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FREE FIELD SIMULATION OF LOUDSPEAKER/ROOM ACOUSTICS

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1. INTRODUCTION

Archimedes is a cooperative psychoacoustics research project funded under the European EUREKA scheme. The three partners are The Acoustics Laboratory of the Technical University of Denmark, Bang & Olufsen of Denmark, and KEF Electronics Limited of England. The primary objective of the project is to quantify the subjective influence of listening room acoustics and loudspeaker directivity on reproduced sound. It is hoped that the results can be used to develop improved systems for sound reproduction.

A study of this nature, relying on the responses of human subjects, becomes impossibly complicated and difficult if carried out with a variety of real loudspeakers and rooms. Control of the experimental variables is much simpler if the stimuli are simulated under controlled conditions.

Simulation of acoustic stimuli can be achieved using either headphones or loudspeakers, though both present practical difficulties. At the present stage of development, headphone simulation suffers from problems of confused localization. Simulation via loudspeakers requires anechoic surroundings and a large number of loudspeakers, amplifiers and signal processors, but is potentially more natural for the test subjects.

The approach chosen for this project is simulation via loudspeakers in an anechoic chamber. The anechoic chamber, located at the Acoustics Laboratory in Lyngby, Denmark, has working dimensions (between the wedge tips) of 8 m (26 ft) high, 10 m (33 ft) wide, and 12 m (39 ft) long.

The simulation technique, which is based on the theory of images, requires a number of well-matched wideband loudspeakers, driven from suitably generated signals, to represent the "images" due to sound reflection within the room. Fig 1 illustrates in a two-dimensional plan view the position of image sources resulting from reflections within a rectangular room. For simplicity, only the listener and a single loudspeaker, corresponding to the right channel of a stereo pair, are shown.

The first part of the Archimedes project is devoted to investigation of timbral effects. The simulation is therefore limited to the representation of a single loudspeaker reproducing a monophonic signal.

2. PRACTICAL CONSIDERATIONS FOR THE SIMULATION SETUP

Even with the very generous capacity of the Lyngby chamber, it is not possible to locate all the psychoacoustically important image sources in their natural geometric positions. Instead, all loudspeakers are located, with correct azimuth and elevation, on the surface of an imaginary sphere of 3 m radius centered on the listening position. Attenuation and delay of the loudspeaker inputs provide for correct level and timing of the received image signals at the listening position.

The supporting structure for the loudspeakers is made as light and non-reflective as possible to minimise disturbance to the simulated sound field. Portions of it are treated with sound absorbing material to reduce the residual reflections below the level at which they might affect the test results.

The listener's ears must be located close to the specified listening position. This position is for practical reasons about 2.5 metres above the wire-netting floor of the chamber. Thus the listener is seated in an elevated "Danish bus driver's seat" mounted on a structure supported rigidly from the true structural floor of the chamber.

The exact chair position is adjustable remotely via motors so that the listener's ears can be placed in the correct position as observed via a fixed video camera. This camera may also be used for monitoring and/or recording the listener's movements. A curtain prevents the listener from seeing the simulation setup, except for a single red LED which defines the "front" angular reference.

3. THE LOUDSPEAKERS AND AMPLIFIERS [1]

Because everything placed within an anechoic chamber tends to disturb the sound field and cause unwanted additional reflections, the loudspeakers must contribute minimal diffraction and reflection. A small, spherical enclosure is best suited to meet these requirements.

The enclosure chosen is a sphere of 280 mm (11 inch) diameter. It is manufactured in large quantities in Denmark, as two mating hemispheres, for use as a float for trawler nets. It comes in one colour: bright orange!

The spherical enclosure is fitted with a 110-mm (4½-inch) coincident-source drive unit specially designed by KEF for the project. This is the largest unit that can be mounted within the spherical enclosure without severely truncating the spherical profile.

The enclosure front hemispheres were modified to take the new drive unit using numerically controlled machine tools at Bang & Olufsen. In addition, a matching front ring was machined from a similar plastic material so that the spherical contour is smoothly blended into the diaphragm profile of the bass/midrange unit. This ring also supports a front cloth dust protector.

The rear hemispheres are similarly machined to take an electrical input connector and a vibration isolation mount for a support rod extending back from the rear of the drive unit. The hemispheres are joined by a permanent adhesive to form the complete enclosure.

Near the rear of the support rod, a counterweight is mounted so as to place the centre of gravity of the complete system at the centre of the sphere. This is required by the simple supporting method used to ensure that all loudspeakers hang exactly in the calculated positions, independently of the direction they are aimed.

The interior of the enclosure is filled with absorbing foam to damp standing waves and to provide a known amount of acoustic damping to the LF unit. The latter is useful because the loudspeaker systems are driven from current-source, rather than voltage-source, amplifiers.

A simplified cross-sectional drawing of the loudspeaker system is shown in Fig 2.

Each loudspeaker system is driven by two current-source amplifiers. The choice of current-source drive was made for several reasons.

Firstly, the amplifiers must be located on the roof of the chamber, and thus the cables between the amplifiers and loudspeakers are necessarily quite long: 15 m (50 ft). They cannot be made very heavy, so their impedance is not negligible. Current drive makes the system performance independent of cable impedance.

Secondly, the use of current drive eliminates the problem of thermal compression in electrodynamic drive units with copper voice coils. The force on the diaphragm is proportional to current, and this is maintained in a current-drive system independently of voice-coil resistance changes due to heating. Thus compression and change of spectral balance is prevented.

Thirdly, current drive reduces mid-frequency distortion from the bass/midrange drive unit by a significant amount.

The amplifiers used are modified Bang & Olufsen Beolab Penta 150 amplifiers. These have a power rating of 150 watts into 8 ohms. With the modification for current-source operation, the output impedance is raised to about 1000 ohms, amply high for use with drive units of 4 to 8 ohm nominal impedance.

These amplifiers were chosen not only for their availability to the project, but also for their low noise level, which becomes very important when a large number of loudspeakers are used for listening tests in an environment of very low inherent acoustic background noise.

Each bi-amplified loudspeaker system also incorporates a low-level active crossover network which includes individual drive-unit equalisation and thermal-damage protection.

Fig 3 shows in block circuit form the structure of each complete "active" loudspeaker system.

4. RECORDING OF PROGRAMME MATERIAL [2]

For psychoacoustic testing of reproduction timbre, test signals of natural sounds such as speech and acoustic musical instruments are desired, in addition to electronically generated signals such as pulses and noise. Because psychoacoustic testing involves many repetitions of the same test signal, to different listeners under different simulated conditions, repeatability is of paramount importance; signal storage must therefore be digital.

It was soon discovered that suitable monophonic digital recordings, made under controlled and documented conditions, were not available. A recording programme was undertaken to obtain the desired material, recorded under both anechoic and semi-reverberant conditions. These recordings were made by Bang & Olufsen and KEF, using selected calibrated microphones, on digital media including DAT, PCM on videotape, and directly to large capacity hard disks. Recordings selected for use were then stored on hard disk for direct playback in the simulation setup.

5. SELECTION AND TRAINING OF TEST SUBJECTS [3]

For the results of listening tests to be treated as useful scientific results, they must be reproducible, reported on an appropriate scale, and a significant and meaningful measure of the prespecified aspect of the perceived sound.

The degree to which these criteria can be fulfilled depends not only on the design of the experiment but also on a number of factors relating to the test subjects' skill and consistency. These factors include hearing threshold level, previous listening experience, and ability to perceive differences in sound stimuli.

Subjects used in the Archimedes experiments are Danish students in the age range of 18-28 years. All candidates are screened for normal hearing thresholds. Previous experience listening to live and reproduced music is evaluated by means of a questionnaire. Final choice is based on the results of subjects' performance in trial tests devised to determine their ability to perceive differences and their consistency.

Test results reported in [3] indicate that with training and experience, previously untrained subjects possessing the requisite basic abilities can greatly improve their sensitivity and their consistency. So much so that one suitably trained and experienced subject can provide the same statistical confidence as <u>seven</u> untrained subjects.

6. PSYCHOACOUSTIC CRITERIA FOR SIMULATION [4]

A simple interpretation of the theory of images might suggest that sound field simulation requires a vast number of loudspeakers. Fortunately, a number of published psychoacoustic results indicate that this is not so.

The direct sound from the loudspeaker being simulated must, of course, be accurately modelled. This requires only one loudspeaker and amplifying channel in the simulation setup.

Reflected sound arriving at the listener after the direct sound is at first relatively sparse in time, but the density increases (and the intensity decreases) with time. For a typical listening room, reflections arriving after about 20 to 30 ms are so densely packed that the hearing mechanism does not distinguish them as discrete events, but rather as a general ambience or reverberation.

In the Archimedes project, this reverberant "tail" of sound is modelled separately with artificial reverberation of the required density and decay, using six of the loudspeaker/amplifier channels. By maintaining low correlation between the signals from these six loudspeakers, the reverberation is made subjectively diffuse; ie, movement of the test subject's head produces no change in the perceived sound, such as "swishing" for a noise signal.

Sound from the remaining reflections, or images, arriving in the first 20 to 30 ms, must be simulated with sufficient faithfulness. There are still a large number of images in this category, so the available results of psychoacoustic tests are used to minimize the number of loudspeaker channels required.

Experiments reported in the literature indicate that where only a single reflection is present, this must exceed a certain threshold level to be audible or to affect perceived timbre. There is good reason to believe that where many reflections and a reverberant field are present, the threshold levels for individual reflections will be higher. As a conservative starting point, all early reflections which exceed the threshold for single reflections of the same delay are included in the Archimedes simulation; those falling below this threshold are excluded. As a further precaution, the threshold values used for this exclusion are taken as those reported for experiments using headphones, which are typically 5 dB lower than those for loudspeaker listening.

As may be deduced from Fig 1, many of the image sources are located in nearly the same direction from the listener as one or more others. When moved to the common 3-m radius, intersecting loudspeakers would be required. It is desirable to keep the number of loudspeakers to a minimum, both for simplicity and to minimise extra reflections within the setup.

Fortunately, existing psychoacoustic results indicate that the hearing mechanism is somewhat imprecise in its localization capability. This property is known as "localization blur". For the Archimedes project, it means that a single loudspeaker may be used to reproduce two or three image sources from nearly the same direction by combining the desired signals before the amplifier.

Using the above simplification techniques, the number of loudspeakers required for the simulation of early reflections is of the order of 20 -- a manageable number for the setup described.

The choice of a 3-m distance for the simulation loudspeakers is based on further existing psychoacoustic results. It is the minimum distance for which the hearing mechanism cannot obtain independent clues as to the actual distance of the source. Hence the use of attenuation and delay may successfully be used to simulate a source at a greater distance.

7. TEST AUTOMATION AND CALIBRATION

Psychoacoustic testing may be described as expensive, time consuming and difficult. It is important to ensure that tests runs smoothly and correctly. This is a natural task for computer control and data collection. In the Archimedes project, three computers (IBM PC or compatible types) are used for various functions.

The most important of these is the "Master" PC, which runs the experiments, collects data from a small terminal attached to the test subject's chair, stores test results, and controls via communications interfaces the other two PCs.

In addition, the Master PC carries out calibration duties which are essential to confidence in the test results. With so many loudspeakers, amplifiers and simulation channels, there is the risk that failure of one channel or drive unit might not be noticed but would invalidate the results of a day (or more) of expensive testing.

When a particular loudspeaker/room model is first implemented on the experimental setup, every reproduction channel is checked for correct response to a microphone located at the listening position and the result stored. Then, when the testing starts, the complete system is checked twice a day for any significant deviation from the original responses.

Use MLSSA-card, programmed by [illegible].

8. OBTAINING THE LOUDSPEAKER SIGNALS

Each loudspeaker is carefully positioned according to computations based on the theory of images. Each must be fed with a signal having the correct characteristics for one or more of the reflected-sound paths for the room being simulated. In addition to delay and attenuation due to distance, these characteristics include the frequency response of the off-axis angle of the reflection path from the original loudspeaker and the frequency dependent attenuation of reflections from the room surfaces.

LOUDSPEAKER/ROOM SIMULATION

In general, the off-axis response of a multi-unit loudspeaker is complicated and non-minimum phase, with inter-unit time delay dependent on off-axis angle. The reflection from a room surface is typically a less complicated function of frequency but does depend also on the angle of incidence. For most purposes, this dependence may be modelled by the familiar "cosine law".

It can be seen that frequency response simulation and implementation are far from simple matters and have necessitated the adoption of a large flexible array of digital signal processors, referred to within the project as the "DSP Engine". The flexibility allows both cascade (serial) and summed (parallel) synthesis paths to be constructed and combined.

Control of the DSP Engine is the job of the second of the project PCs, the "DSP Host". This machine calculates off-line and stores the required processor configuration for a given loudspeaker/room model together with the required filter coefficients and attenuation and delay parameters. These are downloaded into the DSP Engine on command from the Master PC. The third PC, the "SDI Host", controls the Winchester hard disk which delivers test stimuli to the DSP Engine when so directed by the Master PC.

A simplified interconnection diagram of the simulation system is shown in Fig 4.

The basic building block of the DSP Engine is the Motorola DSP56001 processor supplied on a PC plug-in card as the Ariel PC-56. The DSP Engine design uses 32 of these, though test simulations carried out so far have not required more than 16 to be used. Associated with each PC-56 card is a purpose designed "Slave" card which provides digital signal multiplexing and two D/A channels with output filters and line amplifiers.

An example of the simulation requirements of the project is the modelling of an ordinary two-way loudspeaker in the listening room of the Acoustics Lab.

The off-axis response of a two-way loudspeaker can be simulated in a relatively simple way by first simulating the minimum-phase on-axis or front hemispherical responses of the individual drive units and then modifying these for each off-axis direction by the difference curves for the individual drive units. These are summed using the correct relative time delay to obtain the desired non-minimum-phase result.

9. REFERENCES

[1] L R Fincham, A Jones and R H Small, "The Influence of Room Acoustics on Reproduced Sound, Part 2: Design of Wideband Coincident-Source Loudspeakers", AES Preprint No 2877, 87th Convention, New York, 1989 October 18-21.

[2] Villy Hansen and Gert Munch, "The Influence of Room Acoustics on Reproduced Sound, Part 3: Making Recordings for Simulation Tests in the Archimedes Project", AES Preprint No 2836, 87th Convention, New York, 1989 October 18-21.

[3] Søren Bech, "The Influence of Room Acoustics on Reproduced Sound, Part 1: Selection and Training of Subjects for Listening Tests", AES Preprint No 2850, 87th Convention, New York, 1989 October 18-21.

[4] Søren Bech, "Electroacoustic Simulation of Listening Room Acoustics: Psychoacoustic Design Criteria", AES Preprint No 2989, 89th Convention, Los Angeles, 1990 September 21-25.

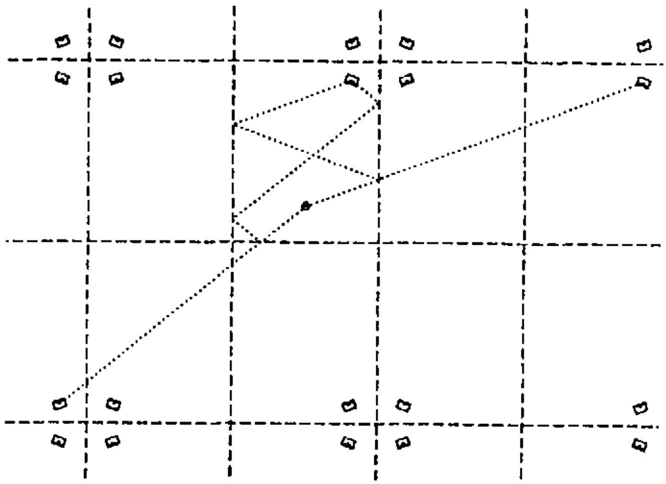

Figure 1. Plan view of the regular array of images produced by reflection in a rectangular room. The dotted lines show how each image corresponds to a particular reflection pattern.

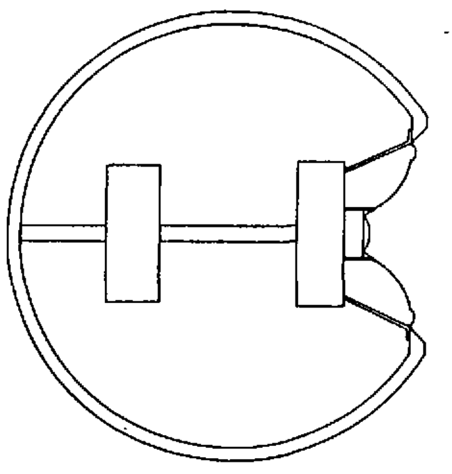

Figure 2. Simplified cross-section of the spherical coincident-source two-way loudspeaker system. A counterweight ensures that the centre of gravity is located at the centre of the sphere.

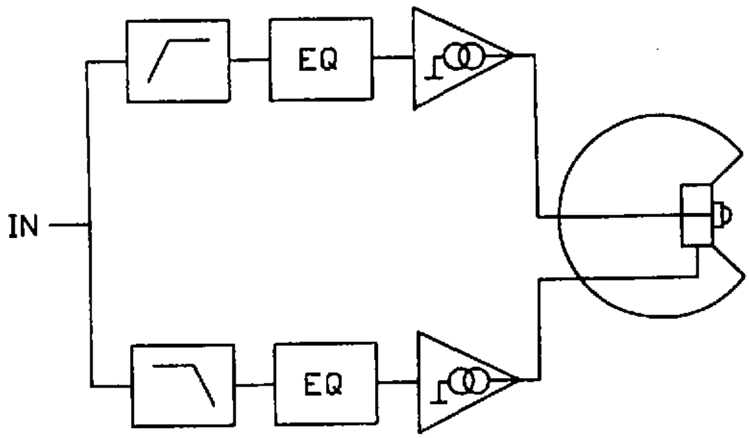

Figure 3. Block circuit illustrative form of the loudspeaker system. The EQ blocks are tailored to the individual LF and HF units of each system. Current-source (or transconductance) amplifiers are used.

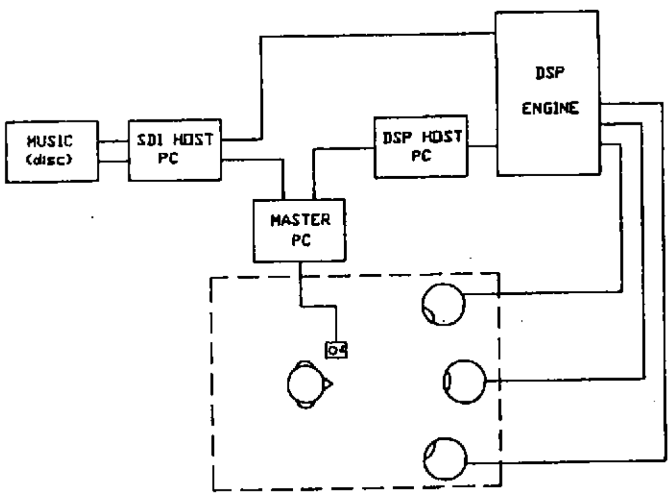

Figure 4. Simplified diagram of the simulation system and its interconnections, showing the three PCs, the hard-disk signal source, and the test subject in the anechoic chamber with the loudspeakers and response terminal.