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DIGITAL SIMULATION OF CONCERT HALL ACOUSTICS AND ITS APPLICATIONS

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1. INTRODUCTION

In recent years, great progresses have been made in computer simulation of concert halls, and it can be expected, that these techniques will replace the more conventional methods of acoustical design in near future, on account of their efficiency and flexibility. But even more: computer simulation offers the possibility to give direct, i.e. aural impressions of the listening conditions expected at the various seats of the hall when it is still in the state of design ("auralization").

The classical tools of the acoustical consultant are a pencil and a drawer to trace sound rays and to construct sound reflections. Another well-proved method of acoustical design is a scale model of the hall in which important aspects of the sound propagation can be investigated by experiment. With sufficiently refined equipment, we can even use a model to present aural impressions on how music will sound in the hall when it will be completed. According to the original ideas of F. Spandöck /1/, who has proposed this method as early as in 1934, this can be done as shown in fig.1:

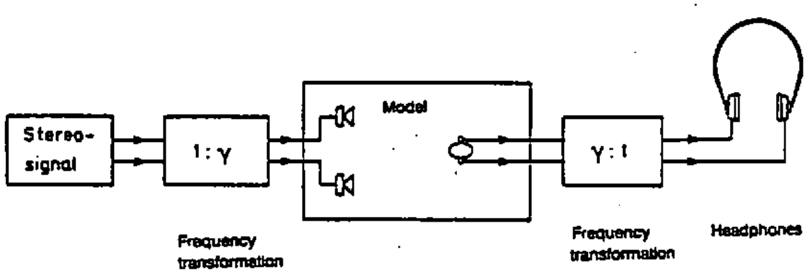

Fig. 1.: Principle of auralization carried out with a physical model

In the scale model, which should be as realistic as possible, "dry", i.e. unreverberated sound signals such as speech or music are replayed from a tape recorder and re-recorded with miniaturized transducers. Then the signal processed in this way is presented to the listener with earphones. Of course, this process requires a two-fold frequency transformation since in the model not only the geometrical dimensions, but also the wavelengths of all spectral components must be reduced by the scaling factor, compared with the normal audio frequency range.

Although model techniques have reached a high degree of perfection nowadays, for instance by the work of J. Blauert and his co-workers in Bochum /2/, or of R. Orlowski in Salford /3/, this procedure is still plagued by several difficulties. Maybe the most severe of them is the necessitiy of achieving high fidelity sound reproduction in the ultrasonic frequency range, and to model correctly the absorptive properties of the boundaries and of the air. Therefore it seems more promising to replace the physical scale model in fig.1 with a kind of digital filter the characteristic parameters of which are obtained by computer simulation of the hall. In this case no frequency transformations are needed, of course. The geometrical and acoustical data of the room are just entered into the computer from the input terminal, the absorption of the walls and of the medium can be taken from literature data. Before discussing the process of auralization, I shall give you a short description of the current methods of sound field simulation in concert and other halls.

2. RAY TRACING

Probably the first authors who applied digital simulation of sound fields to concert hall acoustics were Krokstad, Strom and Sorsdal /4/. The method they developed is known as ray tracing nowadays. Like all geometrical methods it neglects phase differences between different components of the sound field; therefore neither interference nor diffraction effects can be accounted for. This is probably justified for large halls and for sound signals with a wide frequency bandwidth.

The principle of digital ray tracing is outlined in fig.2: A sound source at a given position is imagined to release numerous sound particles into all directions at time t=0.

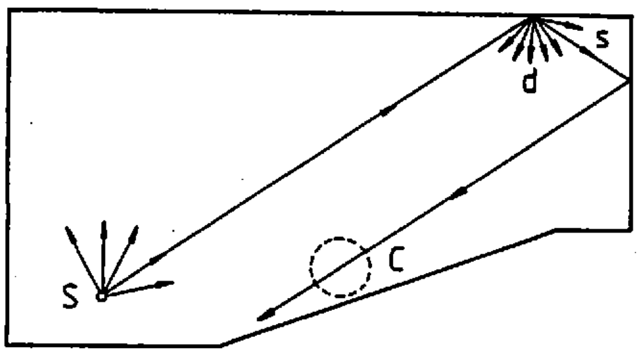

Fig. 2: Principle of ray tracing

Alternatively, we could prescribe a desired directivity to our source. Each sound particle travels along a straight path until it hits a wall which is assumed as plane. At the intersection point the particle will be reflected, either specularly or diffusely. In the first case, its new direction is calculated from the law of geometrical reflection; if diffuse reflection is assumed to occur, the new direction is calculated from two random numbers distributed in such a way that on the average Lambert's law of diffuse reflection is fulfilled. After its reflection, the particle continues its way in the new direction towards the next wall etc. Basically, the absorption of the wall can be accounted for basically in two ways: either by reducing the energy of the particle according to the absorption coefficient, or by interpreting the absorption coefficient as an "absorption probability", i.e. by generating another random number which decides whether the particle will proceed or whether it has been absorbed. In a similar way, the effect of air attenuation can be taken into account. As soon as the energy of the particle has dropped below a prescribed value or the particle has been absorbed, the path of the next particle will be "traced". This procedure is repeated until all the particles emitted by the sound source at t=0 have been followed up.

The results are collected by means of "counters", i.e. by counting areas or counting volumes assigned previously. Whenever a particle crosses such counter its energy and arrival time is stored, if desired also the direction from which it arrived.

After the process has been finished, i.e. after the last particle has been followed up, the energies of all particles received in a certain counter within prescribed time intervals are added, the result is a histogram (see fig.3) which can be considered as a short-time averaged energetic impulse response. The width of the time intervals determines the achieved resolution; however, if it is too small, the result is degraded by statistical fluctuations unless the number of particles and hence the computation time is not increased.

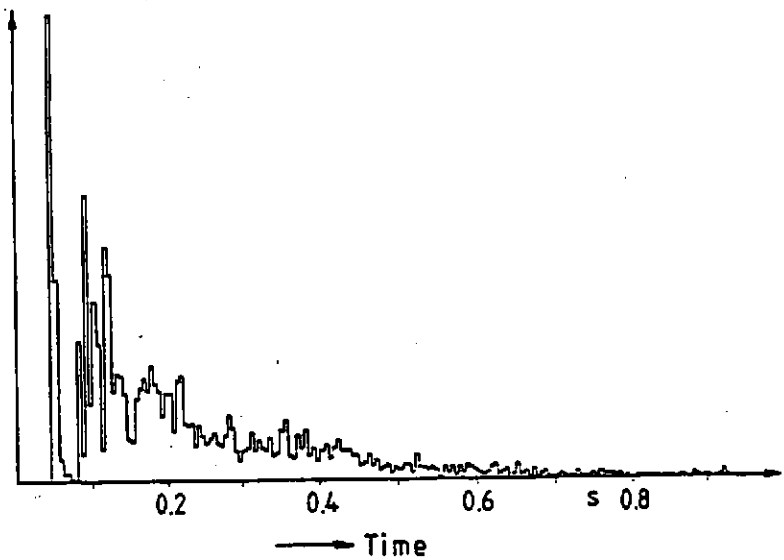

Fig. 3: Histogram obtained by ray tracing

From the energetic impulse response obtained in this way the usual parameters can be evaluated, which give at least some indication on whether the sound will be loud enough, for instance, whether music will sound distinct or even dry, or brilliant, full and warm; whether the reverberance of the hall will be sufficient or not and so on. If we evaluate not only the time at which particles are received but also the directions from which they arrive we can predict whether the listener will feel "enveloped" by the sound field, or, on the contrary, he will hear the music strictly from ahead. So this method is quite useful for collecting important objective data on the sound field in the room which are more or less related to subjective impressions. But are its results also suitable for "auralization", i.e. for directly listening to music processed in such a way as if it were transmitted in the hall? At first glance, this is at least doubtful. Therefore we turn to a different method of sound field simulation, based on image sources.

3. IMAGE SOURCE MODEL

The image source concept is quite old in room acoustics. It is based on the simple idea, that a sound ray which is reflected from a plane wall can be thought of as being emitted by an image source which can be constructed by mirroring the original sound source with regard to the reflecting wall. Successive reflections from the walls of an enclosure can be accounted for by higher order image sources, which are obtained by constructing mirror images from earlier image sources. Of course, this process of constructing image sources of higher and higher order never ends, resulting in an infinite pattern of image sources. Once this pattern is known, or at least its most significant part, we can forget of the enclosure, since we just have to add the energy contributions of these image sources.

However, this method of image source construction requires impractically long computing time. One reason for it is the tremendous number of image sources needed which can easily amount to say 10^9 for realistic conditions. But even worse: only a small fraction of these image sources do really contribute to the sound energy received in one particular point, because most of them turn out to be "invisible" from that point due to the finite dimensions of the wall planes. Now, for each receiver or listener position the visibility of all image sources has to be checked anew, and it is this check, which requires particularly long computation time.

Fortunately, Vian and van Maercke /5/, and independently Vorländer /6/ have found a way to determine the locations of just the valid image sources leaving apart the invalid ones. This is achieved by an abbreviated ray tracing process which preceeds the actual simulation: Whenever a sound particle released form the original sound source arrives at a counter, it must have passed a certain sequence of image sources, which can be determined by backward tracing its fate. After running the ray tracing for a certain while one can be sure that all significant image sources have been found including the walls which are involved in their formation, and no visibility check is needed, because all of these images are visible.

The result of this procedure is the correct energetic impulse response of the transmission path connecting the sound source with the considered receicing point. in contrast to the crude approximation to it obtained with the ray tracing techniques. An examnle of such a "reflectogram" is presented in Fig.4.

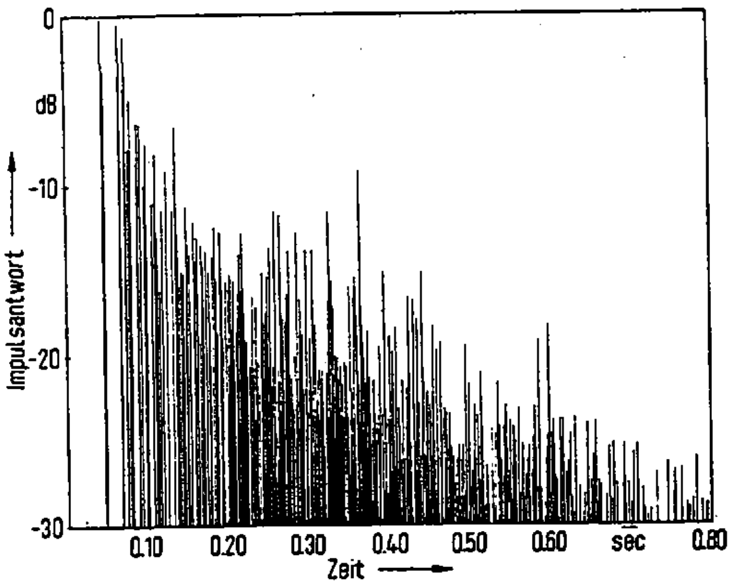

Fig. 4: Energy impulse response (reflectogram) obtained with the imaqe source model

Each vertical line marks the correct moment of its arrival, and its length is proportional to its strenqth. And we know as well the directions, from which each reflection arrives at the observation point. Therefore, all numerical parameters needed for assessinq the acoustical qualitv of the room can be evaluated from the result.

However, to use it for the auralization of the sound transmission, two things are still missing. The first one is that a reflectooram of this kind does not contain the frequency dependence of the wall and air absorotion. Thus we still have to include this important fact in some way. The second is that we usually listen with two ears when attendinq a concert or opera performance. Therefore we need at least two impulse responses. namely one for each ear, and each of them must acoount for the sound diffraction around the human head which depends not only on the frequency but also of the direction from which the reflected sound portions hit the head.

4. BINAURAL SIMULATION

If the sound reflective properties of a wall depend on the frequency as they usually do, the wall's response to an incident Dirac impulse is not just another Dirac impulse, but a more complicated time function which could be named "reflection response". It can be calculated from the complex wall impedance (provided it is known). Then, the contribution of one particular image source to the impulse response is obtained by convolving the properly delayed Dirac function with the reflection responses of all walls which are involved in the corresponding ray path. This is the correct method of accounting for the frequency dependence of the wall properties and is being employed by a French group, namely Martin and Vian /7/. Of course, it is very expensive in computing time.

A simpler way is to compute the impulse response of a particular transmission path several times, each time with a different set of absorption coefficients, and with each of them corresponding to another octave band. The obtained result is passed through the proper octave filter, i.e. convolved with its impulse response, and finally all outputs of the octave filters are added to yield the total impulse response /8/. Fig. 5 shows an example which looks quite realistic.

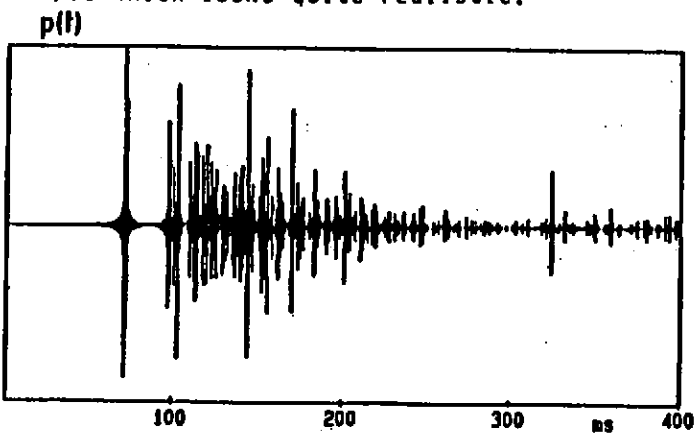

Fig. 5: Reflectogram including frequency dependence of acoustical data

In order make it also sound realistic, the impulse response has to be modified in such a way that it conveys the spaciousness of the real sound to the listener, and this requires binaural presentation of the processed sound signals: When we listen to a sound source in the free field, the sounds arriving at both ears of a listener are modified by the head in a way depending on the direction of sound incidence. This modification can be described by the "outer ear transfer function" or by the Fourier transform of it, the "outer ear impulse response" $h_r(\ldots,t)$ and $h_l(\ldots,t)$ with the subscripts referring to the right and the left ear. To include this modification into our simulation, each component of the room impulse response, i.e. each of the reflections of which it is made up, must be convolved with the proper ear response. The final result are two impulse responses, one for the right and one for the left ear. They represent the characteristics of the filters we need for the process of auralization, and they enable us to process "dry", i.e. unreverberated music signals in the same way as the concert hall would process them, and to present the result to a listener by earphones who now can enjoy the acoustics of a non-existing concert hall in our laboratory. The subjective impression created in this way is surprisingly good /9/, in particular if different halls are immediately compared in this way. Further improvement can be achieved by replacing the earphones with loudspeakers, which can be done by employing the free field cancellation techniques firstly described by Atal and Schroeder /10/. So far, however, it is not possible to perform all these complicated operations in real time; instead, music samples of finite duration must be prepared.

It should be mentioned, that the outer ear transfer functions show considerable individual differences, and that a perfect satisfacory impression cannot be achieved with some average. Therefore, each listener must have measured his personal ear transfer function, strictly speaking, which then has to be combined with the impulse response of the hall. The application of maximum length signals in combination with the fast Hadamard transformation permits relatively fast measurement of such transfer functions /11/. It is possible, however, that even this step can be omitted by storing a number of standardized ear transmission functions.

For real time operation, the whole process of auralization should be simplified. This seems to be possible, since an impulse response as shown in fig.4 contains more information than is needed. This has physical reasons as well as psychoacoustic ones. Firstly, the components of real impulse responses are certainly not so distinct as shown in fig.4, but are blurred by diffuse reflections which are not accounted for by the image model. but which are present in every real hall. Diffraction effects have a similar effect. Concerning the directional structure of the impulse response, we can benefit from the randomizing effect of each real enclosure which makes the reverberant part of the impulse response diffuse; in fact, after the first 150 to 200 ms, the sound field turns out to be diffuse. Hence, it is only the first portion of an impulse response which contains significant directional information, and only this part must be modelled correctly. But even more important are the limited time and directional resolution of our hearing. Therefore we need not use every detail of the physical impulse response for creating a true and natural sounding listening impression. In fact, it seems that the information contained in a ray tracing histogram as shown in fig.3 is sufficient to derive a digital filter from it which can be used for the aural presentation of music.

At any rate, by digital simulation of sound transmission in rooms and by the auralization of the results an old dream of acousticians is going to come true. This methods will be extremely helpful for the practical design of halls of any kind. Furthermore, it is a valuable tool for more fundamental investigations, for instance on optimum reverberation times, on spatial impresssion, optimum ceiling heights, favourable or less favourable room shapes, the arrangement of audience seats and many other interesting questions.

5. REFERENCES

/1/ F SPANDÖCK, Ann. d. Physik V., 20 p345 (1934)
/2/ H ELS & J BLAUERT, Proc. Vancouver Symp. (12th ICA Congr.), p65 (1986)
/3/ R ORLOWSKI, Proc. Vancouver Symp. (12th ICA Congr.), p61 (1989)
/4/ A KROKSTAD, S Strom & S Sorsdal, J. Sound Vibr., 8 p118 (1965)
/5/ S P VIAN & D VAN MAERKE, Proc. Vancouver Symp. (12th ICA Congr.), p74 (1986)
/6/ M VORLÄNDER, J. acoust. Soc. Amer., 86 p172 (1989)
/7/ J MARTIN & S P Vian, Proc 13th Intern. Congr. Acoust., Belgrade, p253 (1989)
/8/ M VORLÄNDER, Acustica, 65 p138 (1988)
/9/ H KUTTRUFF, M VORLÄNDER & T CLASSEN, Acustica, 70 p230 (1990)
/10/ B S ATAL & M R SCHROEDER, Gravesaner Blätter 27/28 p124(1966)
/11/ A SCHMITZ & M VORLÄNDER, Acustica 71 p257.(1990)