Auralization—An Overview*

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Auralization is a term introduced to be used in analogy with visualization to describe rendering audible (imaginary) sound fields. Several modeling methods are available in architectural acoustics for this purpose. If auralization is done by computer modeling, it can be thought of as "true" acoustical computer-aided design. Together with new hardware implementations of signal processing routines, auralization forms the basis of a powerful new technology for room simulation and aural event generation. The history, trends, problems, and possibilities of auralization are described. The discussion primarily deals with auralization of auditorium acoustics and loudspeaker installations. The advantages and disadvantages of various approaches are discussed, as are possible testing and verification techniques. The possibility of using acoustic scale models for auralization is also discussed.

Demonstrations of auralization have been made, but still the technology's ability to reproduce the subjective impression of the audible characteristics of a hall accurately remains to be verified. This limits the credibility of auralization as a design tool, and verification of auralization the foremost problem to be attacked at this time. The verification problem also applies to the basic room impulse response prediction programs. The combination of auralization with transaural reproduction, room equalization, and active noise control could make it possible to expand the applications of the technology beyond the laboratory and beyond simple headphone reproduction. A large number of interesting applications outside room and psychoacoustics research are conceivable, the most interesting of which are probably its use in information, education, and entertainment.

0 INTRODUCTION

Throughout the history of audio and acoustics one aim has been to recreate a particular recording environment or a particular listening environment. Auralization is another step forward in these efforts of presenting a listening experience and is defined as follows: *Auralization is the process of rendering audible, by physical or mathematical modeling, the sound field of a source in a space, in such a way as to simulate the binaural listening experience at a given position in the modeled space.*

The aim is not primarily to recreate the sensation of the speech or music per se, but to recreate the aural impression of the acoustic characteristics of a space, be it outdoors or indoors. Auralization can be done using acoustic (ultrasonic) scale modeling or computer modeling to obtain the binaural room impulse response or transfer functions. The source material, speech, music, and so on, are then filtered by these transfer functions using digital signal processing.

1 HISTORICAL BACKGROUND

The first attempts at aurally creating a planned environment were made by Spandöck and his research team in Munich in the 1930s using physical scale models in a 1:5 scale [C4]. Later a 1:10 scale was applied [C1], [C2]. A custom loudspeaker was used in these scale models and sound was picked up and replayed in a binaural fashion. As a means of checking model accuracy, speech was used as a test signal, and the speech intelligibility was compared between scale model and real room. Other researchers have used similar methods, particularly in Japan. Through the use of digital signal processing it is now possible to overcome the short-

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comings of early physical scale-model auralization using analog techniques.

By using a multiple-loudspeaker auralization system, as outlined in Fig. 1, it is possible to obtain a more flexible system. This method was used by Meyer et al. [A1] and Kleiner [A3]. Using image-source-model computer programs, ray-tracing model computer programs or even manual calculation, it is possible to roughly predict the room impulse response (RIR) so that useful data for room acoustic simulation can be obtained. The multiple-loudspeaker auralization system usually consists of approximately 10 to 50 loudspeakers set up in a hemisphere in an anechoic chamber. At least 50 loudspeakers are necessary in order to allow reasonably correct angles of incidence toward the listener for the various reflections coming from the hall surfaces to the listening position. By feeding the loudspeakers with correctly filtered, attenuated, and time-delayed signals as well as reverberation signals, it is possible to obtain good simulations for the reproduction of speech intelligibility, as shown by Kleiner [A3]. A similar but refined method was used by Bech [A5] and Fincham [A6] in the determination of optimum loudspeaker placement in rooms. Other users of multiple-loudspeaker auralization systems have included various consultants, such as Veneklasen [A2], BBN, Taisei Corp. [A9], Takenaka Corp. [A11], and Kajima Corp. [A12]. The method has also been used as the basis for various commercial products, such as the Yamaha and Lexicon series of audio processors for domestic use.

2 CURRENT AURALIZATION TECHNIQUES

Four basic techniques are available for auralization today. All of the systems are based on approximations of the properties of the sound source, the hall, and the listener. The extent of some of these approximations will be discussed. It is at the present stage often difficult to quantify or even describe the audibility of some of these approximations.

1) In *fully computed auralization* the sound transmission properties of models of rooms are studied through the use of computer programs that predict the binaural room impulse response (BRIR). The sound characteristics of models can then be listened to, using a convolver/digital filter or by convolution by software in a computer. Presentations are made using binaural or transaural systems.

2) A combination of computer prediction, multiple-channel convolution, and a multiple-loudspeaker array yields *computed multiple-loudspeaker auralization*. In this case the convolution system has to have many digital-to-analog (D/A) channels which replace the individual delay and reverberation units in the basic simulator used with older multiple-loudspeaker array systems.

3) Use of a physical scale model using ultrasonic techniques yields acoustic scale-model auralization. In traditional scale-model work the convolution are carried out in "real time." This was achieved by playing frequency-scaled audio signals in the scale model using tape recordings or other techniques. The signals are then converted to full scale. This represents *direct acoustic scale-model auralization*.

4) A later, second, approach to acoustic scale-model auralization also uses physical scale models for ultrasonic techniques. Here, however, one does not use direct scaling of the signal as in the previous case, but the scale model is used to measure the binaural impulse response of the hall. This can be done using modern measurement techniques. Convolution is then used as in fully computed auralization. In this case one speaks of *indirect acoustic scale-model auralization*.

3 OVERVIEW OF AURALIZATION SYSTEMS

In this overview most space has been given to fully computed auralization since it is currently the most available system. Direct as well as indirect acoustic scale-model auralization has been described in depth by other authors, as referenced in Sections 3.3 and 3.4.

3.1 Fully Computed Auralization

The trend in auralization today is to use fully computed auralization instead of scale modeling and other methods. The basic layout of such an auralization system is shown in Fig. 1. The system consists of a computer with source, room, and listener data, and a program using a mathematical model of the transmission prop-

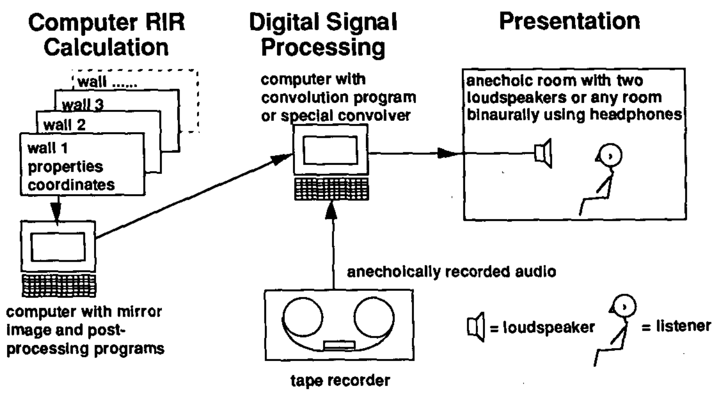

Fig. 1. Basic principles of a system for fully computed auralization.

erties. For fully computed auralization it is necessary to calculate first the RIR and second the BRIR. These are then used to filter an audio signal. This filter process is usually called convolution. The convolved audio signals may then be listened to, for example, using headphones.

Calculating the exact impulse response of the transmission path from source to receiver in a physical situation is simple only for very basic theoretically idealized cases. One such case is the following:

- The source is either an idealized point source or a spherical source translucent for incident sound.
- The source response for the spherical source is given as an even velocity distribution over a spherical surface.
- The reflecting obstacles are planar, having infinite or semiinfinite extension, and having frequency-independent real reflection factors.
- The receiver response is predicted as pressure at a point.

If a more realistic simulation is desired, the model will rapidly become extremely complicated. Radiation properties of sources depend on radiation conditions, which may be altered by sound-reflecting surfaces.

Users of fully computed auralization included in this issue of the *Journal* are Ahnert and Feistel [B22], Dalenbäck, Kleiner, and Svensson [B23], Kuttruff [B24], and Mochimaru [B25].

3.1.1 Overview of Methods for RIR Calculation

A number of methods are available for determining the RIR by computer. For most auditorium acoustics purposes, however, the BRIR may be calculated from the room impulse response by taking the free-field plane-wave-to-ear-drum pressure transfer functions into account. This method provides a poor representation in some cases, as when the incoming wave has a large curvature, for example, close to the sound source or to scattering objects in a hall.

Image-source-model and ray-tracing programs can be used to roughly predict the room impulse response and the current measures of room acoustics. The principles of these methods are outlined in Fig. 2. In contrast to scale-model methods the results obtained using these programs will usually not take wave-related phenomena such as scattering and diffraction into detailed account. These phenomena can, however, be predicted by using other types of computer sound-field prediction methods, such as finite-element method (FEM) or boundary-element method (BEM) programs. The principles of FEM and BEM are outlined in Fig. 3. FEM requires modeling of the entire space, whereas BEM only requires modeling of surfaces. The extremely large number of elements needed for an accurate wide-band model make these approaches impracticable except for cases of small rooms and low frequencies. The BEM calculations particularly are very time consuming.

When using FEM or BEM, the results are initially obtained as complex transfer functions in the frequency domain. These data can, if needed, be transformed into impulse response data using the inverse Fourier transform. The main reason to transform the data into the time domain is to obtain a better feel for the data and to be able to use convolution and evaluation software, which are based on time-domain representation of the room response.

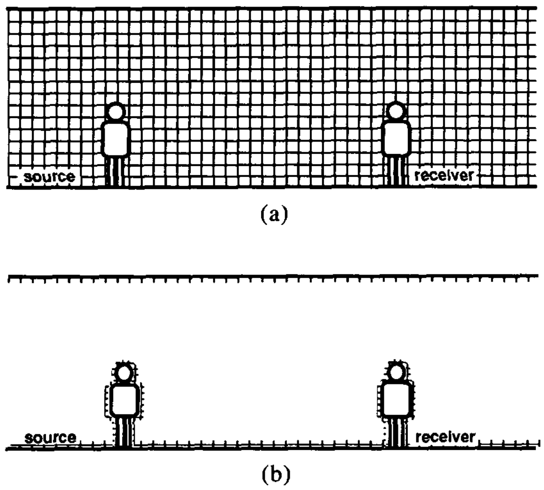

Fig. 3. Finite-element modeling (a) requires generation of a mesh covering the whole room. Boundary element modeling (b) requires only a mesh covering the surfaces of the room.

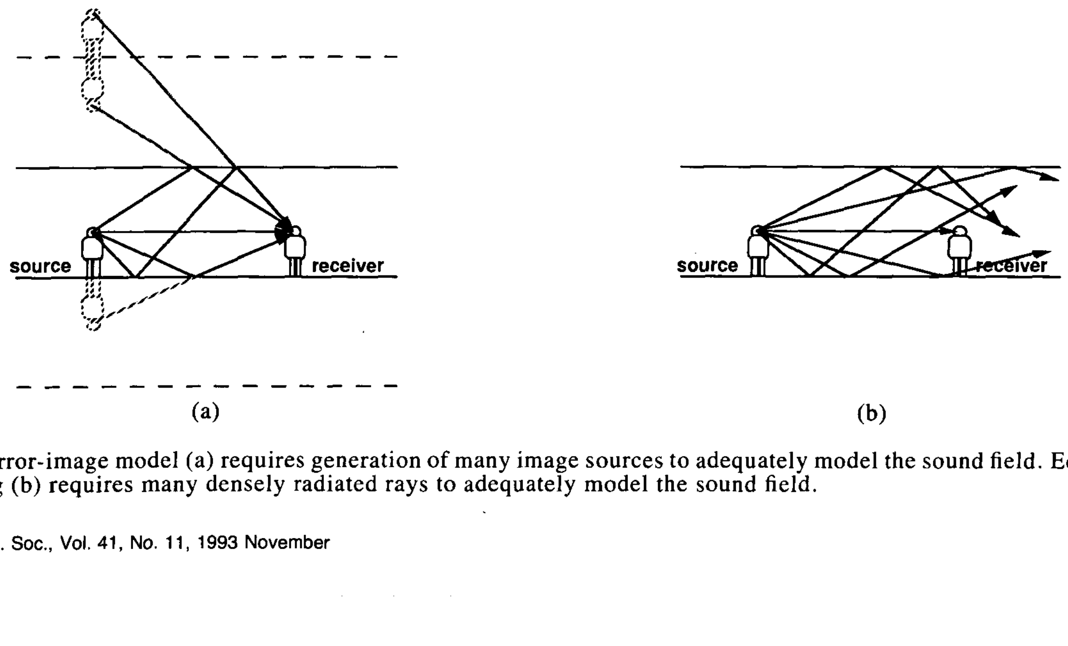

Fig. 2. Mirror-image model (a) requires generation of many image sources to adequately model the sound field. Equivalently ray tracing (b) requires many densely radiated rays to adequately model the sound field.

The RIR may be determined in a number of ways. The most common ones are the following:

- Ray-tracing methods:
  - Pure ray tracing
  - Cone tracing
  - Fully discrete ray tracing
- Mirror-image models:
  - Complete mirror-image base
  - Reduced mirror-image base
- Hybrid methods:
  - Mirror-image and ray-tracing combinations

More involved methods are

- Finite-element methods
- Boundary-element methods
- Finite ray integration

The ray-tracing and mirror-image models usually allow taking the frequency-dependent characteristics of reflective materials and objects into account, usually on an octave or one-third-octave basis.

3.1.2 Influence of Absorption, Scattering, and Diffraction

It is usually rather difficult to calculate the influence of real surfaces and objects on sound propagation. Computer prediction programs based on pure geometric acoustics cannot do this. The nature of sound reflection over a surface of finite impedance and finite extension is complex, even if the surface is planar. Surfaces such as those shown in Fig. 4 are very hard to model.

Thomasson has investigated the absorption properties of planar surfaces with limited areas of sound-absorbing materials [D8]. The results show that the sound absorption of a surface also depends on the ratio of the surface dimensions to the wavelength of the sound. Patches of sound-absorbing materials will give higher absorption than continuous large surfaces. This effect should be fairly easy to take into account for the late RIR.

For first- and probably also second-order reflections the complex nature of sound reflection has to be considered, particularly when the sound is reflected at high angles of incidence. For most locally reacting materials the reflected sound will then be subject to phase reversal and almost no reduction in magnitude. The type of interference effects described as seat-dip effects will result (see, for example, Schultz and Watters [D1]). If these factors are not taken into account, auralization is bound to overestimate the ratio between direct and reverberant sound. Since a large portion of the early sound reaching listeners on the main floor of an auditorium will have propagated at grazing incidence over the audience, this effect should need to be taken into account for auralization work.

Maekawa [D6] and Sakurai and Maekawa [D4] have investigated the reflection characteristics of some reflective surfaces and assemblies of reflective panels. The results of the subjective evaluations showed that listeners could distinguish quite well between the transfer functions of different angles of incidence produced by different types of reflective surfaces. These results indicate that the problem of sound reflection for early reflected sound needs to be addressed more thoroughly, even for angles of incidence smaller than glancing incidence.

The scattering and diffraction of sound by objects or discontinuities close to the listener (such as the audience) are another obvious simulation problem. Kunstmann made model tests which showed the considerable influence of these effects on the propagation of sound over a modeled audience surface for frequencies over 1 kHz [D3].

The sound-field simulation experiments made by Kleiner and Kihlman were designed to investigate the audibility of an added reflection to a binaural reproduction of a natural sound field [D7]. The results indicated that the effects are quite audible even at very low relative levels in a complex sound field and that such scattering and diffraction effects may have a considerable effect on the way we perceive sound quality in auditoriums. These "close to the source and receiver" scattering and diffraction effects are hard to simulate in computer prediction programs based on geometric acoustics. One way, of course, is to calculate the complex pressure and velocity response over the surface of the scatterers and to take these into account via reradiation. Another, much more approximate, way is to use libraries of scatter data for various surfaces. The latter method is not likely to give correct binaural data for the auralization of large diffusing surfaces or diffusing surfaces close to the source or listener.

Room absorption can be taken into account in a simplified manner for the late RIR due to the large number

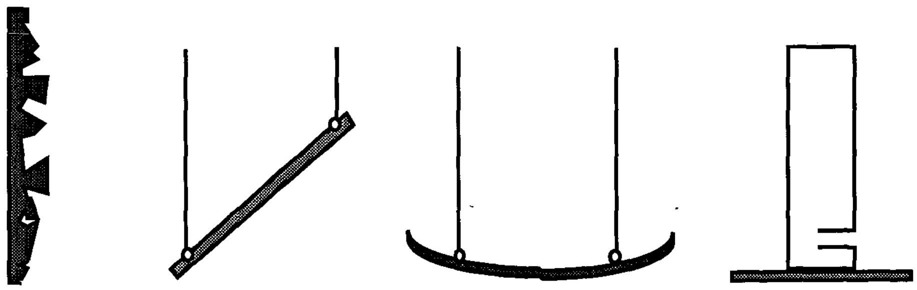

Fig. 4. Building elements such as these are difficult to represent in a geometrical acoustics based model since they are respectively diffusing, limited in size, curved, and resonant.

of reflections and their distribution over many angles of incidence. The exact phase response is also of little interest for this part of the RIR.

3.1.3 Binaural Room Impulse Response Calculation

The BRIR can be considered as the signature of the room response for a particular sound source and human receiver. The response will vary according to source and receiver properties. An approximation of the true BRIR can be obtained by assuming certain properties of the source and receiver and by using an RIR prediction program. Several methods for obtaining the RIR have been mentioned in the previous section. By using a postprocessing program it is possible to transform those data into equivalent BRIR data, which can be used for convolution.

The postprocessing used to obtain a binaural effect or representation may be of increasing complexity such as:

- Two-channel stereo representation of the free field to pick up point pressure transfer functions, receiving points at approximately interaural distance. Various pick-up patterns are possible.
- Semibinaural representation by calculation of the free-field-to-surface pressure functions for a sphere.
- Semibinaural representation by calculation of the free-field-to-surface pressure functions according to Genuit [I8], [I9].
- Binaural representation by measurement of the free-field-to-ear (drum) pressure transfer functions for a particular artificial listening head.
- Binaural representation by measurement of the free-field-to-ear (drum) pressure transfer functions for a particular listener's head.

A number of problems particular to binaural sound reproduction, well known from other experiments, are:

- In-head localization
- Back–front ambiguity
- Lack of head tracking.

The first two problems are probably due to a lack of similarity between the transfer functions used for BRIR calculations and those particular to the listener. Head tracking reduces these side effects of the reproduction process. The head tracking system in the Convolvotron, used by Wenzel, Foster, and Wightman, gives a considerable improvement in realism [B6].

3.1.4 Transaural Presentation

The pressure functions of a binaural signal can be generated in a stereo reproduction in an anechoic chamber by using crosstalk cancellation filters inserted before the stereo loudspeakers, as shown in Fig. 5. Damaske devised an analog hardware filter for this purpose [F1]. The process is now available in various digital signal-processing hardware implementations such as those described by Griesinger [F6]. It is important to realize that also the transfer functions needed for the crosstalk cancellation process are individual, as are the free-field-to-binaural transfer functions discussed earlier. With most implementations the main problem is the back–front confusion. New solutions must be found to eliminate this problem, although listener-specific free-field-to-ear-related transfer functions seem to eliminate much of this problem.

Recently much work has been devoted to the improvement of these transaural techniques, and examples are given by Cooper and Bauck [F5], Miyoshi and Koizumi [F11], and Uto et al. [F12]. A good compilation of binaural and transaural techniques is given by Møller [F14].

3.1.5 Convolution

Convolution may be performed directly in the computer in the time domain. This is, however, a slow process unless special computer architecture is used. Convolution carried out on general-purpose computers is usually in the form of its frequency domain equivalent since fast Fourier transformations of the audio signal and impulse response, followed by their multiplication and inverse fast Fourier transformation of the result, are faster than direct convolution. This method can be implemented with software or hardware. Convolution using this approach is often performed using a computer coupled to an array processor. The advantage of this system is that input signals and room impulse responses may be arbitrarily long, limited only by computer hard disk space. However, a disadvantage of the system is the comparatively long processing time if the impulse response is long.

The equivalent process may be implemented by a dedicated signal processor, for example, the Lake FDP 1 Plus digital filter, which can convolve a two-channel

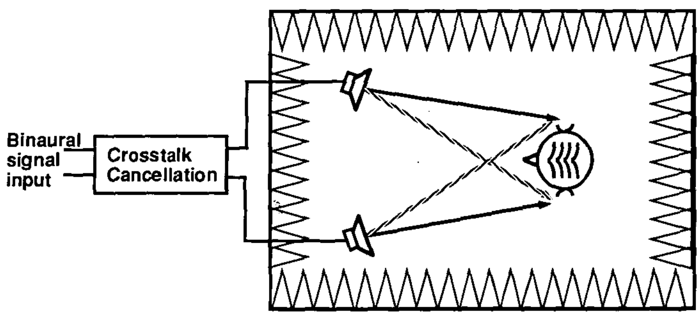

Fig. 5. Transaural playback of binaural recordings requires cancelation of interaural crosstalk by using a filter before the loudspeakers.

impulse response of a maximum length of 2.1 s at a 48-kHz sampling frequency in real time (but with a few seconds delay) [H5], [H9]. Another alternative is using the DSP chips found in the commercial room simulators for hi-fi and studio use. Yamaha has developed such a real-time eight-channel convolver which features 8 s of impulse response on each channel at a sampling frequency of 48 kHz.

3.2 Computed Multiple-Loudspeaker Auralization

Computed multiple-loudspeaker auralization shares all principles with those of fully computed auralization except for the mode of presentation as discussed in Sections 3.1.1, 3.1.2, and 3.1.5. The principles of this system are shown in Fig. 6. The largest advantage of computed multiple-loudspeaker auralization over fully computed auralization, which uses the binaural or transaural technique, is the natural directionality of the sound field in the loudspeaker array. Sounds coming from behind really do come from behind, and one does not have to rely on calculated or measured transfer functions of the ear for the sound to be localized correctly. The largest disadvantage, on the other hand, is the need for the array to be located in an anechoic chamber. With active techniques it might be possible to reduce the influence of room reverberation to allow the array to be used in rooms with some reverberation.

The loudspeakers used in the array must be small and full range. Large loudspeakers will give rise to scattering, which will contaminate the calculated RIR and create a false impression of diffuseness. A computer-controlled mixer and filter bank are also necessary unless the computer uses a multichannel digital analog converter. The multichannel RIRs still have to be computed using the methods outlined in the previous section. Often dummy-head recording of the sound fields is convenient for comparison purposes, and the transaural technique described later may apply.

In this issue of the *Journal* simplified computed multiple-loudspeaker auralization is discussed by Hidaka [A11] and Korenaga and Ando [A12]. Such simplified systems are typically realized as shown in Fig. 7.

3.3 Direct Acoustic Scale-Model Auralization

Auralization by physical scaling uses three-dimensional ultrasonic acoustic models. Using a scaled down model of a hall, the acoustical conditions of the full-size hall may be investigated. The principles of such a system are shown in Fig. 8. The advantages of acoustic scale modeling over computer modeling is the convenience of having nature include all the complicated phenomena such as scattering and diffraction into the model in a correct way instead or relying on more or less adequate mathematical approximations of those phenomena. In the acoustic model, for example, higher order effects such as multiple scattering are taken into account, which are difficult to include in the mathe-

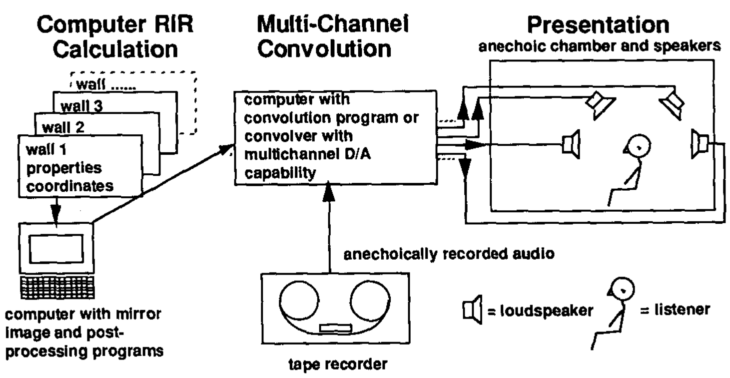

Fig. 6. Basic principles of a system for computed multispeaker auralization.

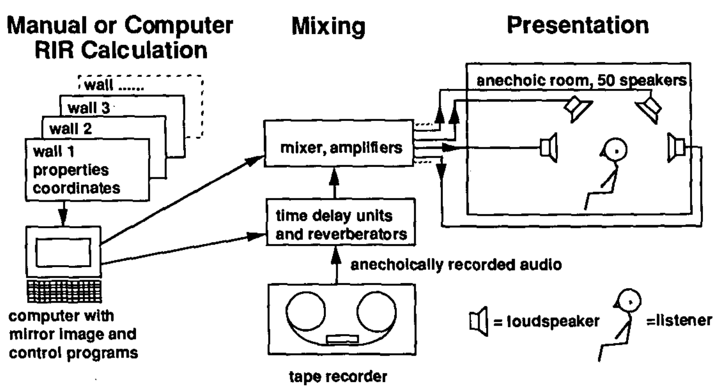

Fig. 7. Basic principles of a system for simplified computer multispeaker auralization using time delay units, reverberators, and mixer instead of detailed convolution of the audio signals.

matical model.

All of this is strictly true only if the acoustic model is an exact model of reality, such as a hall, loudspeaker system, orchestra, microphone, or artificial listening head. These conditions are impossible to achieve in practice, so ultrasonic scale modeling and auralization are an approximation, but of a different kind than computer modeling. Problems arise when one wants to specify particular loudspeaker characteristics and arrays or when one wants artificial listening-head listening capabilities or omnidirectional microphones. It is not feasible at this time to construct, for example, 1:10 or smaller models of a ¼-in (6.35-mm) microphone or an artificial listening head with detailed features such as in a full-size high-quality artificial listening head. Scaled down model artificial listening heads are only available commercially from Japan at this time [H10].

One must also not forget that in order to obtain correct reflection properties a sizable library of scaled model absorbers may be needed. These are not available commercially, so some work has to be devoted to measuring their properties. On the other hand the physical effects of small patches or scattering will be included in the sound field. In order to obtain correct reverberation times it is also necessary to correct for the differing properties of the sound absorption of air at natural and scaled frequencies. It is possible to compensate for the (otherwise) excess sound absorption of air in the ultrasonic scale model by using dry air or by using nitrogen instead of air.

The similarity to which it is necessary to work is not particularly well investigated and not much has been published on this subject. Kleiner, Orlowski, and Kirszenstein [I16] investigated the agreement between an ultrasonic scale model and a computer model using conventional acoustic measures as criteria and found reasonably good agreement. This does not imply, however, the auralization results will agree. At this time direct acoustic scale-model auralization is favored in Japan. It is described by Oguchi et al. [C5] in a paper included in this issue. Several other consultants in Japan use this method.

3.4 Indirect Acoustic Scale-Model Auralization

A later development in acoustic scale modeling uses a digital signal acquisition unit to record the binaural impulse responses of the loudspeaker, room, and artificial listening-head combination. These binaural impulse responses can then be convolved with audio in the same way as in fully computed auralization. This method is outlined in Fig. 9. In all other aspects it is equivalent to direct acoustic scale-model auralization.

Indirect acoustic scale-model auralization has been used, for example, by Xiang and Blauert [C3], [C4] to study the acoustical conditions of rooms and by Kleiner et al. [D10] to study the properties of various diffusor arrangements. The advantages of indirect over direct acoustic scale-model auralization are that the signal-to-noise ratio may be improved through averaging, linear as well as nonlinear distorsion may be reduced, and the storage of impulse responses is much more convenient than the storage of models. In this

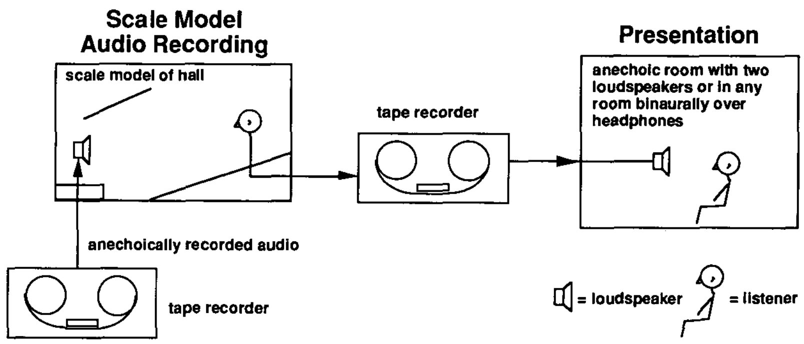

Fig. 8. Basic principles of a traditional system for direct acoustic scale model auralization.

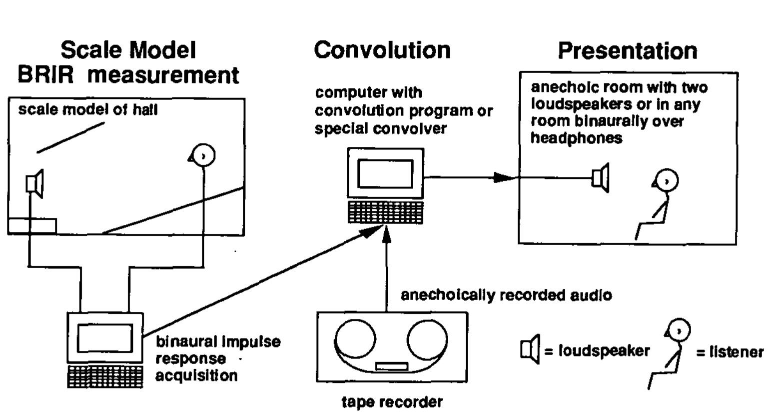

Fig. 9. Basic principles of a system for indirect acoustic scale model auralization.

case mathematical compensation of binaural impulse responses due to air losses may be made. In this issue indirect acoustic scale modeling is discussed by Polack, Meynial, and Grillon [C6].

4 AUDIO SOURCE MATERIAL FOR AURALIZATION

The problem of sound-source characterization is common to all auralization systems. The auralization of loudspeaker installations and the auralization of auditorium acoustics are similar, yet in a sense they represent different problems. An advantage to the auralization of loudspeakers is that sources are dealt with which, if one disregards level-dependent nonlinear distortion and compression, are time invariant and may be reasonably well specified in terms of directivity and frequency response, since they are small objects. Auralization of loudspeaker clusters can be done with multitrack technology or by digital sound editing technology using additions.

The adequate simulation of natural source test signals is one of the largest problems confronting the auralization of auditorium acoustics. The characteristics of sound sources (such as humans and musical instruments) are difficult to describe in physical terms, not only because of the acoustic complexity of the sources, but also because of the time variance of these sources and the differences between single sources and groups of sources. In analogy with the problems discussed later in conjunction with the listener environment, the diffraction and scattering of the players in a group are considerable and vary between performances.

Obtaining both an adequate number of channels and a correct representation of the source directivities seems a formidable task. It is useful at this stage to compare the efforts that go into auralization with those of recreating the impression of an orchestra on stage. It would seem doubtful that a large orchestra could be used in a computer auralization if orchestra sound cannot even be simulated by loudspeakers on stage in the hall. Reichardt and Kussev [I5] investigated sound-level balance in a concert hall by trying to recreate an orchestra on stage using a number of loudspeakers. With quite elaborate arrangements they were able to recreate the balance of an orchestra to the threshold of perceptibility. This is an indication that the problems of source characterization in fully computed auralization may be overcome.

For auralization of small ensembles a pair of simulated sources is probably sufficient, and for larger orchestras three or more sources have to be simulated. Another possibility is to use measured directivities for musical instruments.

The availability of suitable source material is poor today. Only a few standard mono and stereo recordings are available, notably those from the Building Research Station, Denon, Yamaha, and the Archimedes Project recordings [H1], [H3], [H8]. These recordings offer a limited range of speech and music material. The usability as source material in auralization is open to debate. This applies especially to those recordings that include floor reflections.

Mixdowns from multichannel microphoned recordings may also be useful for simplified auralization work since they may fill in "insufficiently dense" computer-calculated RIR functions where scattering is often neglected.

5 FUTURE ISSUES IN AURALIZATION

5.1 Auralization Verification

The first question to ask with regard to the verification of an auralization process is: "Are we interested in a general aural simulation which would be suitable, for example, for educational usage, for in comparisons, or are we interested in absolute accuracy of sound reproduction?"

The least one can expect from any auralization system is an ability to render the differences between various halls or loudspeaker installations in a subjectively convincing manner, as reported by Dalenbäck, Kleiner, and Svensson [B23].

The objective of the auralization experiments reported by Kleiner in 1980 was limited to a comparison of the speech intelligibility results for a real hall with those of the auralized hall [A3]. The test was simple, and the results would probably not have been as good if sound quality or timbre had been compared. Dalenbäck [B20] has published comparisons which indicate that good agreement can be obtained for RIR predictions under conditions where geometric acoustics apply.

Since the correct reproduction of the sound of real musical instruments or soloists is at present so far from being realized, verification must begin with a comparison between binaural recordings of sound in halls where music or other signals have been radiated by loudspeakers on the one hand, and the auralization of the same setups on the other. If these results are satisfactory for a large number of source and receiver position combinations, then the auralization process itself, that is, the RIR prediction and BRIR synthesis routines as well as the presentation equipment, must be considered satisfactory, and the remaining problem is the simulation of the sources.

In our experience speech is often a good test signal. It is well defined and people seem to relate easily to its acoustic quality, taking less notice of the way it is spoken or of its information content. One drawback, of course, is that people are so used to listening to speech in low-reverberation-time surroundings that speech is a poor test signal for live halls. Solo instruments are also useful. It is reasonably easy to obtain anechoic recordings of solo instruments, and often such recordings are not musically distracting. Chamber and orchestra music is, of course, necessary to judge the quality of larger halls. Noise, transients, or noise bursts serve well for the study of specialized phenomena such as timbre or echoes. These signals are comparatively easy to transmit, even in the real hall, and are well defined.

Another way of comparing the real and the auralized sound fields is to compare the just noticeable differences for various parts of the binaural impulse response. This can be done by a BRIR cut and paste technique. By switching between real and auralized parts of various lengths and positions within the BRIR it would be fairly easy to identify the parts of critical nature, as discussed by Kleiner, Svensson, and Dalenbäck [B9].

When comparing auralized auditoriums with real auditoriums, it is necessary not only to achieve the correct spectral balance and level, but also to include the noise that is always present in a real hall. If realism is to be achieved to the auralization of auditoriums, then it is necessary to add background noise of sufficiently random character with level and quality similar to those of the real hall. A more difficult problem in hall auralization is the addition of audience-generated noise in a subjectively convincing manner. Kleiner has outlined a method for the estimation of this noise [I7].

5.2 RIR and BRIR Calculation

Improving the speed and performance of computer RIR and BRIR programs is of great importance if auralization is to move from laboratory to practical applications as a tool in industry and consultancy. The programs can be improved by using combinations of ray-tracing and mirror-image prediction schemes. Coupling to CAD databases will probably also make data entry faster and more efficient. Faster computers will enable the use of more advanced prediction models than the mirror-image and ray-tracing models used on PCs now.

Binaural response can be obtained more rapidly if simplified methods using sphere approximations instead of true head-to-ear transfer functions are used. This may be a workable approximation of reflections of higher orders.

Better methods of taking absorption, scattering, and diffraction into account are needed. The simulation of curved surfaces, which is now made by piecewise linear approximation, is unsatisfactory.

A particular problem appears in the simulation of the late part of the RIR. If one generates a simulated reverberation space, which is easy to calculate, it is also necessary to provide the spatial attributes of the original reverberant sound field for this simulated space. Even if a reduced set of data is used, accounting for the free-field-to-binaural pressure transfer characteristics of each reflection is a time-consuming process. For less demanding applications involving only a spacious feeling of reverberation an exponentially faded noise signal can be used to generate the late part of the BRIR, as used by Martin [B17].

5.3 Computer Program Standardization Interfaces

To compare different approaches to the problems and possibilities of auralization, the capability to interchange files between programs and hardware would be useful. A consultant could then purchase an auralization facility separately from a supplier of RIR software.

A standardized interchange file format between various RIR and BRIR prediction programs would make it possible to select and use the program one prefers for RIR or BRIR calculation. Such an interface has been suggested by Møller [I14]. Although it is probably not possible for the most advanced BRIR calculation to separate RIR prediction and BRIR synthesis, it would be useful to have such a format for ordinary RIR programs using ray-tracing or image-source models.

To calculate the BRIR from the output data of a regular RIR program, the minimum of information required is 1) the calibrated band level of each reflection as a function of frequency, 2) the incidence angles of each reflection, and 3) the data for late reverberation generation as a function of frequency.

The RIR-to-BRIR postprocessing programmer would then have access to enough data to introduce the free-field-to-ear transfer functions as well as to take the frequency response of the reflection processes into account.

The BRIR output file would then have to be given in a time or frequency domain format suitable for convolution. Here it would also be useful to have a common interchange file format specifying at least 1) the sampling frequency, 2) headphone or loudspeaker equalization, and 3) the type of transaural system.

Gotoh et al. suggested a scheme for binaural recording normalization which would, of course, be of great value if successfully developed [F4].

5.4 Binaural or Transaural versus Multichannel

The experience of binaural headphone listening to artificial listening-head recordings is often quite disappointing due to in-head localization and front–back ambiguity. However, in our experience the introduction of a transaural system will improve localization tremendously, even with nonindividualized transfer functions, since the transaural systems almost perfectly prevent the formation of an in-head localization experience. The in-head localization, in the authors' opinion, must be rated as very severe spatial distortion in questions of auditorium acoustics. Researchers generally consider the lack of adequate free-field-to-ear transfer functions as the primary reason for the observed problems of front–back ambiguity in binaural room simulation. This thesis was tested by, among others, Morimoto and Ando [F3] and Wightman and Kistler [F8]. They found much better localization when individual free-field-to-ear transfer functions were used.

Even as early as 1975, experimental head-tracking systems were available to improve the out-of-head localization of binaural systems. Head tracking of all information available in the simulated BRIR is probably unnecessary if the aim is only to improve out-of-head localization and front–back discrimination. The Convolvotron system, mentioned earlier, uses head tracking to improve sound-source localization.

Unless progress is made in the binaural reproduction techniques, it is not unlikely that the multichannel simulator, particularly in view of the rapidly decreasing cost of computer and audio hardware, will return to favor as the highest quality auralizer. It will also allow for head movements and make individually adapted free-field-to-ear transfer functions unnecessary.

5.5 Data Reduction

It is likely that it is possible to reduce the amount of information needed to create a particular "audible environment," that is, the aural impression of a particular sound field. Masking in both the time and space domains can probably be used in order to reduce the required information. Experiments by Iida and Morimoto substantiate this [F13]. The results may be used to simplify the BRIR patterns in order to lower the technical demands on the BRIR synthesis process.

At least for the late (reverberant) part of the RIR or BRIR it is possible for a simplified process to produce impressions similar to those created by a more exact process. Quite natural sounding reverberation can be obtained with a large reduction of computing effort, as many commercial reverberation units have shown.

In the case of auralization though, such simplified processes are not sufficient by themselves, since the angular distribution of the reverberation of the actual hall must also be considered. It would probably be possible to spatially sectorize the reverberant part of the RIR or BRIR, and to calculate a simplified process for each of these sectors. Wagener and Damaske have investigated the audibility of different reductions of the reverberant sound fields [I2], [I3]. Damaske showed that a minimum of five sectors in the lateral plane where necessary to obtain a sound field which had sufficiently subjectively diffuse properties. If at least one more sector is added to account for sound coming in from above, a minimum of six spatial sectors is needed. It is not clear, however, whether six sectors are adequate to simulate all conceivable halls.

5.6 Combination with Other Techniques

An interesting possibility exists in using auralization in conjunction with active sound-field control. Presentation of auralization in the office instead of in an anechoic chamber will probably necessitate the reduction of ambient noise most of the time. The reduction would be necessary not only because the noise has a masking effect, but also because, if signal levels are raised to compensate for ambient noise, the spectral content and the auditory feeling of spaciousness will be altered. Since essentially the same types of hardware and software are used, these techniques could posibly be merged successfully. The noise reduction available by active noise control is often in the range of 20 dB at low frequencies, which may be sufficient in office surroundings where low-frequency noise is predominant. It is also conceivable that adaptive filtering could be used to control unwanted reflections from the listening room when transaural listening is used.

5.7 Other Uses of Auralization

There are a number of interesting alternative uses for auralization, or contexts where the knowledge gained from auralization or its technology could be used. Apart from the uses mentioned previously, other potential uses include the following:

- Training of architects, acousticians, and audio professionals
- Training of musicians
- Training of blind people
- Factory noise prediction
- Noise quality assessment
- Studies in psychoacoustics
- Studies of the room–loudspeaker interface
- Studies of reverberation-enhancement systems
- Investigation of (stereo) microphone pick-up patterns and placement properties
- Increasing realism of virtual reality systems
- Effects used in video games
- Improvement of binaural sound for in-flight entertainment
- Improvement of automotive stereo reproduction
- Presentation of non-sound data, such as radar information, to pilots and other (military) personnel where visual information systems are already fully utilized.

6 CONCLUSION

Auralization is becoming a useful tool thanks to the advent of powerful personal computers and software. Together with new hardware implementations of digital signal processing, this forms the basis of a powerful new technology for room simulation and aural event generation.

Convincing demonstrations of the qualities of auralization have been made, yet the verification of the technology's ability to accurately reproduce the aural effect is lacking. This limits the usage of auralization as a design tool. It is important at this stage to emphasize that few documentations exist in research publications of even the acoustic measures predicted by the computer room impulse response prediction programs.

The combination of auralization with transaural reproduction and active sound-field control should make it possible to use auralization outside the laboratory with techniques other than headphone reproduction. A large number of new and interesting applications are conceivable, the most interesting of which appear to be in various training and sales situations and in virtual reality systems.

7 REFERENCES

This is a structured list of references for those interested in further study of auralization and its scientific background. Often several subjects are included in a paper, so compromises have been necessary to the ordering of papers under various subject headings. The year of publication and alphabetical arrangement of authors' names have been chosen to order each subgroup. Not all references have been used in the text. Dissertations, articles published in scientific journals, and papers given at international scientific conferences have been included. The list is not in any way complete and is intended as an introductory guide only. Many more references will be found in the references cited. References appear roughly in chronological order.

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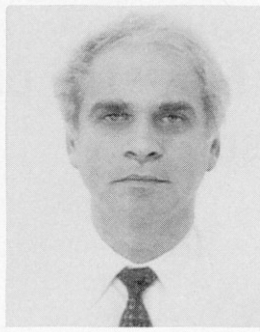
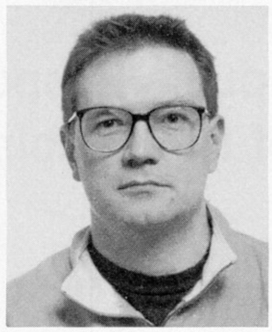


M. Kleiner



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