early echoes for contributing to the early signal build-up at following the direct signal. For the evaluation of these parameters, the impulse response, IR, of the investigated enclosure is necessary. In room acoustical contexts the IR of a room is defined as the response that would be recorded by a microphone in response to a sharp and intense sound signal emitted within the enclosure. For the sake of calculation convenience a medium sized rectangular room is considered in this study and which represents a coarse approximation to a small performance hall. The theoretical model used for evaluating the IR makes a combination of the image sources method for wave reflections at the hard surfaces, and an exact model accounting for the diffraction of waves at the corners of the balconies. Furthermore, and in view of a more realistic determination of, especially the IACC, the phenomenon of sound scattering by the head of the listener is also taken into account. The variation of the value of the acoustic parameters with the position within the room at a constant height position for a presumed listener's ears is then presented in three-dimensional plots. Furthermore, psychoacoustical experiments have shown that the early part of the IR is decisive for the subjective assessment of performance spaces, and further, that the low frequency components of a signal have an important contribution to the feeling of spaciousness. Therefore, a low frequency filtering of the early part of the IR was accomplished for the needs of this study. As a completion to this study, an attempt has also been made toward enhancing the existence of some possible correlation between the various parameters investigated herein. As a result, simple relations may in fact be established between these parameters, and the usefulness of these relationships may find applications in subjective room acoustical assessments for instance for estimating one of these parameters, when inaccessible, from knowing the value of any of the two other ones.

1. Introduction

In the practice of several performance arts and in room acoustical science one is often in need of some objective predictive quantities that permit one to judge the quality of hearing conditions in an enclosure. Although the simulation of testing on down-scaled models, either physical or using numerical techniques, may give clues on how a performance hall would sound when achieved,

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acousticians, performers, and also sometimes architects agree on the need for well-defined objective descriptors that permit one to quantify specific subjective impressions on the acoustics of the performance space. There is a myriad of such parameters, which may be found listed in textbooks (Cremer et al., 1982; Ando, 1985; Kuttruff, 1991) or in review articles (Beranek, 1992). Depending on the purpose of use of the enclosure, different parameters may be of interest. As an example, a parameter that is of common prime interest in almost all room acoustical contexts is the Reverberation Time, RT, which is defined as the time measured in seconds made by the sound level to decrease by 60 dB from the time a sound source has been shut off in the room. The main purpose of using the Reverberation Time is for estimating the degree of sound absorption in a room. This parameter was for a long time considered as the most important indicator of the sound quality in a room, but as common experience shows that rooms having the same value of the RT may sound very differently, researchers considered the possibility of the existence of other physical descriptors that may explain the different subjective ratings made by people to the acoustics of a space, be it a concert hall, a theatre or a place of worship. This urge instigated a challenging research activity in several parts of the world, which started in the early fifties of the last century, and extended over nearly all the four decades that followed. As a result, several physical descriptors have been proposed, some of which have found useful applications due to their positive correlation with subjective impressions. One can name for instance the parameter Early Lateral Energy Fraction, ELEF, first proposed by Barron and Marshall (1981), and which evaluates the proportion of early lateral sound energy reaching a listener as compared to the total sound energy. In concert hall assessments this parameter has been found to be strongly correlated to the impression of spaciousness or as sometimes known also as apparent source width: the larger the value of this parameter is, the stronger is the impression of spaciousness in the enclosure. This gave partly some explanation to the praise won by the old narrow rectangular halls in which the sound signal reaching a member of the sitting audience is strongly reinforced by early reflections occurring on the side walls. A parameter having a close connection to the ELEF is the spaciousness, often referred to by the letter S. Another parameter, this time for estimating the degree of sound diffuseness at some position in a concert hall is the Inter-Aural Cross-Correlation, IACC, the definition of which was the subject of controversy between several authors for a long time. The IACC is evaluated through a simple cross-correlation of the impulse responses measured at the entrance of both ears of the listener. However, and for reasons of practicality the value of the IACC is often determined from the signals recorded by small microphones placed inside an artificial head and fixed at a position approximate to that of the eardrum. If the acoustical space of interest is instead intended for the purposes of speech presentations, then instead, the intelligibility of the message conveyed to the listener through the acoustical space is the matter of prime interest. For lecture rooms or theatres, the Modulation Transfer function, MTF, used for evaluating the Speech Transmission Index, STI, or its simplified version the Rapid STI, RASTI, may be appropriate (Houtgast & Steeneken, 1980; Steeneken & Houtgast, 1980; Schroeder, 1981).

2. The method of image sources

Sound waves as they propagate bounce against obstacles, and each time a wave is reflected part of the energy it is carrying is transmitted to the reflecting medium or lost upon reflection, whereas another part is reflected. When the reflecting obstacle is heavy, well polished and hard most of the incident energy is reflected, and very little of it is lost otherwise. In case the falling wave originates from a point source the wave reflected at an ideally hard surface has the same characteristics as if it were emitted by a point sound source situated at a position symmetrical to that of the original sound source through the reflecting surface. At such reflection there occurs then a doubling of the pressure resulting from the simultaneous contribution of two similar sound sources having the same strength and emitting at the same time. The original sound source is then said to be mirrored through the reflecting surface, and through successively imagining a sound source every time its corresponding wave reflects on a boundary, one can then come to building a lattice of image sources distributed in space around the origin sound source. The impulse response (IR) of a room is simply defined, as its name indicates, as the signal that would be recorded as a result of emitting a very short and intense signal within the room. Ideally, such a signal is defined in mathematics as a Dirac pulse, but signals in practice are limited in time and intensity. Hence, and with reference to the lattice of image sources, the impulse response of the room would be obtained from recording a succession of pulses emitted by the fictive image sources, these having emitted simultaneously a similar signal to that of the original source. This treatment of the problem of wave propagation is called geometrical, and is valid only at relatively high frequencies, when the mode density, the number of modes per frequency unit, is large. In the limit of low frequencies, when the typical size of a room dimension is of the order of or larger than a wavelength, and besides the diffraction effects which become more pronounced, the sound wave propagation is better described by the theoretical wave theory. Another algorithm, also popular in the simulation of wave propagation within closed spaces is the particle ray

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tracing method, which consists in following a ray as it propagates and considering its energy damping every time it bounces on a reflecting surface within the enclosure. These two methods have been subject to study and continuous improvement by acousticians for several decades, and the choice of either method is most often discussed in terms of both computing time and memory storage capacity (Lee and Lee, 1988). In the special case of rectangular rooms the image sources method is by far superior than the ray tracing method (Hammad, 1988; Lee & Lee, 1988; Stephenson, 1990). In the case of a hard room the lattice of fictive sources consists of first-order image sources (the reflections of the real source through the real walls of the room) and of higher order images (the images of the first-order images through the walls or the images of the walls).

3. Diffraction of sound around obstacles and screens

In modelling sound propagation within closed spaces one often encounters reflecting surfaces and obstacles with edges of various angles, such as corners and pillars. At low frequency sound waves have a more pronounced capacity to propagate around obstacles due to diffraction. This phenomenon is behind the common observation making it possible to hear a signal emitted by a bass loudspeaker from a lateral position, and not that emerging from a treble unit, which is then said to be more directive. Diffraction has been studied for several centuries now, and the theories that have been developed to describe accurately its physical causes are often complex and use sophisticated mathematics. At the turn of the last century Sommerfeld (1954) presented an exact theory for explaining the scattering of a plane wave by a half plane. This theory has inspired several of Sommerfeld's successors for generalizing the case of a half plane to that of a wedge of any angle and that of a plane incident wave to that of a spherical one. The theory, which uses very refined mathematical concepts, predicts that at sufficiently high frequencies (when the wavelength becomes much smaller than a typical distance in the problem) the total field at any point in space may be decomposed into two contributions, one wave satisfying the classical geometrical optics considerations (as discussed above for wave reflection), and another wave due to the existence of the edge of the obstacle. This has revived an old simplified approach to the problem to explaining the phenomenon of diffraction as attributed to a wave propagating from the edge of the screen with such characteristics as to ensure a smooth transition between the light and the dark parts of space, delimited by the obstacle. Theory would then be in full agreement with experimental observations. It had however to wait for another formulation of the problem, namely that considering a pulse wave instead of a continuous harmonic wave to make it possible to extract the edge wave from the total field. This was due to groundbreaking work of Biot and Tolstoy (1957), which has even been further simplified by Medwin (1981). The theory thus predicts that when an ideal spherical pulse propagates towards a hard wedge, an observer would sense the direct pulse if the pulse source is seen by him, with one or two pulses resulting from the reflection on one or both sides of the wedge, depending on the shape of the wedge and the position of the observer, but there always exists a wave diffracted by the tip of the wedge. Careful measurements give indeed evidence to the existence of the wave diffracted by the edge of the scattering screen. Figure 1 shows the results of an experiment where a loudspeaker emits a pulse towards a hard flat screen and a microphone to collect the response. The signal recorded by the microphone shows clearly the presence of the direct pulse signal emitted by the loudspeaker, which takes the shortest time to reach the microphone, and shortly afterwards the signal scattered by the edge of the flat screen. The experimentally determined signal agrees well with the theoretical prediction, both in the time domain and in the frequency domain (Ouis, 2002).

An important motivation for the use of transient waves in acoustical studies is the importance that the impulse response has for extracting useful information about the system under investigation. At the early stages of the experimental investigation, the use of scale models and short excitation pulses are often preferable due to the small size of available anechoic spaces, and also because usually mathematical models are developed for the case of ideally infinite geometries, a fact that is surely not always true in reality. Another advantage of using impulses in room acoustical studies is that once the impulse response is determined, the response of an enclosure to any signal may be processed. Often the signal is a short musical sequence, and a convolution, the name given to the signal processing operation, is acquired through integrating the tested signal, previously recorded under dry (anechoic) conditions, onto the impulse response of the room to give it the spatial dimension of the enclosure.

4. Impulse response of a rectangular enclosure with simple reflecting elements on two side walls

The impulse response of a room for a pair of well-defined positions of the sound source and the observer is defined as the signal recorded by a microphone at the observer position when a single intense and very short signal is emitted at the sound source position. On a time scale axis, the impulse response would then be represented by a series of impulses starting by the direct one, and then

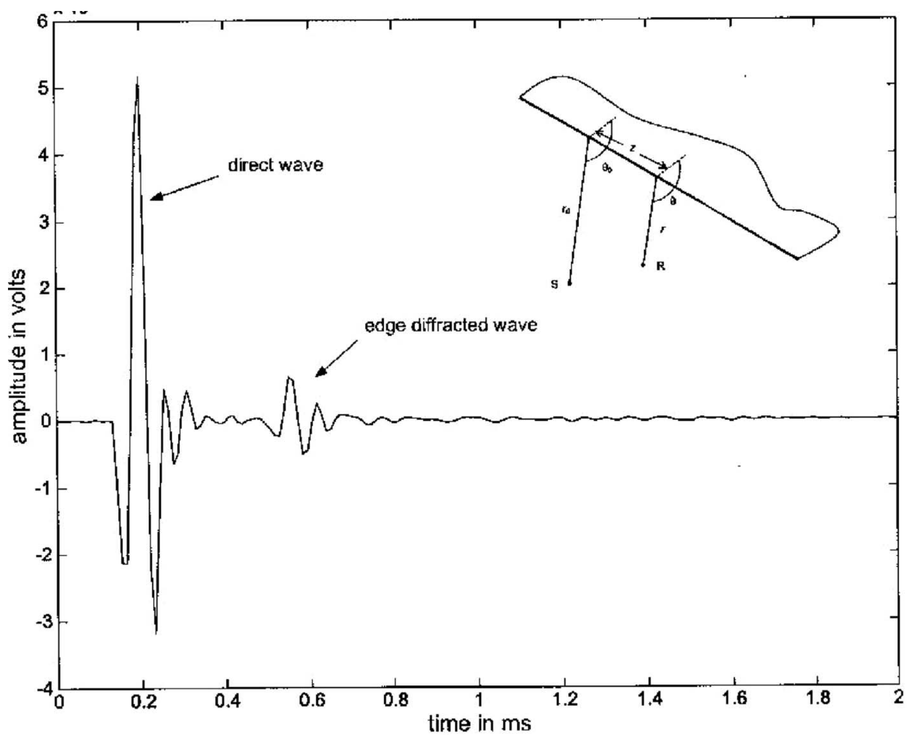

Fig. 1. Experiment showing the physical existence of the wave scattered at the edge of a screen. S and R are respectively the point source and the receiver. $r_0 = 8.9$ cm, $\theta_0 = 92°$, $r = 6.9$ cm, $\theta = 91.6°$, $z = 2.6$ cm.

followed by its successive reflections on the various reflecting surfaces within the room. The impulse response in room acoustical studies is a powerful means for describing the transmission path between the source position and the receiver position (Barron, 1984). For a room with hard walls the Impulse Response may in the simplest way be evaluated by means of the image source method. The geometry of the room intended for the present study is depicted in Figure 2 where all the room boundaries are taken as perfectly hard save the floor which is taken as perfectly absorbing, i.e. no sound reflections take place on it.

When the rectangular room includes extra reflecting elements, it is then no longer enough to rely only on the image sources method, and a visibility test is necessary for the accountability of all the image sources. This is due to the fact that the path from an image source to the receiver may be obstructed by a reflecting surface, be it real or fictive. There are some of such visibility tests that are quite efficient (Applied Acoustics, 1993), but in our case the simplest visibility test would be to draw a straight line between the image source and the receiver, and then to consider the possibility of intersection of this line with an image surface. A typical impulse response, for a sharp pulse, limited to only its early part in this case, is presented in Figure 3.

As mentioned earlier, the usefulness of impulse response calculation in room acoustical studies is to be able to predict the response of the room to a sound signal just from calculations. The effect of convolving a short dry saxophone piece of play on the IR of the room above is presented in Figure 4, where the progressive build-up of the signal at each reflection on the room walls is clearly seen on the step-like time signal as time progresses. This signal sequence was processed from a wav.-file.

For a more detailed picture of what happens when the diffraction effects are taken into account, a triangular pulse was convolved to the IR of the room, including the effects of diffraction of all pulses from the image sources at the edges of both balconies. Figure 5 shows these effects both in the time domain and in the frequency domain, i.e. the transfer function obtained from Fourier transforming the impulse response.

The continuous line on the right-hand side of Figure 5 is the frequency response of the room when both single and double edge diffraction effects are taken into account, whereas the dotted line accounts only for single diffraction effects. The result of taking higher orders of diffraction is seen to have some effect, especially at the lower part of the frequency spectrum. Figure 6 shows some detail on the effect of the edge diffraction phenomenon on the IR.

It is worth mentioning in connection to this last figure that depending on the signal characteristics the edge diffraction effects can be clearly audible when the

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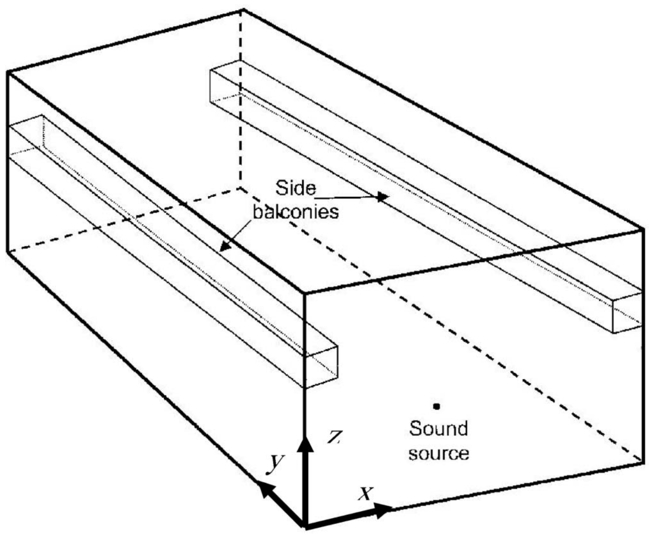

Fig. 2. Sketch of the rectangular room with the hard diffracting elements on the side walls.

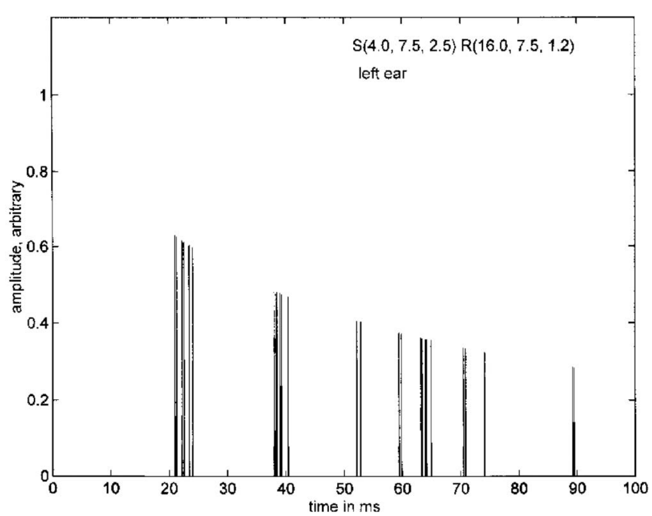

Fig. 3. Early portion of the IR of the room in Figure 2 as predicted by the image sources method. S and R are respectively the point source and the receiver with coordinates (x, y, z) in the coordinate system illustrated in Figure 2.

computed room impulse response is convolved with the anechoic signal. The simulation procedure, also called in the audio jargon 'auralization', has been the subject of intensive research with applications to concert hall acoustics, and several interesting findings may be found in some of the most recent publications (Torres et al., 2000).

5. The inter-aural cross-correlation, IACC

The history behind this parameter begins by the late sixties when Keet in 1968 found that the IACC is related

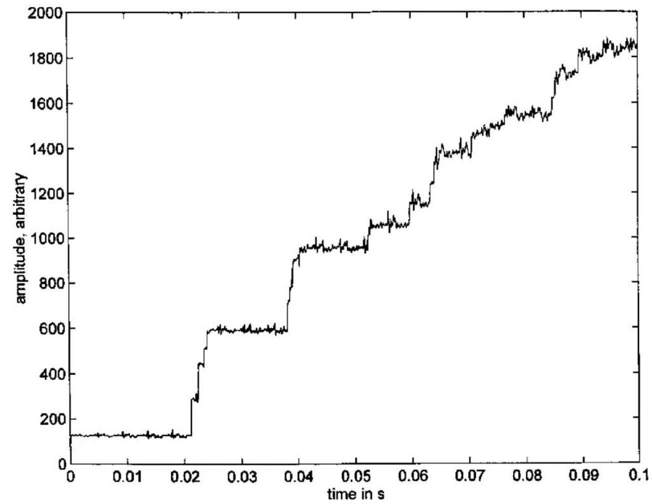

Fig. 4. Convolution of the early part of the room's IR with a short saxophone instrumental.

to the apparent source width. A little later, in 1974, Schroeder and his co-workers established further that the IACC is one of three important factors in listening impressions (Shroeder et al., 1974). As a refinement to the IACC, Hidaka conjectured that two separate values of IACC are to be distinguished. One value is attributed to the early sound energy, determined over the first 80 ms of the impulse response, and which is better correlated to spatial impression (SI), and the other one for the late sound energy calculated in the range between 80 ms and 3 s, and which gives a measure of the feeling of envelopment (Cremer et al., 1982).

Regarding the definition of the IACC, different forms for its expression have been suggested by different authors. Cremer, in 1976 found it reasonable to call the absolute value of the coefficient,

$$\kappa(t) = \frac{\int_0^\infty p_l(t) p_r(t+t)\mathrm{d}t}{\left[\int_0^\infty p_l^2(t)\mathrm{d}t \int_0^\infty p_r^2(t)\mathrm{d}t\right]^{1/2}}, \tag{1}$$

the "Inter-Aural Cross-correlation Coefficient", p_l and p_r being respectively the pressures measured at the left, and the right ears of a dummy head (Cremer et al., 1982). Damaske recommended the maximum of $|\kappa(\tau)|$ as a room acoustical criterion, and Keet proposed a limited time value t_g instead of infinity, for the integration in Eqn (1), giving hence a sort of "Short Time Correlation Coefficient" after evaluation at $\tau = 0$. Gottlob on the other hand preferred to use the maximum of $|\kappa(\tau)|$ for $t_g = 50$ ms with the further restriction that $\tau_{\max} = 1$ ms. This last definition is the one Kuttruff (1991) opted for with the slight change that t_g be taken to be 80 ms instead of 50 ms. Finally, Ando (1985) chooses the concept of long time IACC defined in a similar way as in

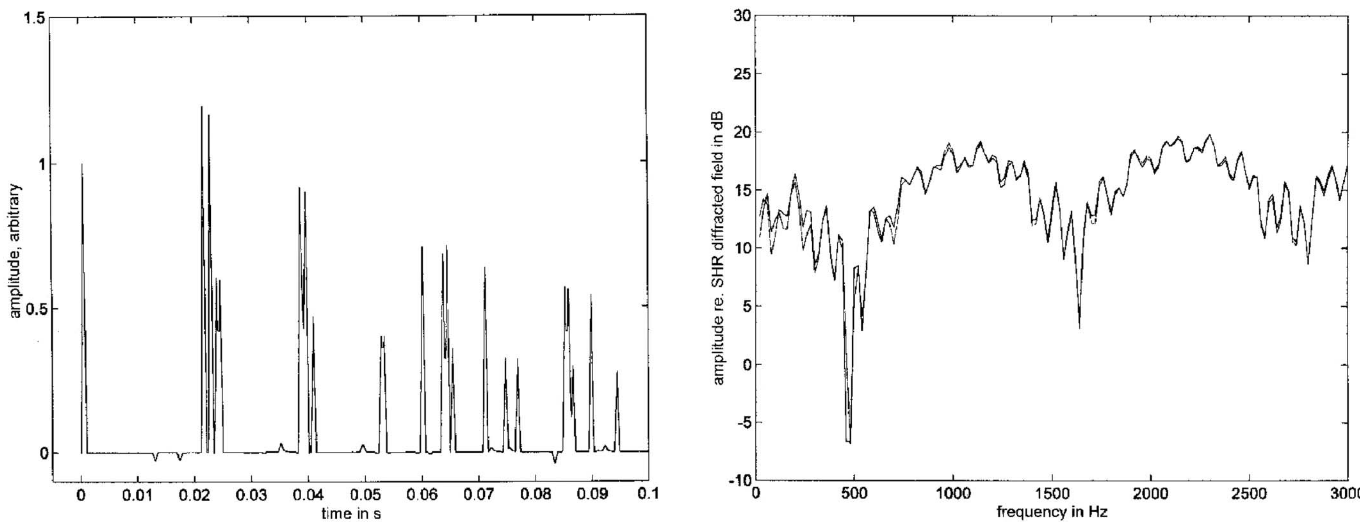

Fig. 5. Early portion of the IR for a triangular pulse. Left: time domain, and right: frequency domain. – – – : single diffraction, —— : single + double diffraction.

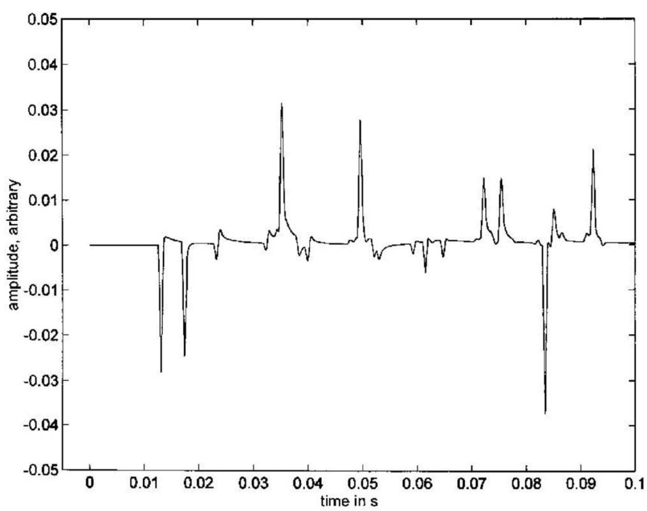

Fig. 6. Extraction of the edge diffraction effect from the IR in Figure 5, left.

Eqn (1) with the exception that the integrations be performed in the interval $[-35\ \text{s}, 35\ \text{s}]$. A study in more in-depth of this measure with several references may be found in Beranek's (1992) updated review paper on concert hall acoustics.

Hence, with the cross-correlation between the impulse responses at both ears defined as

$$\kappa(t) = \frac{\int_0^\infty p_l(t) p_r(t+t)\mathrm{d}t}{\left[\int_0^\infty p_l^2(t)\mathrm{d}t \int_0^\infty p_r^2(t)\mathrm{d}t\right]^{1/2}} \tag{2}$$

we consider the IACC as the maximum of $\kappa(\tau)$ within $|\tau| < 1$ ms (Applied Acoustics, 1993).

The lower the value of the IACC, the lesser is the coherence between the ear signals. The IACC is therefore an indication of the diffuseness of the early sound field and can in practice be interpreted as a sensation of a more apparent extension of the sound source.

It is however important here to take into account the diffraction effects around the head, and therefore for a more realistic evaluation of the IACC the Head Related Transfer Function, HRTF, of a dummy head were also considered in evaluating the impulse response. Binaural hearing combined with diffraction around the head gives a listener in the room an impression that cannot be obtained when using monaural hearing. When hearing with both ears, localisation of the sound source is said to be made possible through two phenomena occurring around the head, and operating at different frequencies. One mechanism uses the phase difference between the ear signals, and is active at low frequencies, whereas at rather higher frequencies the brain uses the sound pressure level differences as a clue for localising the position of the sound source (Cremer et al., 1982). Gardner and Martin measured the HRTFs at the entrance of each ear canal of a dummy head in an anechoic chamber for different positions of the sound source. These HRTFs were taken for loudspeaker positions on a hemisphere centred at the dummy head at angular positions separated by 10° along the azimuthal and the elevation directions. A file containing all these data is made available by their authors on the Internet and may be downloaded easily by interested researchers (Kemar, 1994). Hence, after positioning each of the virtual sound sources on the image source lattice, its signal was convolved with the HRTF corresponding to the nearest direction to the image source. A typical impulse response calculated at the left ear of a dummy head positioned in the middle of the room is shown in Figure 7.

Figure 8 represents a surface plot on the variation of the IACC with the position in the room, as evaluated

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according to Eqn (2), but without taking into account the effect of diffraction around the head.

Figure 8 shows a plot of the IACC for an unfiltered impulse response, but the effect of frequency content is not to be neglected when considering this parameter. Indeed, earlier research has shown the role that may be played by the low frequency components of a test signal on the judgement of spatial impression in a room (Ando, 1985). A new plot of the IACC, now with a low pass filtered impulse response, is shown in Figure 9.

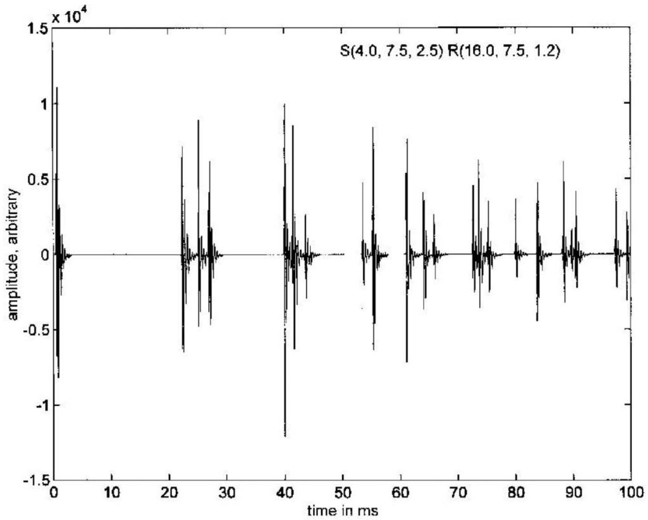

Fig. 7. Impulse response of the room in Figure 2, including the effect of diffraction around the head through convolution of a dummy head's HRTF at the left ear.

The cut-off frequency of the filter was chosen at 800 Hz, and for which a wavelength of 42 cm corresponds to about twice the average distance between the ears on the human head. This frequency is at about where the ear has its highest sensitivity and is also critical for the perception by the human brain of the incidence direction for narrow band signals. The filtering of the impulse response was accomplished through a convolution of the unfiltered impulse response with the time domain form (inverse Fourier transform) of the low pass filter in the frequency domain.

The effects of both masking by the head and filtering are clearly seen to have an important effect especially at positions localized laterally to that of the sound source. This effect is particularly enhanced at even lower frequencies, as revealed by the plot on Figure 10 where the cut-off frequency of the low pass filter is instead set to 300 Hz.

6. The early lateral energy fraction, ELEF, and spaciousness, S

Much information can be extracted from the impulse response of a room, but this is valid only as far as the directional characteristics of the sound field in a room are of minor importance. However, the perception of

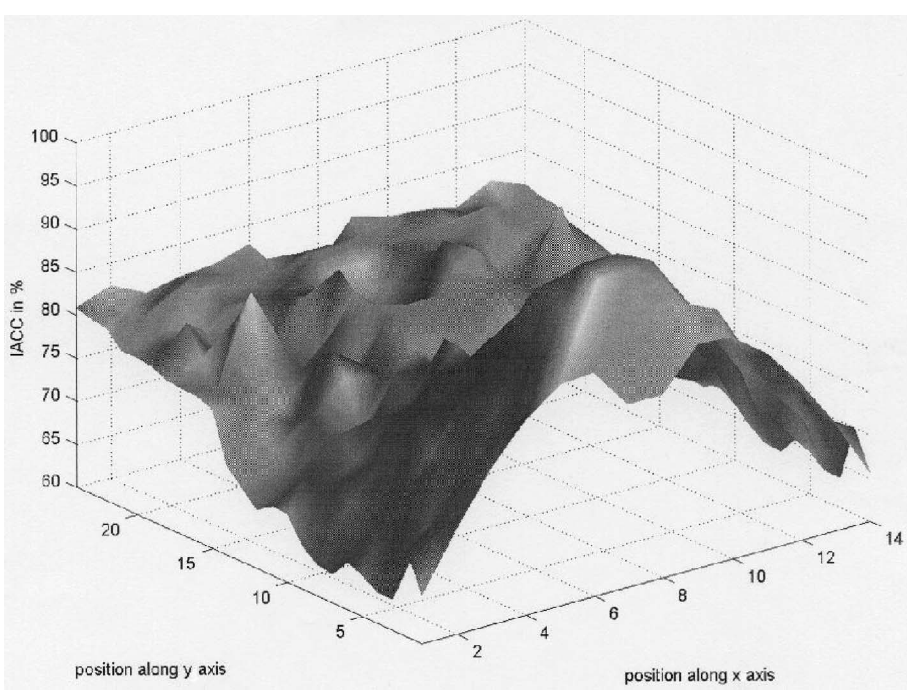

Fig. 8. Variation of the IACC within the room as a function of the position, with diffraction around the head. Triangular pulse with a base $\Delta t = 100$ μs, unfiltered response.

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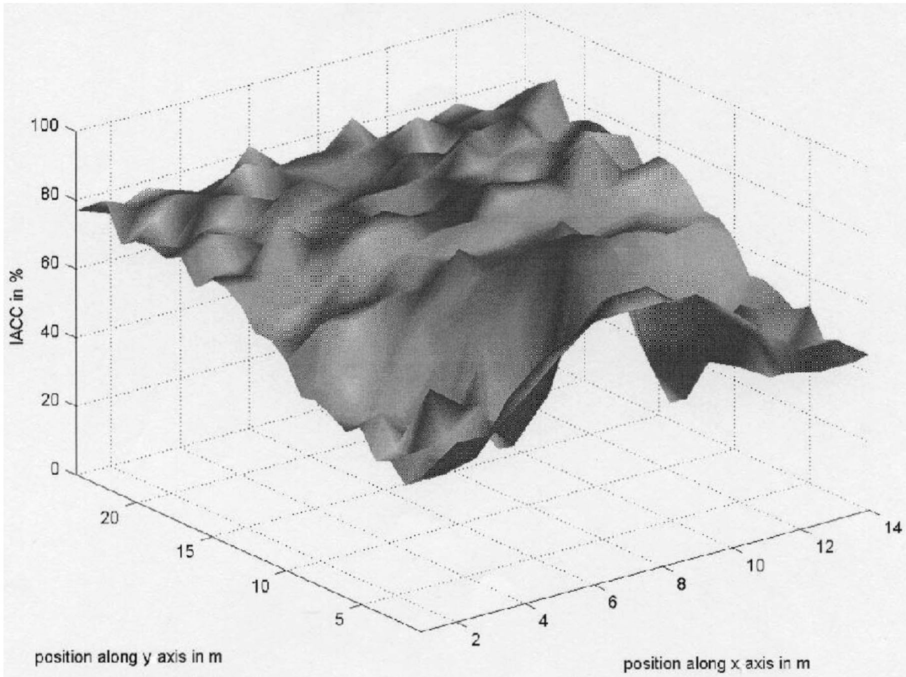

Fig. 9. Same as Figure 8 but with adding the effect of a low pass filtering at 800 Hz.

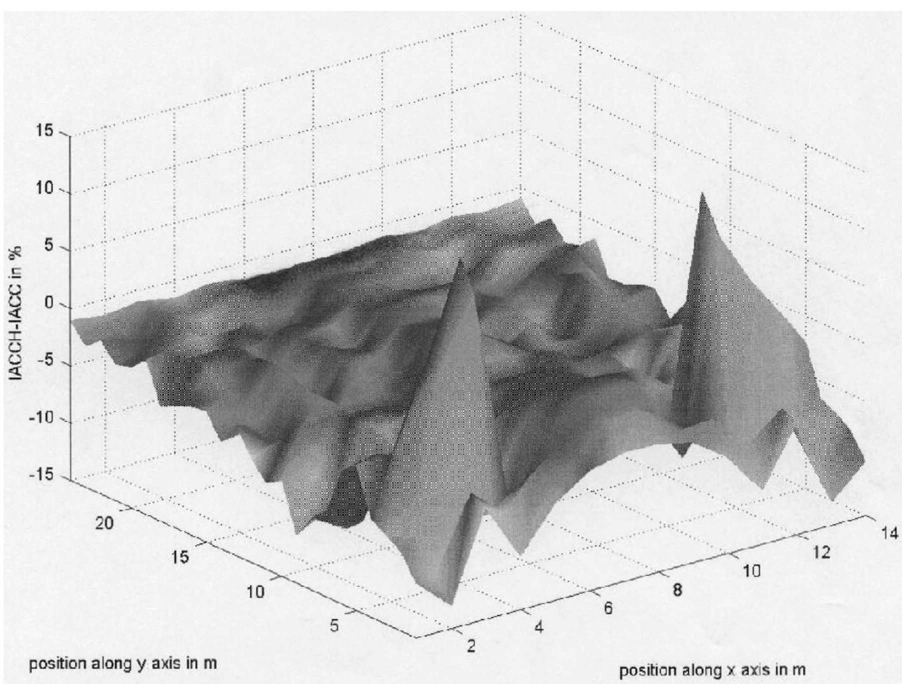

Fig. 10. Effect of diffraction around the head and low frequency filtering on the IACC. IACCH and IACC are respectively the IACC with, and without account of KEMAR's HRTF.

auditory information has also a directional dimension, which is made possible to the brain through the simultaneous processing of the signals arriving at both ears. Moreover, when the signals reaching the ear are large in number, their analysis is made in a global manner rather than analysing each signal separately. Researchers in room acoustics were therefore confronted with the challenge of finding a rating that emphasizes the

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effect of sound reflections incoming from different directions on the sensation of spatial impression. The different authors who made a significant progress of research in this subject used different denominations for this subjective feeling such as "spatial responsiveness", "ambiance", "apparent source width", "räumlichkeit", "spaciousness" or "subjective diffuseness". In the beginning, and based on the reputation of old concert halls presenting the characteristic of high-field diffuseness resulting from the presence of small decorative elements on the walls, spatial impression was believed to be a consequence of the uniformity of sound directional distribution. Later on, and with the introduction of tools for generating synthetic sound fields, it showed that spatial impression may be brought about with just a few reflections provided the following conditions are satisfied (Kuttruff, 1991):

- mutual incoherence of the reflected sound fields;
- their intensities are above some minimum level relative to the direct sound;
- their arrival times must be no later than 100 ms after the direct sound;
- they must be lateral.

These general observations were attained when Barron and Marshall's work made a decisive turn in this research field. It was especially due to Barron with his extensive work on listening conditions that the decisive conclusions were made on the strong correlation between spatial impression and early reflections. More specifically, spatial impression, or also called objective envelopment (Barron, 1993), is directly related to the ratio of lateral energy to the total energy during the first 80 ms of the impulse response. Formally, the ELEF is given by

$$\mathrm{ELEF} = \frac{\int_{5\ \mathrm{ms}}^{80\ \mathrm{ms}} p_{\mathrm{L}}^2(t)\mathrm{d}t}{\int_0^{80\ \mathrm{ms}} p^2(t)\mathrm{d}t}, \tag{3}$$

where $p_{\mathrm{L}}^2(t) = p^2(t)|\cos\theta|$, θ being the angle made by the incident pulse and the line normal to the median of the listener's head, that is the line passing through both ears of the observer when looking towards the stage, and p is the pressure. Subsequent research on the ELEF was merely devoted to the confirming of this pioneering work (Cremer et al., 1982; Beranek, 1992).

Calculations made on the variation of the ELEF in the room are shown in the plot of Figure 11, where it is to be noted that only the plain impulse response is required for the calculations without the need of incorporating the effects of diffraction around the head.

Another parameter also used sometimes for describing the feeling of envelopment is Spaciousness, denoted by S, and which is defined by

$$S = \frac{\int_{5\ \mathrm{ms}}^{80\mathrm{ms}} p_{\mathrm{L}}^2(t)\mathrm{d}t}{\int_0^{80\ \mathrm{ms}} \left(p^2(t) - p_{\mathrm{L}}^2(t)\right)\mathrm{d}t} \tag{4}$$

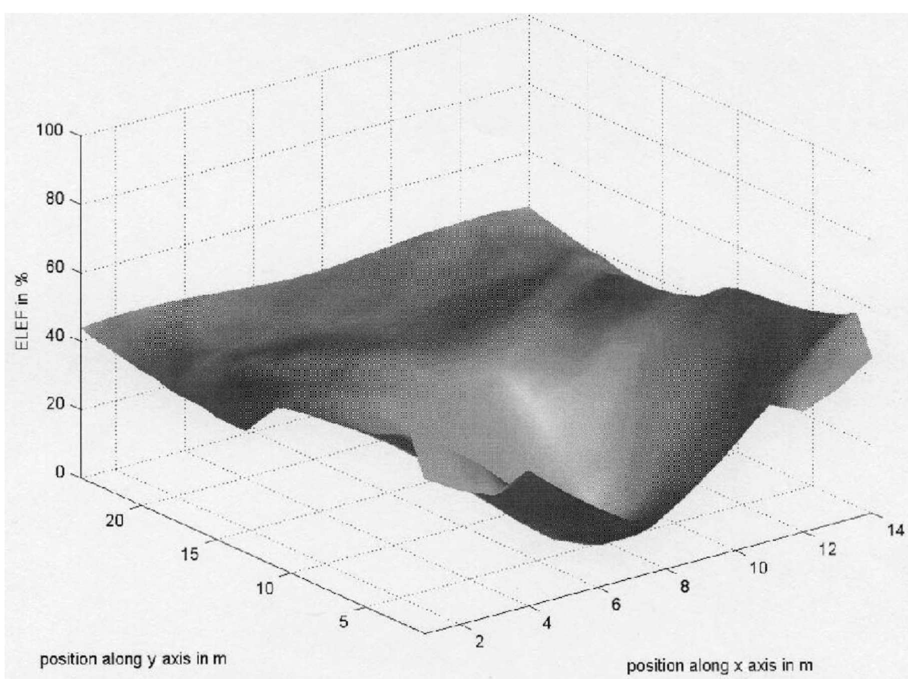

Fig. 11. The ELEF as calculated at different positions in the room with side balconies. Low pass filtering at 800 Hz.

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or also sometimes approximated to the ELEF through the relation:

$$S \approx \frac{\text{ELEF}}{1 - \text{ELEF}/100}. \qquad (5)$$

The plot corresponding to this parameter in the room of our study case is presented in Figure 12.

According to Eqn (5), and as shown also on this last figure, the values taken by S may be larger than 100%, meaning that in total early energy, the lateral energy stands for a larger proportion than the non-lateral one. Note in both Figures 11 and 12, and as expected, that the lowest value of the lateral energy is for the listener's positions nearest to the sound source, and which also correspond to the lowest possible values of spaciousness. At these positions, the sound field is predominantly composed of the near-field component, and the time of arrival of the lateral reflections following the direct field is at its greatest value, diminishing therefore the amount of lateral energy reaching the listener. An even lower cut-off frequency of the low pass filter has the major effect of narrowing the area of the favourable lateral energy around the sound source, thus spreading to more positions the pleasant feeling of spaciousness.

7. The initial time delay gap, ITDG

This parameter is defined as the time counted in ms between the arrival of the direct sound and that of the first reflection reaching the ear of the listener. The ITDG was introduced by Beranek (1962) after his extensive survey over different concert halls world wide. Since Haas' (1951) experiments on simulated sounds on the effect of early reflections, it was established that the localization of the source is decided by the first wave arriving at the listener and that delayed reflections contribute mostly to enhancing speech intelligibility. In auditorium acoustics, the ITDG was merely intended to be a measure of perceived acoustic intimacy, a subjective property of concert halls strongly related to proximity to performers. The ITDG has however been the subject of several criticisms (Barron, 1993), but perhaps the most obvious one is that the earliest reflection reaching the subject after the direct signal is not obligatorily a lateral one, and therefore may be one resulting from reflection at the ceiling or any area of sufficient size near the stage. This study tries however to shed light on any possible correlation of this parameter to the IACC or the ELEF, and a mapping of the values of the ITDG as calculated in seconds in the study room is shown on the plot of Figure 13.

As expected, the largest values of the ITDG are around the sound source where the first reflections, in this case from the lateral walls, have the longest delay following the direct signal from the sound source.

8. Relation between IACC and spaciousness, or ELEF

For each of the positions of interest in the room the IACC and Spaciousness S, or alternatively the ELEF,

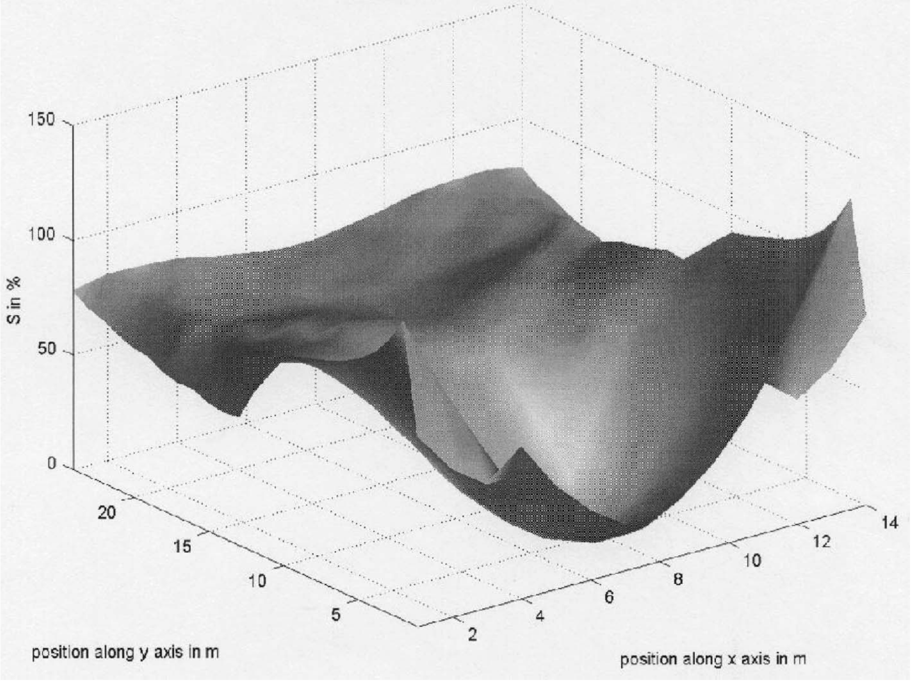

Fig. 12. Spaciousness S for the room with side balconies. Low pass filtering at 800 Hz.

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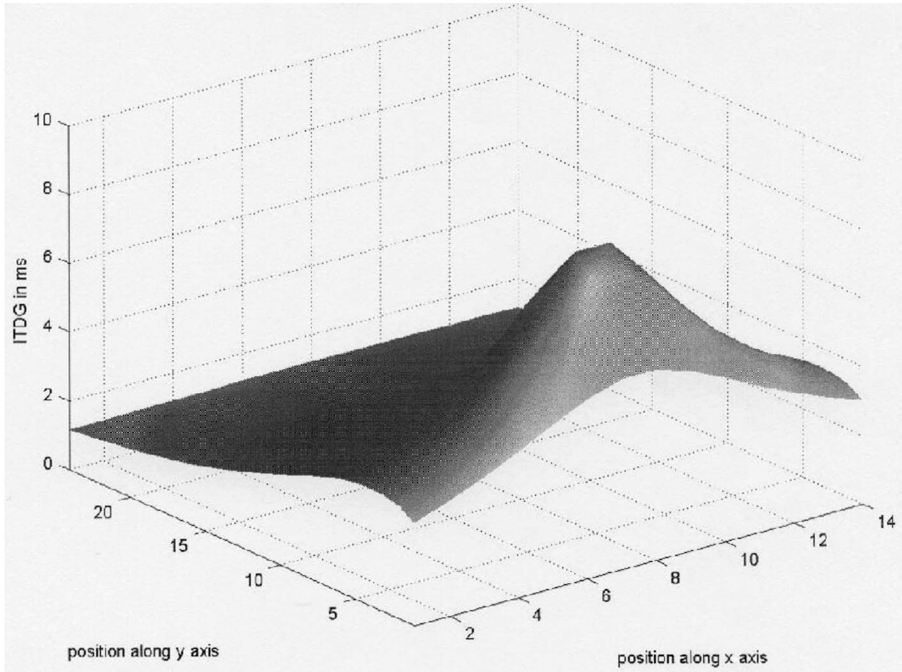

Fig. 13. Plot of the value of Initial Time Delay Gap, in milliseconds, as a function of position in the room.

were evaluated in percent and on a coordinate system where each point has for coordinates the corresponding values of the parameter of interest. The plot for the ensemble of all listener positions is summarized in Figure 14.

The two curves drawn on the plot are on the one hand the curve for the hypothetical relationship (Kuttruff, 1991):

$$\mathrm{IACC} = \frac{1}{1 + S/100} \times 100\% \qquad (6)$$

and on the other hand the one resulting from executing a curve fitting for the calculated points, the equation of which is given by

$$\mathrm{IACC} = \frac{1}{1 + S/200} \times 100\%. \qquad (7)$$

This latter equation is also valid when the calculation positions are chosen at some distance from the side walls (2 m from nearest side wall, encircled dots on the plot), but shows even better agreement for positions at the immediate neighbourhood of the sound source, positions where the variations of IACC and S are the most pronounced, although not in the immediate vicinity of the sound source, that is for points with IACC less than about 85% as shown in Figure 15.

For positions far away from the sound source, in the rear half of the room, the dots of IACC versus

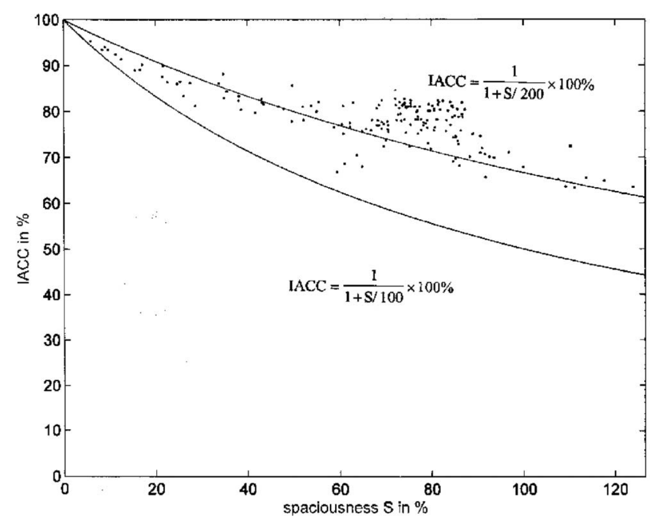

Fig. 14. IACC versus spaciousness in the room, all room positions included.

S cluster in the region $75\% \leq IACC\% \leq 85\%$ and $60\% \leq S \leq 90\%$ without the possibility of establishing a clear relationship between the measures. For the relationship between IACC and ELEF, using the expression for S as given by Eqn (5) and putting this into Eqn (7) yields

$$IACC = \left(1 - \frac{ELEF}{200 - ELEF}\right) \times 100\%. \qquad (8)$$

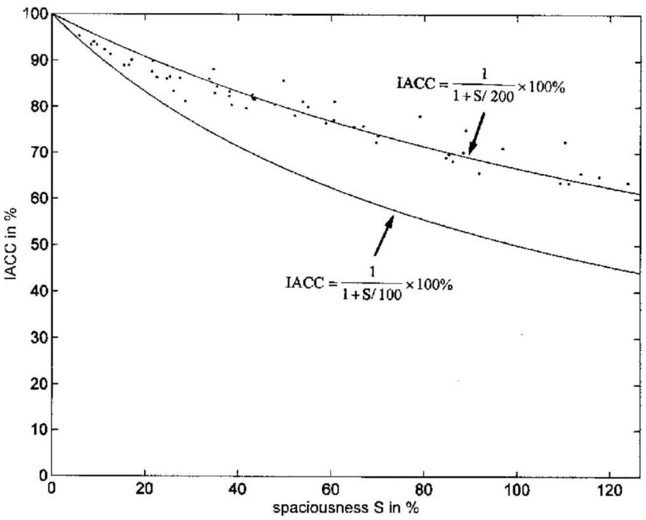

Fig. 15. As Figure 14 but for points around the sound source.

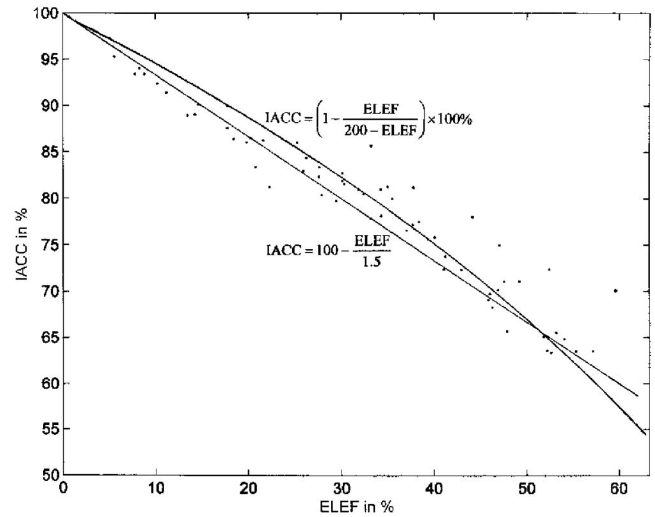

Fig. 16. Plot of IACC versus ELEF for positions around the sound source.

However, this expression is somehow cumbersome to handle, and instead, a curve fitting made on the calculation positions shows a more tractable equation given by

$$\mathrm{IACC} = 100 - \frac{\mathrm{ELEF}}{1.5}. \tag{9}$$

Furthermore, using Eqn (9) instead of Eqn (8) amounts to a maximum relative error of 5%, and this applies for values of ELEF up to 60%. These results are summarized in Figure 16.

9. Relation between IACC ITDG

A plot of IACC versus ITDG is presented in Figure 17 and the equation of the line is given by

$$\mathrm{IACC} = 10 \times \mathrm{ITDG} + 6 \tag{10}$$

with the IACC as earlier being given in percent and the ITDG in ms.

Using Eqn (9) would result in a relationship between the ELEF and the ITDG as given by

$$\mathrm{ELEF} \approx 140 - 15 \times \mathrm{ITDG}. \tag{11}$$

It is worth reminding that the ITDG, as opposed to the IACC and the ELEF, is frequency independent and consequently different equations than Eqns (10) and (11) are expected to relate it to the IACC or the ELEF for other frequency filterings of the impulse response.

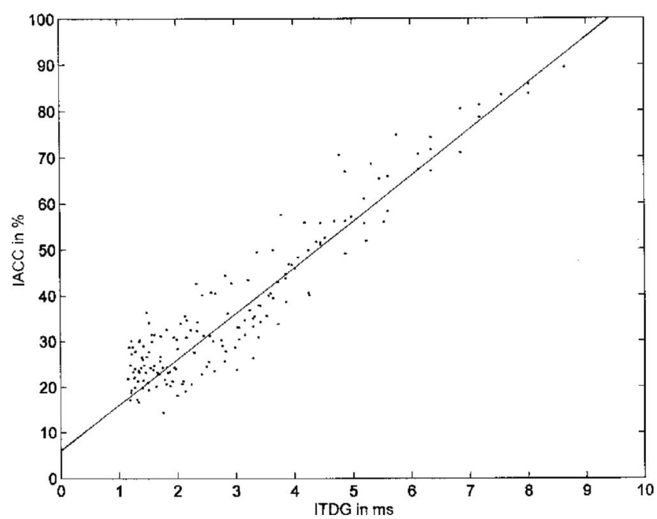

Fig. 17. Plot of IACC versus ITDG.

10. Conclusions

In this paper, a theoretical study was made on the relationship between some room acoustical descriptors. These descriptors are the IACC, the ELEF (alternatively the spaciousness) and the ITDG, often used as physical measures of subjective listening preferences in performance halls. The room used for the calculations was taken as rectangular with perfectly hard side walls and ceiling having two simple side reflecting elements representing balconies and with a perfectly absorbing floor. As the evaluation of these parameters necessitates the knowledge of the impulse response, this latter was calculated using the image sources method to which was included the edge diffraction phenomenon up to second order of multiple wave interaction between edges. Only

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the plain impulse response at the listener position is needed for calculating the ELEF and the ITDG, whereas the IACC requires knowledge of the impulse response at both ears of the listener. This was accomplished by incorporating in the calculation program the HRTFs as measured on a dummy head in an anechoic environment and for different positions of the sound source which where convolved to the pulse emanating from the corresponding image source. The important results drawn from this study are that it is possible, at least for the modelled room specified above, to establish simple relationships between these parameters, and these are given by Eqns (7), (8), (10) and (11). Similar studies in more representative acoustic spaces will allow determination of the degree to which these results can be generalized.

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References

Ando, Y. (1985). *Concert hall acoustics*. Berlin: Springer-Verlag.

Applied Acoustics (1993). Special issue on Computer modelling and auralisation of sound fields in rooms. *Applied Acoustics*, *38*(2–4).

Barron, M. (1984). Impulse testing techniques for auditoria. *Applied Acoustics*, *17*, 165–181.

Barron, M. (1993). *Auditorium acoustics and architectural design*. London: Chapman & Hall.

Barron, M. & Marshall, A.H. (1981). Spatial impression due to early lateral reflections in concert halls: derivation of a physical measure. *Journal of Sound & Vibration*, *77*, 211–232.

Beranek, L. (1962). *Music, Acoustics and Architecture*. New York: John Wiley.

Beranek, L.L. (1992). Concert hall acoustics – 1992. *Journal of the Acoustical Society of America*, *92*, 1–39.

Biot, M.A. & Tolstoy, I. (1957). Formulations of wave propagation in infinite media by normal coordinates with an application to diffraction. *Journal of the Acoustical Society of America*, *29*, 381–391.

Cremer, L., Müller, H.A. & Schultz, T.J. (1982). *Principles and applications of room acoustics. Vol. I*. London: Applied Science Publishers.

Haas, H. (1951). Uber den Einfluss eines Einfachechos auf die Hörsamkeit von Sprache. *Acustica*, *1*, 49–58.

Hammad, R.N.S. (1988). Simulation of noise distribution in rectangular rooms by means of computer modelling techniques. *Applied Acoustics*, *24*, 211–228.

Houtgast, T. & Steeneken, H.J.M. (1980). Predicting speech intelligibility in rooms from the modulation transfer function. I. General room acoustics. *Acustica*, *46*, 60–72.

Kemar (1994). http://sound.media.mit.edu/KEMAR.html

Kuttruff, H. (1991). *Room acoustics*, 3rd edition. London: Elsevier Applied Science.

Lee, H. & Lee, B.H. (1988). An efficient algorithm for the image model technique. *Applied Acoustics*, *24*, 87–115.

Medwin, H. (1981). Shadowing by finite noise barriers. *Journal of the Acoustical Society of America*, *69*, 1060–1064.

Ouis, D. (2002). Scattering by a hard half plane: experimental evidence of the edge diffracted wave. *Journal of Sound & Vibration*, *252*, 469–477.

Schroeder, M.R. (1981). Modulation transfer functions: definition and measurement. *Acustica*, *49*, 179–182.

Schroeder, M.R., Gottlob, D. & Siebrasse, K.F. (1974). Comparative study of European concert halls: correlation of subjective preference with geometric and acoustic parameters. *Journal of the Acoustical Society of America*, *56*, 1195–1201.

Sommerfeld, A. (1954). *Lectures in theoretical physics, Vol. IV, Optics*. New York: Academic Press.

Steeneken, H.J.M. & Houtgast, T. (1980). A physical method for measuring speech transmission quality. *Journal of the Acoustical Society of America*, *67*, 318–326.

Stephenson, U. (1990). Comparison of the mirror image source method and the sound particle simulation method. *Applied Acoustics*, *29*, 35–72.

Torres, R., Svensson, U.P. & Kleiner, M. (2000). Computation of edge diffraction for more accurate room acoustics auralization. *Journal of the Acoustical Society of America*, *109*, 600–610.