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COMPUTER MODELS IN ROOM ACOUSTICS - STATE OF THE ART[1]

Michael Vorländer, Institute of Technical Acoustics, RWTH Aachen University, *mvo@akustik.rwth-aachen.de*

Plenary Lecture

Abstract – *In this paper the basic algorithms of room acoustical computer simulations, the observed uncertainties and recent activities in real-time modelling are summarized. Computer simulation software for sound fields in rooms is commercially available for application in research and consulting. Typically these computer models are based on geometrical room acoustics and/or on statistical methods. Accordingly they do not include simulation of wave phenomena such as diffraction or modes. The uncertainty of typical simulation software was investigated in international verification tests. The results were partly promising but many programs were not as reliable as the operators expected. Recently advanced models have been developed for real-time application such as virtual environments.*

1. INTRODUCTION

Computer modelling of room acoustics was proposed during the 1960’s by Schroeder [1] and practically used first by Krokstad et al. [2]. Although scale model experiments are a powerful tool still today, computer simulations are more and more taking over the part of scale models in consulting. Commercial software became more user-friendly, more accurate and, last but not least, cheaper than scale models. As soon as the room drawing is transferred into a computer file and the wall data, source and receiver locations are stored, the sound propagation in the room can be simulated quite fast and modifications can be tested without large effort.

The algorithms of typical programs are based on geometrical acoustics. In geometrical acoustics, diffraction and interference effects are neglected. The description of the sound field is reduced to energy, transition time and direction of rays. This approach is correct as long as the dimensions of the room are large compared with wavelengths and if broadband signals are considered. These approximations are valid with sufficient accuracy in large rooms intended for speech and music above cut-off frequency f_c . shown below

$$f_c = 2000\sqrt{T/V} \tag{1}$$

with V denoting the room volume in m^3 and T the reverberation time in s.

2. GEOMETRICAL ACOUSTICS: RAY TRACING AND IMAGE SOURCES

In geometrical acoustics the two basic models of geometrical sound propagation, rays and image sources are applied. They differ in the way of calculation ray paths. Image sources are constructed for a given geometry, and valid ray paths are derived from back-tracing from the receiver to the original source along a chain of image sources. In "ray tracing", the propagation of sound is constructed in a "forward" procedure, from the source to the receiver. Whether we use the models of image sources or of rays, therefore, makes no difference in the results. An important difference, however, must be taken into account as regards the modelling of sound energy. One group of models are based on a statistical approach (Monte Carlo method). The energy is obtained by counting events, just as it is done in nuclear physics with a Geiger counter.

2.1 Ray Tracing

Conventional ray tracing is based on the idea of discrete sound rays or sound particles propagating on straight lines from wall to wall. The detected energy is related with the number of rays hitting a receiver volume (typically a sphere) which is hence just a question of counting events. The distance law of free field energy propagation is implicitly included since the ray density and the rays hitting a receiver decreases with r^{-2}.

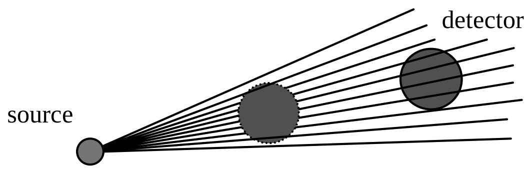

Fig.1. *Monte-Carlo ray tracing: The distance law is implicitly included in the ray density*

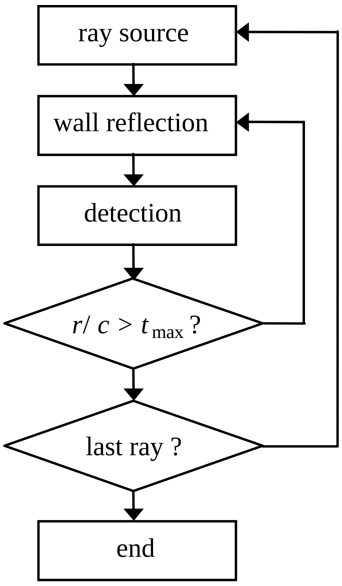

Fig.2. *Ray tracing flow chart.*

[1] This contribution is an updated version of a previous paper published in: Acoustical Society of The Netherlands, Nieuwe ontwikkelingen in de zaalakoestiek., NAG journaal nr. 158, nov 2001, p. 39-50.

The flow chart (figure 2) is illustrating the conventional ray tracing algorithm. Sound sources can be modelled by distributing rays with a certain power (number of rays) and directivity (distribution of rays in the solid angle). Alternatively the directional characteristic can be accounted for by direction-dependent ray energies. The procedure “wall reflection” is the most important loop of the algorithm since it is the inner sub-routine. Here, the next reflection point must be determined. The ray is considered mathematically as straight line, and the walls are modelled by planes surrounded by polygons. The main test is whether an intersection point of a straight line with a plane is located inside or outside a polygon (figure 3). Detection is happening if the distance of the ray (straight line) is smaller than the detector’s radius.

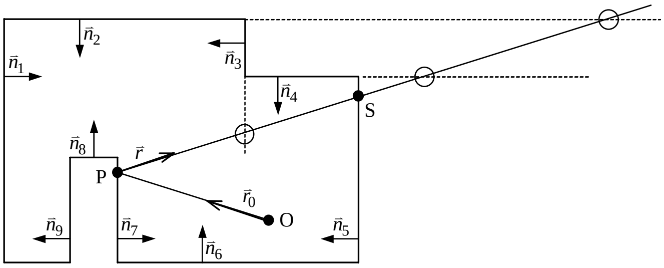

Fig.3. Determination of the next reflection point S. Points marked by ○ *are outside the walls' polygons.*

The rays are reflected according to Snell’s law with incident angle equals reflected angle. Rays can also be scattered according to Lambert’s law [3]. The direction of reflection is then chosen from random numbers. Finally, absorption must be modelled, and there are two possibilities: Firstly, the ray energy can be decreased by a factor (1-α) with α the absorption coefficient. Secondly, the ray can be annihilated by a stochastic process if a random number is smaller than α.

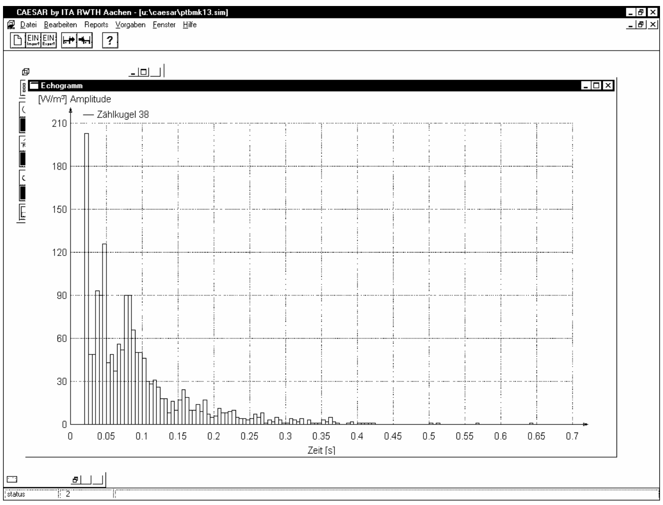

Fig.4. *Energy impulse response (ray tracing simulation)*

In case of detection, the energy and the transition time is recorded in time intervals (typically some milliseconds). The results are energy histograms. i.e. low resolution energy impulse responses as illustrated in figure 4.

Energy impulse responses can further be processed for various single number quantities. According to ISO 3382, important parameters are Strength G, Definition D, Clarity C, Early Decay Time EDT, Lateral Energy Fraction LF, etc. They are obtained after integration of energy impulse responses. If impulse responses have been calculated for complete audience areas, the single number quantities can be visualised in colour maps (example in figure 5).

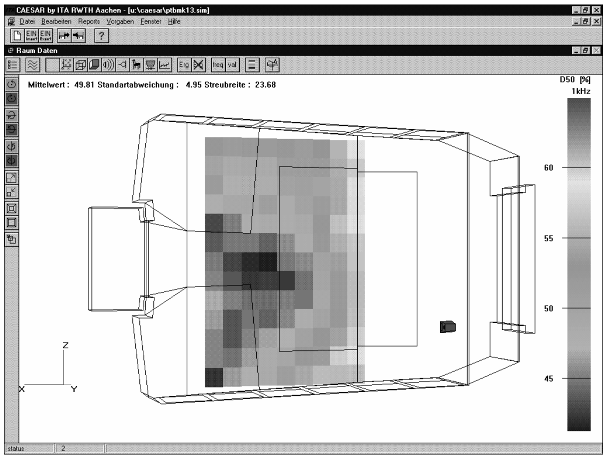

Fig.5. Example of Definition (Deutlichkeit). Distribution on the audience in a lecture hall.

2.2 Image sources

Another basic algorithm for geometrical room acoustics is the image source model. The most important restriction is that random scattering is not possible. On the other hand, the great advantage is that impulse responses are not limited to energy histograms, but can be calculated with extremely high temporal resolution.

Provided each wall reflection is purely specular, the image source principle can be formulated [4]. At first, image sources must be constructed for the room under test. The original sound source S is mirrowed at each wall to construct image sources of first order. With the first-order image sources the procedure is repeated. Hence sources are created which represent rays with several wall collisions, taking into account all possible permutations of the collision order due to the wall numbers (figure 6). This procedure is repeated until a chosen maximum order of image sources is reached. After that, a so-called „visibility test“ (audibility test) is carried out. This test is receiver-dependent and is necessary to check the validity of image sources. Each source is the last link in a chain of successive image sources and, with its indices, one can read which walls were hit by the associated ray path. A chain related to a source of *i*th order is

$$S \rightarrow S_{n_1} \rightarrow S_{n_1 n_2} \rightarrow \ldots \rightarrow S_{n_1 n_2 \ldots n_{i-1}} \rightarrow S_{n_1 n_2 \ldots n_i} \qquad (2)$$

where $n_k \neq n_{k\pm1}$ can have values from 1 to the total number of walls. In the construction process, some geometrically shadowed regions can be excluded from the mirroring algorithm which reduced the number of constructed sources [5]. In figure 6 the audibility test is illustrated for an image source of second order. The actual receiving point R is connected with the image source (S_{12}), the audibility of which is to be checked. Its last index indicates that wall plane 2 was involved last in the complete sound path and in the mirrowing algorithm. If the intersection point of this connecting line is situated within the polygon that defines the actual wall portion on this plane, the image source is (provisionally) considered audible. In this case, we proceed in the same way, considering now the intersection point as receiver point R and aiming at the predecessor (S_1) in the chain of image sources. This procedure is repeated until it ends at the original sound source. If at least one of the calculated intersection points is not situated within the real wall boundaries, the image source is not audible and, hence, will be omitted. The total sound field can be composed by adding the contributions of all audible sources according to the $1/r^2$ law which sets as well the amplitude of each contribution as its time delay. The directivity of the original source, the wall absorption and properties of the receiver can be modelled easily and with amplitude and phase information, if necessary (see below).

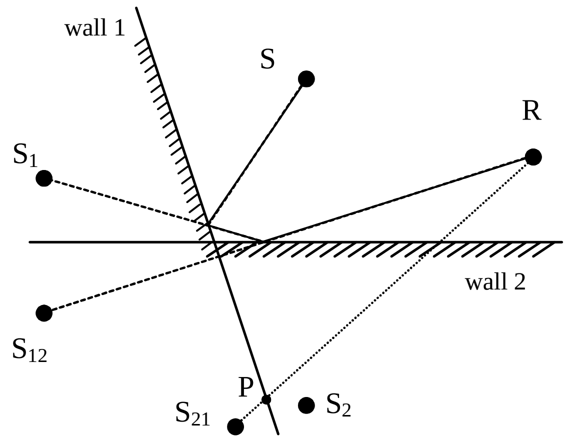

Fig.6. *Construction of rays by using image source simulation*

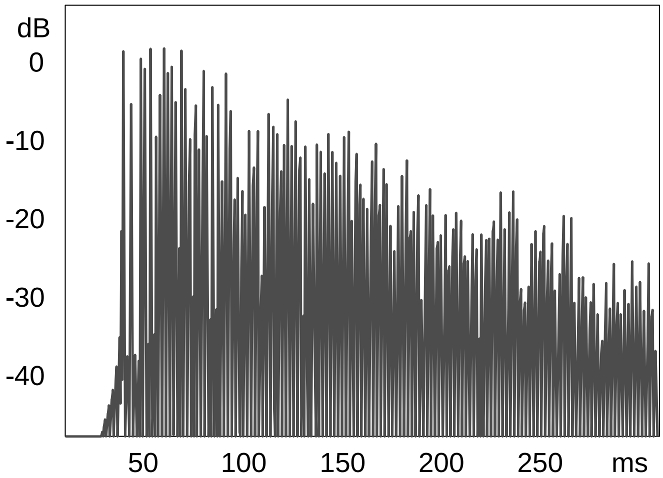

Fig.7. *Energy impulse response (image sources). Compare with figure 4.*

2.3 Hybrid models

Deterministic models like the image source model may suffer from systematic errors introduced by the model itself and thus by limitations of geometrical acoustics in general. Due to the contradictory disadvantages of ray tracing and image sources it was tried to combine the advantages in order to achieve high-precision results without spending too much complexity or computation time. Either ray tracing algorithms were used to overcome the extremely high calculation time inherent in the image source model for simulation of the late part of the impulse response (adding a reverberation tail), or ray tracing was used to detect audible image sources in a kind of „forward audibility test". The latter procedure is named „cone tracing", „pyramid beam tracing" etc. The idea behind is that a ray detected by a receiver can be associated with an audible image source. The order, the indices and the position of this image source can be reconstructed from the ray's history with storing the walls hit and the total free path. Hence the total travel time, the direction and the chain of image sources involved can be addressed to the image source. The calculation time of this kind of image source algorithm is no longer increasing drastically with the length of the impulse response, t_{IR}, but with the third power:

$$t_{\text{calc}} = (4c^2 \bar{n} t_{\text{IR}}^3 / r_{\text{r}}^2)(n_{\text{w}} + n_{\text{r}}) \cdot t_0 \qquad (3)$$

r_r denoting the receiver radius, n_w and n_r the number of walls and receivers, respectively and $\bar{n}$ the mean reflection rate ($cS/4V$) [3]. The calculation time of this hybrid ray tracing model is different from the corresponding statistical ray tracing since a specific relation between the number of rays and t_{IR} must be considered in order to detect all relevant image sources contributing until t_{IR}. It should be noted that the calculation speed of the „forward audibility test" is faster than the classical algorithm by orders of magnitudes, at least from a certain transition order (figure 8).

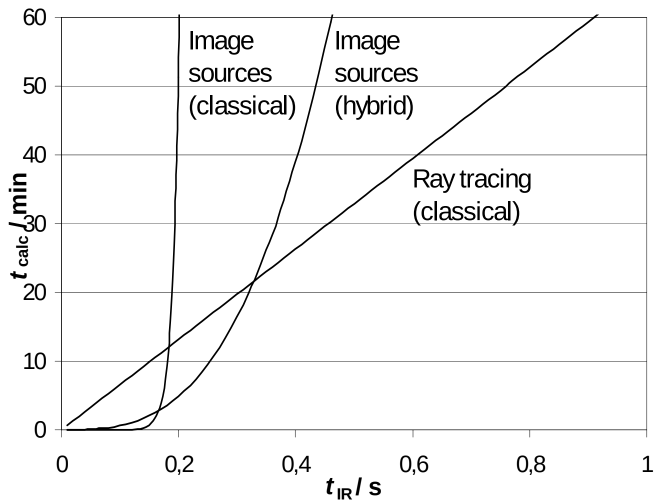

Fig. 8. *Example of calculation times (Pentium PC)*

Almost all other algorithms used in commercial software are kind of dialects of the algorithms described above: "cone tracing, beam tracing pyramid tracing, etc.". The specific choice of dialect depends on the type of results, particularly on the accuracy, spatial and temporal resolution. One important example for specific demands is the so-called „auralisation", where the impulse responses are intended to be used for filtering reverberation-free (dry) speech or music.

3. AURALISATION

Impulse responses for auralisation purposes must be generated with a temporal resolution adequate to the sampling rate of the test signals. It is well known that usually 44.1 kHz is used as standard audio sampling rate. In room acoustics music or speech may be sampled at half that rate which still enables signals to be used with frequency components up to 10 kHz. This upper limit is also typically the limit of knowledge of absorption coefficients. Accordingly the time resolution of the impulse response is less than 50 µs which illustrates the demands for the simulation. Conventional ray tracing cannot be used since the statistical approach generates only rough temporal histograms. The number of rays would be much too high if the (very small) time intervals must be filled with reflections. On the other hand, image source algorithms are well qualified, at least up to a certain length of the impulse response (see figure 8).

It is clear that auralisation involves binaural hearing. Mono reproduction gives some impression of the simulated sound but a very important feature of hearing in rooms is then omitted. In binaural technology, interaural delay and diffraction around the head are basic parameters. Assuming linear sound propagation, both is included in so-called „head-related transfer functions“ HRTFs or in corresponding HRIRs (head-related impulse responses). HRTF data were measured at different places and for different purposes. They can be determined with probe microphones and a group of test subjects, or they can be measured with dummy heads. This aspect is also important for comparison of simulated and measured binaural room impulse responses.

Overall, the binaural sound pressure impulse response is composed of numerous reflections. For each reflection the corresponding complex spectrum H_j can be calculated as follows (after Dalenbäck [6]):

$$H_j = \frac{e^{-jkr_j}}{r_j} H_S H_R H_a \prod_{i=1}^{n_j} H_i \qquad (4)$$

with r_j denoting the distance between image source and receiver, k the wave number = $2\pi f/c$, H_S the (directional) spectrum of the source, H_R the HRTF (right or left ear), H_a the air absorption, and H_i the reflection factors of the walls involved. The same equation can, of course, be expressed in the time domain by convolution of all relevant impulse responses. The total binaural impulse response (r, l = right, left ear) is then obtained by inverse Fourier Transformation:

$$h_{\text{total,r,l}}(t) = \mathcal{F}^{-1}\left\{\sum_{j=1}^{N} H_{j,\text{r,l}}\right\} \qquad (5)$$

Due to the fact that the later part of the impulse response needs not to be created with the same accuracy as the early part, it is often tried to apply hybrid methods by separating the simulation into two or more steps. This is justified by the low energy contained in the late part and by temporal masking. With running speech or music the later part of the reverberation tail is masked and only noticeable with impulsive or rhythmic signals. The early part is considered important at most up to a length of 400 – 500 ms. The methods for simulation of exact binaural impulse responses are described above. One principle for creation of reverberation tails is based on the assumption of an exponential decay envelope with reverberation time estimated from Sabine's or Eyring's formula. The fine structure in this diffuse field approach is purely stochastic and can be taken, for instance, from a Poisson process of Dirac pulses. Octave band filters and summation enable transition into narrow-band or broadband sound pressure signals. Another possibility replaces the exponential decay by the envelope found by a low resolved ray tracing, which is more accurate than using Sabine's formula.

The appropriate playback arrangement is either headphone presentation or free field reproduction with loudspeakers and crosstalk cancellation. The equalisation of the headphone or the loudspeakers is extremely important if the sound localisation and spatial impression should be created correctly. However, the binaural technology is well investigated and can be applied for these purposes (see e.g. [8]).

The criteria for binaural impulse responses can be summarised as follows:

Table 1: *Features of impulse responses for auralisation*

Parameter	Order of magnitude	Reason
temporal resolution	50 µs	Sampling rate and Nyquist frequency
upper frequency limit	≥ 8 kHz octave	Highest components in music and speech
lower frequency limit	≤ 63 Hz octave	Lowest components in music and speech
Length	half reverberation time: *T*/2	Minimum required at temporal masking of music or speech (not valid for percussion)
accuracy of time structure (early)	50 µs	Early reflections (< 100 ms) must have exact (binaural) delay
accuracy of time structure (late)	5 - 10 ms	Statistical superposition of late reflections allows less resolution, as long as the envelope is correct

4. VERIFICATION TESTS

The efficiency and accuracy of computer simulations can only be checked if existing rooms are modelled and the results compared with measurement results. Auralisation, of course, can also be checked in listening tests with recordings in the original room. The requirements on the playback system have then particular importance.

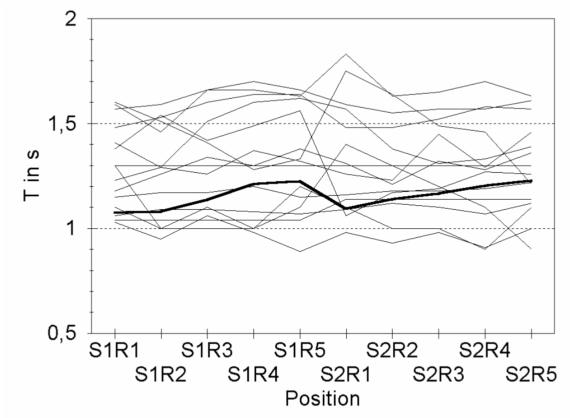

Fig.9. *Reverberation time: Results of simulations and measurement (thick line), see[9].*

Validity tests can be performed in intercomparisons [9]. The participants are asked to simulate a certain room under certain conditions (with or without audience or orchestra, etc.). If the room under test is existing in reality, impulse responses can in parallel be measured and serve as reference. This procedure was carried out recently in a first intercomparison in Braunschweig in 1993 and 1994, Germany, on a lecture hall. The first results were partly disappointing [9].

It was evident that many operators tend to estimate absorption coefficients too low and predict, thus, too large reverberation times (figure 9). The differences exceeded 50% in some cases. Even „simple" quantities like sound level (strength) were predicted with up to ± 5 dB maximum deviation. In an overall accuracy rating, however, only a few programs were identified reliable.

Due to the fact that room acoustical algorithms do have some problems in practical cases, it was reasonable to continue the intercomparison and to extend the work to more frequency bands and to another room shape and volume. The aim was to give the software developers the opportunity to improve their programs and to give better advice to the operators how to choose input parameters.

Similar problems and additional information was achieved in the evaluation of the third round robin [10], http://www.ptb.de/en/org/1/14/1401/_index. Today, the fourth project is planned, with a curved-shape church as test object. More to be announced under http://www.eaa-fenestra.org//TCs/RBA.

5. Typical limitations on accuracy

The reasons for deviations between simulations and measurements are shortcomings in the algorithms and the modelling approach behind it. As described before, the ray tracing and the image model are the basis for all simulations. In the following, some examples are discussed in which the physics of wave propagation is only too roughly approximated. Errors may possibly occur, and it is the question if these affect parameters like reverberation time or clarity, or if the approximations are audible.

5.1 Curved surfaces

To the authors knowledge, none of the simulations packages allows modelling of curved surfaces. Usually curved surfaces are approximated by a number of planes. Curved surfaces produce very special features like focus points or caustics. The questions is if an approximation by planes produces a focus as well and if the sound level in the focus region is correct ([11, 12] It was shown that only deterministic approaches with coherent image source contributions can be used. This is obvious since the sound pressure in the focal region cannot be obtained by energetic models. The approximation of a cylinder with radius a by planes of width b is sufficient if

$$f\ b^2 = \frac{1}{2} c\ a \tag{6}$$

is fulfilled. It is interesting that this equation is frequency dependent. Thus, the condition must be checked for the highest frequency band involved, or the shape must be modelled frequency dependent. For example, a cylindrical shape with radius 10 m must be modelled by panels of width 50 cm, if the frequency range should reach up to 8 kHz, which means that the cylinder should be subdivided into 136 (!) plane elements.

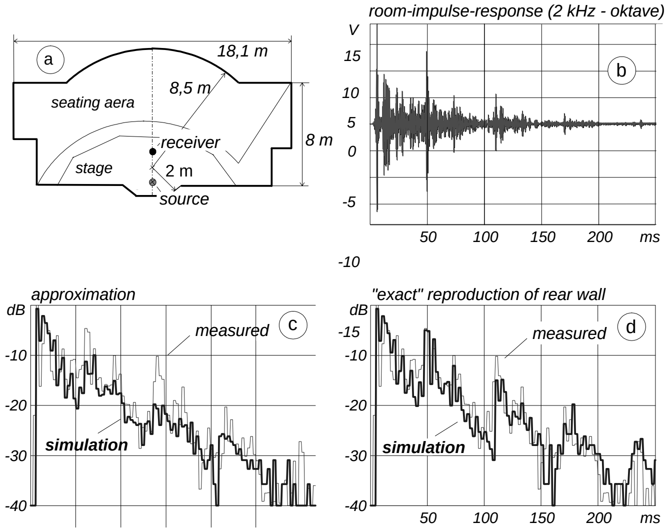

Fig. 10: *Impulse responses from curved surfaces (after [12])*

5.2 Scattering

Surface scattering occurs if wall surfaces are rough. The specific reflection pattern depends strongly on the frequency. However, with diffuse field conditions and the corresponding uniform sound incidence, not the detailed reflection characteristic is needed, but knowledge about a random-incidence scattering coefficient s which can be defined as the ratio between the scattered sound energy and the totally reflected sound energy.

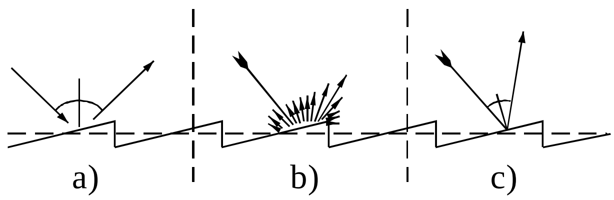

Fig. 11. *Reflection from rough surfaces, low (a), mid (b) and high (c) frequencies.*

This approach is based on the fact that a spatial scattering characteristic is composed by directional lobes of different order, the zero-order lobe of which corresponds with the specular reflection. Thus it is reasonable to define s by means of the energy of the zero order lobe (with reflection factor R_{specular}) and the reflected energy 1-s:

$$\delta = 1 - \frac{\left|R_{\text{specular}}\right|^2}{1-\alpha} \tag{7}$$

Hence it is the task of the measurement to determine the random-incidence scattering coefficient, in order to create a database similar to data on absorption coefficients. One idea for a measurement technique makes use of an impulse test on a surface sample and the phase shifts, if the surface under test is moved or turned. The specular component, however, doesn't change phase and direction but the amplitude only. Thus, by correlated summation during the test sample movement, the specular lobe increases by correlated superposition, and the scattered field cancels by stochastic interference. The technique is applicable also in reverberation rooms [13] and is therefore expected to be standardised in an ISO standard [14], similar to the ISO method for measurement of absorption in the reverberation chamber. In figure 12 an example (see also [13]) is shown for a free field measurement on a structure of hemi-cylinders and rectangular battens.

The results indicate that – in agreement with theory – the scattering coefficient is nearly zero for frequencies smaller than $c/2a$, (with a the roughness size) and increases to values around 0,4 to 0,8 above that frequency. If the frequency, however, is much larger than $c/2a$, the surface roughness must be modelled with its specific shape in all details.

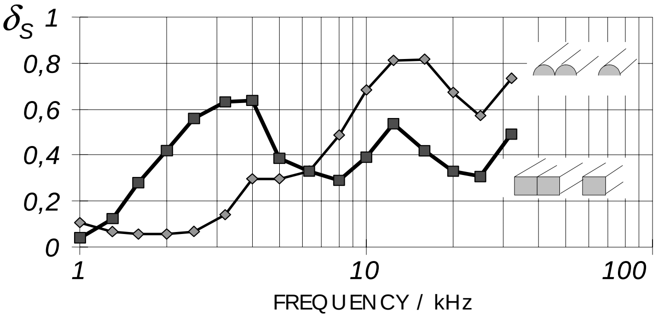

Fig. 12. *Scattering coefficients of battens placed on a rigid surface*

In addition to the discussion of the effects of random-incidence scattering, one is interested in the specific directionality of scattering. This would be necessary in simulation of dominant effects of first-order scattering, for instance in almost "dry" rooms like recording studios. To describe this feature of a wall surface, the diffusion coefficient and the uniformity of the scattering characteristic are important parameters [15].

5.3 Diffraction

Diffraction in room acoustics mainly happens for two reasons: There can be obstacles in the room space (e.g. stage reflectors), or there can be edges at surroundings of finite room boundaries. In the latter case, either the boundary is forming an obstacle, such as columns or the edge of an orchestra pit, or the boundary is forming the edge between different materials with different impedances (and absorption). Since diffraction is a typical wave phenomenon, it is not accounted for by the basic simulation algorithms listed above. In the past there were some ideas of including diffraction as a statistical feature into ray models. But the success was quite limited because the increase in calculation time is a severe problem. In optics and radiowave physics, ray tracing models were generalised into so-called UDT (uniform geometrical diffraction theory [16]). Other approaches were presented by Svensson [17], who applied the model by Biot and Tolstoy [18]. They are very powerful for determination first-order diffraction. All methods of geometrical diffraction are, however, very time consuming for simulation of a multiple-order diffraction and corresponding reverberation.

Another possibility is to apply finite element or boundary element methods. These methods are based on wave theory in a discretized formulation of the problem. At first the room shape (boundaries or volume) must be discretized in a mesh with certain boundary conditions (pressure, velocity or impedance). The sound propagation is simulated in the frequency domain by solving the simultaneous Helmholtz equations for all mesh elements (see 5.5).

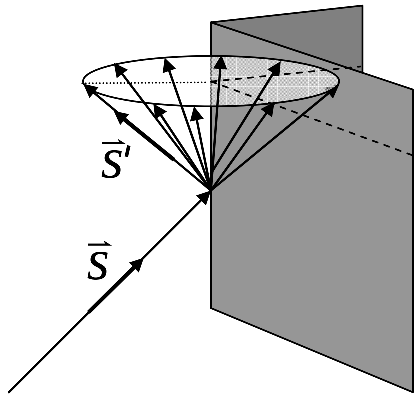

Fig. 13. *Scattering according to Fermat's principle in UDT.*

5.4 Audience and seats

Concerning the audience we have to separate two effects: The diffraction of sound by the heads and shoulders of the audience and the so-called „seat-dip effect“ which is a kind of resonance in the depth of the seat rows. Both effects were investigated by theory and measurements, and they are found to be particularly important at grazing sound incidence (below 7^o incidence angle). The effects can be described by a linear filter depending on the row spacing and the seat arrangement. In the simulations the effect can be introduced as an additional filter for the grazing incident reflections (see eq. (5) and [12, 19]).

5.5 Spherical waves

Scattering, diffraction etc. are examples of wave phenomena which are not covered by geometrical acoustics. Nevertheless it is sometimes assumed that image source algorithms including the possibility of complex wall reflections factors can yield “correct” modal sound fields in rooms. This, however, is a wrong assumption. The image source model is a correct solution of the wave equation for one rigid boundary. For one non-rigid boundary, it can be shown that the spherical wave solution based on complex impedance is a good approximation, if the position of source and receiver are located not too near (> one wavelength) to the wall. This is basically the content of the large room assumption. The relevant dimension compared with the wavelength is then not the room volume but the smallest dimension, height or width. In flat rooms or in long rooms like corridors grazing incidence angles may occur quite often, so that the plane wave reflection is too rough an approximation at low frequencies.

Therefore, in all cases where room modes are to be calculated, in small studio rooms, in living rooms or in other examples, only wave-based models can be used, such as BEM or FEM or similar.

6. REAL-TIME MODELS: VIRTUAL REALITY

Virtual reality (VR) is the technique of creating a computer-generated environment which allows real-time interaction between human and machine. Concepts of virtual reality have a wide variety of applications such as of human-machine interaction, product prototyping and evaluation, visualisations in engineering, scientific and medical applications. This includes visual, aural and haptic aspects of the presented scenery. Immersing a human receiver into a virtual room geometry with respect to aural stimuli is not a trivial task, as position, level, and directivity of each sound source located inside the room as well as their corresponding sound field, which includes sound reflections on the walls, must be synthesised and reproduced in parallel and matching to the visual component in real-time. So far, sound reproduction and simulation is well studied mainly for non-interactive applications, which are mostly not fit for the demands of real-time applications such as VR.

Acoustic models used for real-time VR-applications mainly focus on the determination of low-order specular sound reflections in sceneries of low geometrical complexity. One of the major difficulties is that the temporal aspect differs strongly from graphics as the speed of sound is much slower than the speed of light. Transients have to be computed involving three decades of the frequency range. Thus, fast techniques from computer graphics cannot be applied directly.

For real-time applications, two aspects are important: responsiveness and smoothness, which reflect the overall lag of the simulation and the number of updates per time, respectively. In general, a refresh rate of 60 Hz and 50 ms lag can be regarded as suitable for auditory real-time applications. For sound reproduction in a CAVE (see fig. 14), for instance, a slower update of the presented scenery and its acoustical components is sufficient, as a receiver usually moves slowly through the virtual environment, i.e. the binaural signals do not change rapidly.

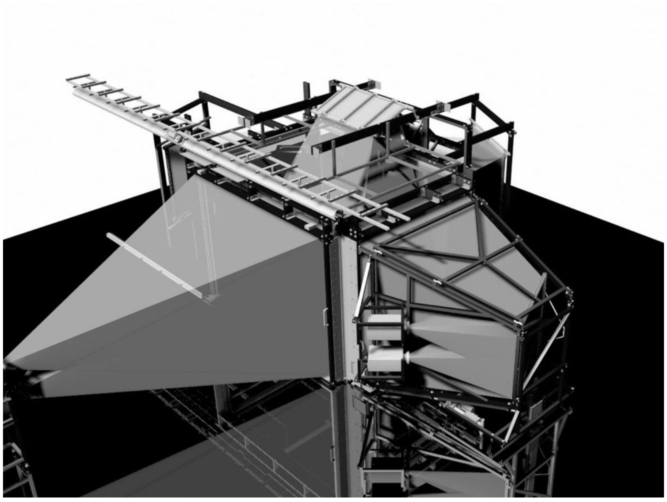

Fig 14. *Cave Automated Virtual Environment (CAVE).*

To apply methods of geometrical acoustics in real time, data structures are required for an efficient representation, determination of spatial relationships between sound rays and the room geometry and fast computation of occurring ray-polygon intersections. In contrast to the naive approach of testing the position of an arbitrary point against all room's planes, the hierarchical binary tree structure of binary space partitioning (BSP) trees [20] allows the determination of a point's location by testing only a subset of planes, which is defined by the path in the BSP-tree.

The principle of BSP is to subdivide an (N)-dimensional space into (N)-dimensional subspaces by means of (N-1)-dimensional directed partitioners. The crucial part is to find appropriate partitioners to generate a balanced tree, i.e. a tree of minimum height. Arbitrary planes could be chosen as partitioner, but it would decelerate the BSP-tree creation unreasonably, as there exists an infinite number of possible partitioning planes in space. The use of non-linear curved partitioners is also possible, but not suitable due to analytical complexity. Therefore, using planes spanned by the polygons as partitioners makes the BSP-tree creation faster and transparent. Although this is suboptimal, it already provides quite compact tree structures.

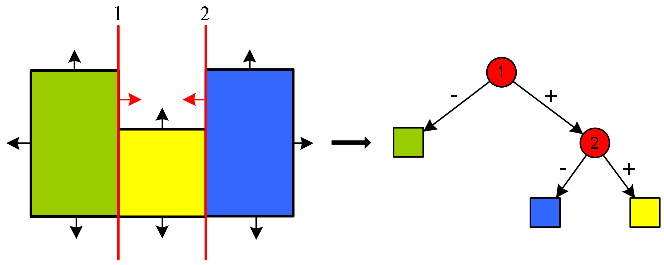

Fig. 15. *Possible BSP-tree for a typical room geometry.*

Each tree node contains one partitioner that divides the current (sub-)space into two smaller subspaces, whereas a leaf node refers to a subspace that cannot be further subdivided reasonably, i.e. the remaining partition is a convex set of polygons (see fig. 15). The point in question can have three possible relative locations regarding the partitioning plane: a) on, b) in-front-of and c) behind the partitioner. In the first case, the query can be stopped. The latter two cases are pursued by branching to the respective son of the node, left for `behind', right for `in-front'. If a leaf node is reached, the query terminates as well, because the final point's position has been determined.

By using a balanced tree the number of required tests can be minimized, which drops the complexity from O(N) up to O(log(N), where N is the number of polygons, which apparently speeds up the point's location search significantly. For the determination of intersections between a ray and the scene's elements, i.e. polygons, a BSP-tree based search is a very fast and efficient strategy. Similar to the determination of a point's location relative to a scene by querying the BSP-tree, this can also be done recursively with a ray connecting two points, i.e. source and receiver.

The great advantage of the BSP-tree structure is the possibility to control the search behavior. As for instance the image sources audibility test (see 2.2) demands only ray-polygon intersections closest to the receiver, the BSP-tree traversing strategy can be modified efficiently to satisfy real-time demands [21]. Another high performance approach for beam-tracing has been proposed by Funkhouser et. al [22] who also uses hierarchical tree-structures to gain real-time capability.

CONCLUSIONS

It is obvious that computer simulations in room acoustics are an important tool in research and architectural design. The progress in the last decade made it possible that computer simulations can be applied in practice without large doubts on the accuracy. However, due to the complexity of acoustical problems, sources of errors must be identified. The shortcomings and limitations are known and must be taken into account by a skilled operator or by software support during the room design. The latter point is particularly important for commercial programs if the software engineer has no direct influence on the choice of input parameters by the operator. It is hence a question of an efficient user manual or automatic control of input errors.

Research activities are focused on several improvements of the algorithms and on new algorithms. Recently published or actual work on room acoustical computer simulations, for instance, can be observed in the following subjects:

- Low frequencies (modelling of diffraction by boundary elements)

- Auralisation (extended sources like choir or orchestra, moving sources)

- Fast prediction of non-diffuse sound fields (workrooms, factories)

- Computation speed (real-time applications, virtual reality)

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