The Active Listening Room
A Novel Approach to Early Reflection Manipulation in Critical Listening Rooms*

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A novel method of controlling reflections, using flat panel loudspeakers, in a reference listening room is described. Models and implementations are presented in the case of single-transducer monophonic reproduction as well as two-channel and five-channel stereophonic arrangements. The results of a pilot listening test showed that differences in reflection patterns were readily detected by a panel of experienced listeners.

0 INTRODUCTION

Traditionally experiments involving the simulation of sound fields are carried out in anechoic chambers where 95% of the source sound energy is absorbed and which are considered to be inert acoustic environments. A reference listening room on the other hand is designed to resemble a domestic listening room with controlled acoustic characteristics. However, it is a fact that any listening environment, including a reference listening room, has its own acoustic characteristics that, in terms of subjective impressions, make it quite different from any other. Although complying with current standards, the sound field of a reference critical listening room is far from being considered acoustically inert, and it has an influence on the characteristics of reproduced sound. The variations in perceptual and physical effects which can arise in such listening environments are the primary features of the research project described here, known as the active listening room, where key acoustic features (that is, the early and late reflection patterns) can be varied in a very flexible and controlled manner without changing the acoustic fabric of the existing listening environment permanently.

Reflected sounds influence the timbre and spatial character of live and reproduced sounds. Most investigations of reflections have focused on live sounds in large halls. Current interest in the spatial impressions of reproduced sound and the effects of the acoustical interactions of loudspeakers, rooms, and listeners gives rise to a need for further experimentation. Such experiments may be facilitated by means of a flexible and inexpensive sound field simulation technique.

With the standardization of reference listening rooms, such as the ITU-R BS.1116 [1], where the reverberation time and early reflection patterns are controlled within critical limits for a given shape and size of the room, the simulation of artificial sound fields should be possible (at least at middle to high frequencies) by strategically arranging multiple active sources around the listener, without changing the existing acoustic fabric of the room permanently. This will give another dimension to the acoustic properties of critical listening environments, where a listening test can be carried out in a variety of acoustic conditions without relocating the experimental set to different listening environments, saving time and money. The low-frequency characteristics of the room, particularly its standing-wave patterns, remain as dictated by the room's basic structure.

This is in recognition of the fact that there are slight differences in the various standards such as the ITU-R BS.1116, IEC 286-13 [2], EBU R22 [3], OIRT Rec. 86/3 [4], and Nordic Rec. N 12-A [5]. For example, the ideal reverberation time window for a given room volume differs in [1] and [2], and the frequency range and the arrival time of early reflections in [1]–[3] differ significantly from those in [4], [5]. It is hoped that this novel simulation technique will allow the simulation of subtle differences in the acoustic character of various listening conditions in a given listening room.

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A brief review of the relevant literature is followed by a description of various experiments in which acoustical simulations and physical implementations of controlled reflection zones in a standard listening room were evaluated, both through measurement and by means of a brief subjective evaluation.

1 EARLY REFLECTIONS IN SMALL ROOMS

1.1 Detection and Perceptibility of Individual Reflections in Listening Rooms

In 1988 Olive and Toole [6] reported on a comprehensive and detailed set of experiments relating to threshold detection and image shift in different listening spaces. Fig. 1 shows their experimental setup. The same set of experiments was conducted in ah anechoic chamber, then in an IEC standard listening room in an untreated (reflective) mode, and finally in the same room in its "relatively reflection-free" (RRF) mode. A comparison of simulated reflections versus reflections recorded in the original source signal is also presented in [6].

Energy–time curve (ETC) measurements were taken at each stage to establish the relationship of such measurements with subjective assessments. They reported that the frequency spectrum of reflected sounds is normally reduced in high-frequency content, but reducing the high-frequency content of reflected sounds had a relatively minor effect on the absolute threshold of several common sounds. However, the ETC measures of these sounds changed considerably, leading to the possibility of underestimating the audibility of a reflection when using the peak ETC level as an indicator. The frequency response of the reflection seemed to be a more reliable indicator. The unsuitability of ETC as a measure is discussed in detail in later sections.

One of the most important findings of this research, which is taken as a benchmark for the work described here, was the results of the image shift experiments in three different listening conditions—anechoic chamber, RRF listening room conditions, and standard IEC listening room untreated. The IEC listening room in untreated or "live" form had a midfrequency reverberation time of 0.4 s. The transformation of this room into an RRF room involved careful positioning of drapes and the addition of patches of sound-absorbing material to reduce the amplitude of early reflections from room boundaries. This treatment reduced the amplitude of early reflections to 15 dB below direct sound. Figs. 2 and 3, reproduced from [6], show the absolute and the image shift thresholds, respectively. (The absolute threshold was the point at which a listener could just detect the effect of a reflection and the image shift threshold was when it began to cause a change in the perceived position of a source.) It is no surprise that the lowest threshold was obtained in the anechoic conditions. However, up to a reflection time delay of approximately 30 ms the thresholds changed relatively little between anechoic and RRF conditions and went up by 6 dB in moderately reverberant conditions. In contrast, above 30 ms the change pattern was quite clear, with the thresholds for delayed reflections rising sharply as the listening conditions become more reflective.

They found that discontinuous and continuous sounds exhibit fundamentally different absolute thresholds as a function of the reflection delay, but only when there is little or no reflected sound in either the recording or the listening room. In nonanechoic conditions, however, the reflected sounds and the room reverberation add continuity to otherwise discontinuous sounds. Hence the absolute thresholds for most sounds under these conditions tend to be similar. They also found that natural reflections and reverberation in a listening room have relatively little effect on the absolute threshold of individual lateral sounds delayed by less than 30 ms. This was true for speech only.

Another key research project, similar to Olive and Toole's work, was the Archimedes project, started in 1987 jointly by B&O A/S, KEF Audio Ltd., and the Technical

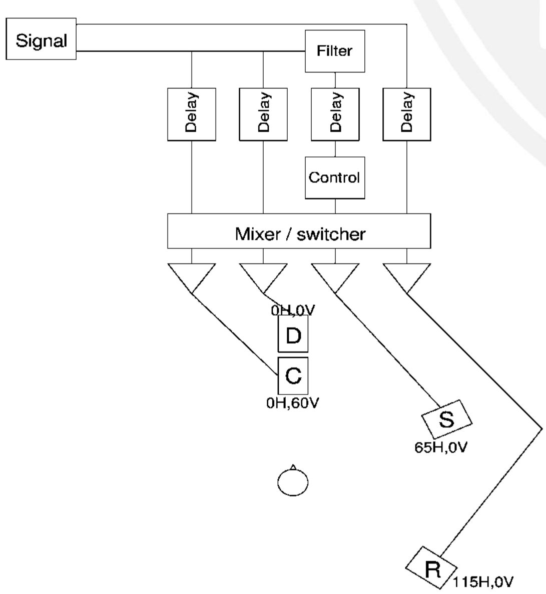


Fig. 1. Experimental setup of Olive and Toole [6]. D—direct sound; C—ceiling reflection; S—sidewall reflection; R—reverberation.

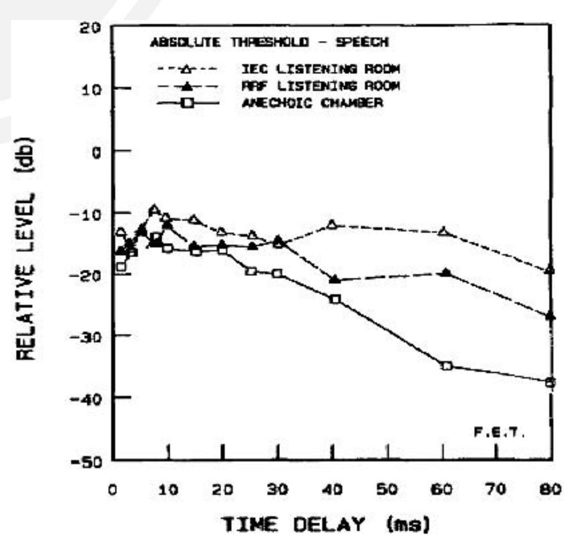


Fig. 2. Absolute thresholds for single lateral reflection using speech, measured in anechoic chamber, RRF room, and IEC-type listening room [6].

University of Denmark. The aim of this project was to investigate the influence of individual reflections on the timbre of reproduced sound. A single loudspeaker with frequency-dependent directivity characteristics, positioned in a room of normal size with frequency-dependent absorption coefficients of its surfaces, was simulated using an electroacoustic setup in a large anechoic chamber (1000 m^3). The model included the direct sound, 17 individual reflections, and the reverberant field [7]. The dimensions and the acoustic properties of a listening room built in accordance with IEC 268-13 [2], generally representative of a domestic listening room, were modeled in the experiment. Dimensions of the room were 7.52 m long by 4.75 m wide by 3.16 m high with a ceiling suspended at 2.8 m. In order to determine and limit the number of loudspeakers producing individual reflections, the mean impulse density of the room to be modeled was calculated. The early part of the sound field is characterized by an increase in the impulse density as a function of time. However, the human hearing system, shortly after the onset of continuous sound, perceives the whole sound field as continuous, which suggests that changes in impulse density are relevant only up to a certain point. Research and experiments suggest that 50% of the subjects cannot detect an increase in impulse density if the mean impulse density is higher than 2000 s^{-1} [7]. The mean impulse density as a function of time for a rectangular room with reflecting walls can be approximated by

$$\frac{\Delta N}{\Delta t} = \frac{4\pi c^3}{V} t^2 \quad [s^{-1}]$$

where c is the speed of sound (340 m/s), V is the room volume in cubic meters, and t is the elapsed time in seconds after the arrival of the direct sound at the receiver position [7]. By solving this equation for the room studied, an onset time of 21.4 ms relative to the direct sound was calculated for the subjectively diffuse part of the sound field. It was soon established that the total number of reflections with delays in the range of 0–22 ms was too large for individual representation by a loudspeaker in the anechoic chamber. Thus only reflections with a level above −20 dB relative to direct sound were modeled, and several reflections were represented by the same loudspeaker if their angles of incidence were not subjectively different. This resulted in the simulation set described, consisting of direct sound, 17 reflections within 22 ms after direct sound, and a reverberant field of reflections arriving later than 22 ms after the direct sound. The electroacoustic setup used is shown in Fig. 4.

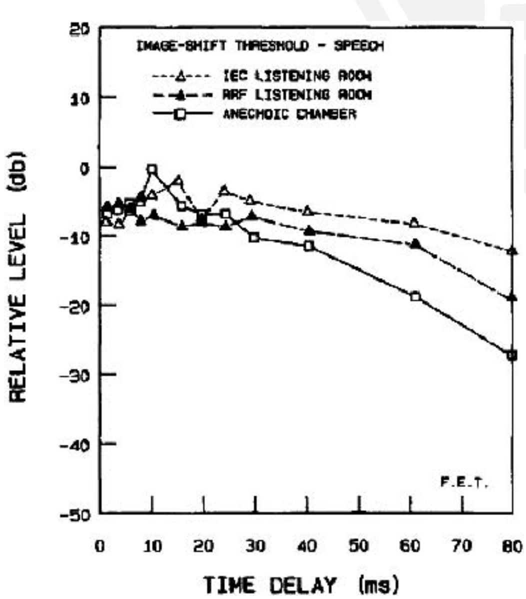


Fig. 3. Image shift threshold indicating minimum level at which lateral reflection is perceived to cause a spreading or shift in localization of auditory event associated with direct sound [6].

Bech gave a detailed account of these experiments in [7], [8]. His general comments on this electroacoustic simulation of a room environment implies that this setup is the closest one can get to reality and that the results are therefore believed to be representative of conditions in real rooms. However, the missing influence of the standing-wave structure of the modeled room was noted. Bech revised the setup with added improvements by modeling both the absorption coefficient of the room surfaces and the directivity of the simulated loudspeaker as a function of frequency. The subjective diffuseness of the reverberant field was also improved by increasing the number of uncorrelated channels generating the reverberant part of the sound field, as reported in [8].

A summary of results in [9] suggested that only first-order ceiling and floor reflections individually contributed to the timbre of the reproduced noise samples. It was also suggested that the threshold of detection is determined by the spectral changes in the dominant frequency range of 500 Hz to 2 kHz. The effects on the threshold detection (TD), by the application of low and high-pass filters to individual reflections, agreed with the findings of Olive and Toole.

Bech also looked at the effects of early reflections on the spatial aspect of reproduced sound in small rooms using the same experimental setup. The results indicate that the spectral energy above 2 kHz of the individual reflection determines the importance of the reflection for spatial changes to the perceived source image, and only the first-order floor reflection is strong enough to contribute separately to the spatial aspect of the sound field [9]. The contribution of low-frequency reflections to spatial attributes such as envelopment and spaciousness was not considered.

1.2 Early Reflections in Critical Listening Rooms

Bech's TD values were used by Walker in the design of a new type of control room [10]. His design was based on using hard reflectors to create a reflection-free zone, as opposed to using traditional absorbent materials around the source and listener. This increased the reverberation time because nonabsorbent materials were used to redirect the early reflected sound energy away from the main listening position. He reported that by using this approach, a higher and a more natural reverberation time could be achieved, and a clearer stereophonic image that was less dependent on the room was predicted.

Walker [10] reported the results of acoustic measurements of a test room which, after the loudspeaker heights were adjusted, showed that the reduction of all reflected sound energy at the main listening position, in the period up to about 15 ms after the arrival of the direct sound to levels below −15 dB relative to the direct sound, had essentially been achieved.

The ITU-R BS.1116 standard [1] puts forward a reasonably good set of design parameters for stereo and surround reproduction. Early reflection control is given high priority, and any reflection after the first 15 ms of the direct sound must be at least 10 dB lower than the direct sound. However, the allowable reverberation time window in relation to the room dimensions results in the rooms being less reverberant than rooms designed to the IEC 268-13 standard [2].

2 DESIGN CONCEPT OF ACTIVE LISTENING ROOM

2.1 Need for a Flexible Approach to Sound Field Simulation in Rooms

Sound field simulation in an anechoic chamber is usually very elaborate and expensive. Moreover, it does not bear much resemblance to the sound field in real rooms as only some of the acoustic characteristics can be reproduced due to the limitations of loudspeakers and the associated processing that can be used in a confined space. As stated previously, Bech noticed in [9] the lack of naturalness of the elaborate experimental simulation used in the Archimedes project, described earlier, and commented on the lack of low-frequency features that would be found in a real room.

Olive and Toole's approach in [6] was more adaptive. However, the transformation of the IEC room to the RRF room involved the use of absorption materials to reduce the amplitude of early room reflections, which in turn reduced the reverberation time of the room. Therefore no independent control of early and late energy was available, and the level of early reflections in the RRF room, although low in amplitude, was fixed in relation to the single simulated artificial reflection used to establish the absolute threshold of detection and the image shift detection thresholds.

For the purposes of this paper, the scope of sound field simulation in critical listening rooms is limited to early reflections alone. Olive and Toole's findings in [6], relating to detection thresholds of image shift and the associated room conditions, are taken as benchmark values for the design of a novel simulation arrangement that involves the creation of a perceptually reflection-free zone at the listening position without using absorptive materials. Walker's principles of sound deflection [10] to reduce the amplitude of early room reflections, so as to create a reflection-free zone comparable to Olive and Toole's RRF room conditions, are employed. The proposed deflector panels have built-in transducers, which are excited with the processed source signal to simulate artificial early reflections.

This paper presents a new concept of sound field simulation in listening rooms. Here the original low-frequency acoustic properties of the listening room remain unaltered by the creation of a perceptually reflection-free zone (level of early reflection −10 dB relative to the direct sound up to 15 ms within the frequency range of 500 Hz to 8–10 kHz) using the deflection of sound by means of panels. These panels in turn are used for generating artificial reflections,

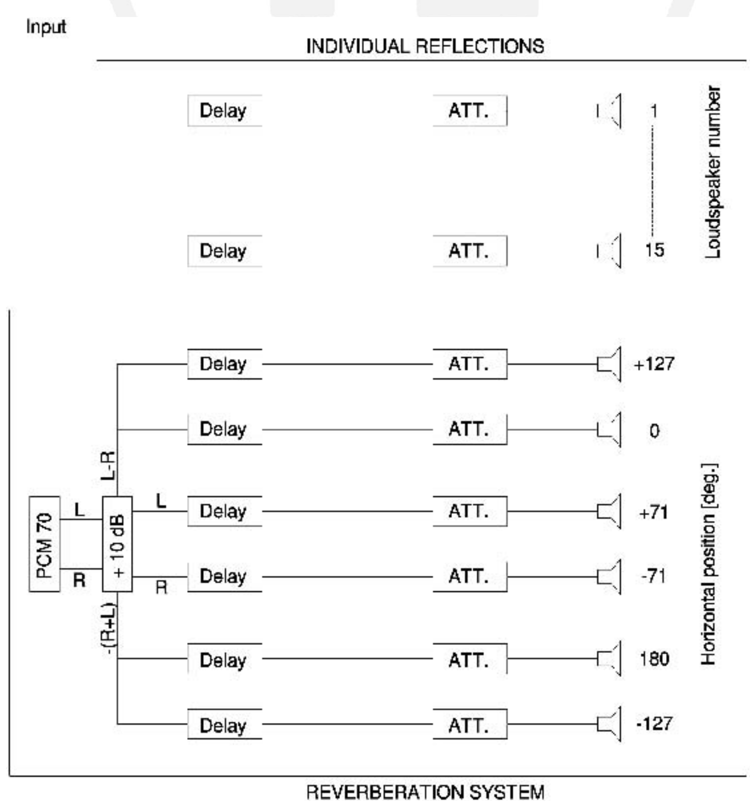


Fig. 4. Block diagram of Bech's complete experimental setup [7].

describing the various design and modeling stages, validation experiments, room measurements, and results of a pilot listening experiment.

In the following sections the acoustic properties of the listening room and the source loudspeaker used in all experiments relating to this new sound field simulation technique are documented along with a brief description of the deflection panel arrangement used for a pilot validation experiment.

2.2 Acoustic Properties of Listening Room

The experiment was set up in a listening room built to the ITU-R BS.1116 specification [1] at the University of Surrey. The approximate internal dimensions of the room were 7.35 by 5.33 by 2.50 m. Internal room finishes were carpet on floor, lay-in grid tile absorbent ceiling, and full-range acoustic absorber boxes on the walls, as illustrated in Fig. 5.

As prescribed in ITU-R BS.1116 [1], the amplitude of the early reflections, measured at the listening position, was found to be −10 dB relative to the direct sound from all five loudspeakers, within the frequency range of 1–8 kHz, during a time interval of 15 ms.

The measured reverberation time was 0.245 s at 1 kHz and was found to be within the specified reverberation time window for upper and lower limits in all relevant one-third octave bands. The measured values are shown in Fig. 6.

The measured ambient background noise level in the room with the ventilation system and technical power switched on is presented in Fig. 7. ITU-R BS.1116 [1] requires the background noise levels not to exceed NR15. This was found to be the case.

2.3 Source Loudspeakers

The chosen source loudspeakers were two-way active units with integral amplifiers. According to the manufacturer's literature, they were classified as wide-dispersion studio monitors with an operational bandwidth of 60 Hz to 18 kHz ± 3 dB. The loudspeaker was mounted on a stand with the center of the cabinet elevated 1.25 m from the floor. The unequalized frequency response of all five loudspeakers, arranged according to ITU-R BS.1116 [1] for multichannel listening, was found to be within the recommended tolerance window.

2.4 Pilot Experiment

In a pilot experiment four large, angled deflector panels between the source loudspeaker and the listening position were used in a mono source configuration. The deflector panels around the source loudspeaker, which was located

Fig. 5. Listening room finishes and five-channel loudspeaker arrangement.

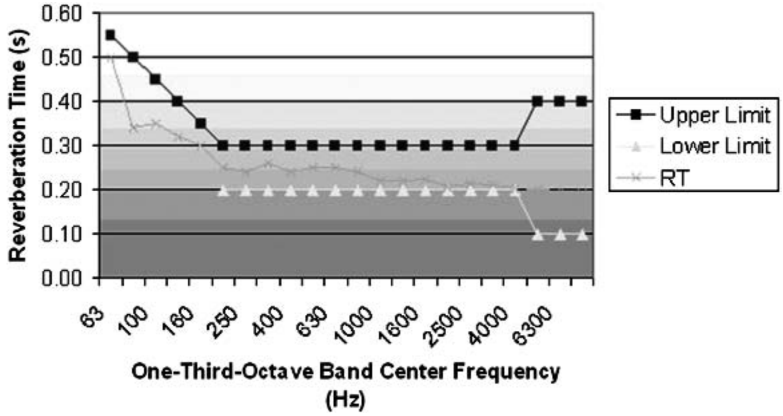


Fig. 6. Measured reverberation time of LR1 at University of Surrey.

in one corner of the room, were angled in such a way that any sound hitting these panels was forced away from the listening/measurement position. The reason for setting up this arrangement in the corner was to force any reflections from room surfaces away from the measurement position so that the effectiveness of the deflector panels could be discretely quantified.

2.5 Computer Modeling

The experimental setup was based on a computer model created in a commercially available software package, CATT Acoustic.[1] The two main reasons for creating an acoustic model were as follows:

- To optimize the experiment setup in terms of angles and positions of the panels to create a reflection-free zone at the listener/measurement position.
- To predict the reflection patterns from the panel arrangement in particular and the reflection patterns of the setup as a whole within the first 20 ms time window.

The model was constructed for a closed room with internal elements as floating objects and is shown in Fig. 8. The absorption coefficients of walls, ceiling, and floor were adjusted to get a close match between predicted and measured values of reverberation time. The directivity/dispersion of the source loudspeaker was modeled accurately. The positions of the source loudspeaker and the deflector panels were adjusted for maximum deflection of early reflections away from the listening and measurement position. The predicted early reflection pattern is shown in the CATT Acoustic prediction window in Fig. 9.

Fig. 9 shows the level and time delay of the first-order primary reflections with the panels in place around the source loudspeaker. No first-order diffused reflections or any second- or third-order reflections were modeled. As stated in [10], most of the second-order reflections can be ignored on the basis of the total time delay and natural attenuation with distance, and any reflections with a total path longer than 6.34 m will already be attenuated by at least 10 dB relative to the direct sound by spreading loss. Also, due to the absorbent nature of the surrounding room surfaces, any diffused reflections within the 20 ms time window will be attenuated to well below the −10 dB level.

The model predicted a complete reflection-free zone up to 20 ms with the exception of a single reflection at 12 ms, which was found to be coming from the spaces around the floor and ceiling deflector panels. As this reflection was attenuated by 20 dB in relation to the direct sound, the panel arrangement was considered to be satisfactory. This arrangement was used in the construction of the physical experimental setup.

The physical size of the panels determined the wavelength of the sound waves that could be deflected. Thus lower frequency sound waves having a wavelength comparable to or larger than the panel size would not be deflected by this arrangement. The lower frequency cutoff limit was calculated to be approximately 400 Hz. Each of

[1]CATT Acoustic is a registered trademark of CATT-Acoustics.

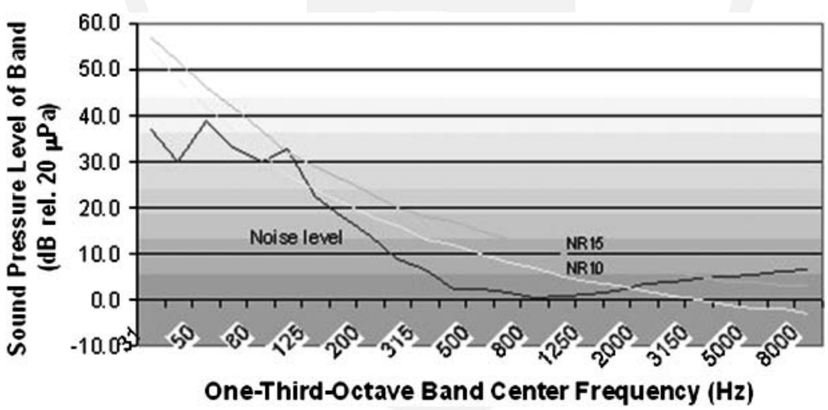


Fig. 7. Measured background noise in LR1 at University of Surrey.

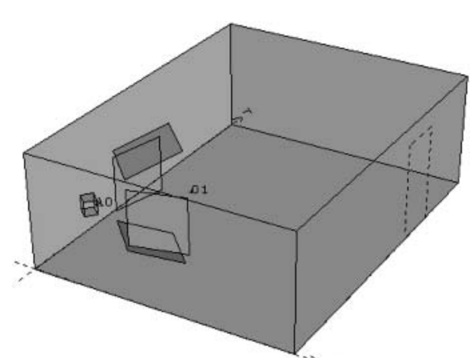


Fig. 8. Three-dimensional plot of modeled panel arrangement.

these panels was made up of 18-mm-thick MDF (medium-density fiber board) with four 600- by 600-mm cutouts for the flat panel loudspeakers. The flat panels were embedded within the rebated recesses so that mounting was flush with the surface of the MDF panel. The physical setup is shown in Fig. 10. A detailed discussion of the dispersion characteristics of flat panel loudspeakers in comparison with conventional loudspeakers is presented in [11]. Based on measurement data, the case for using flat panel loudspeakers in this application is also discussed in [11].

2.6 Room Measurements

The measurement of early reflections is a complex issue. It is well known that the impulse response of a linear time-invariant system theoretically contains all of the information necessary to specify the system response fully. However, due to the unavoidable combined time–frequency resolution limits, a simultaneous and complete identification of time delay, bandwidth, and amplitude of an acoustic event is not possible [12]. For the purposes of this experiment it was assumed that a measurement system with well-defined Fourier transform windows having a time resolution of 1–2 ms and a frequency resolution of about 500 Hz would be adequate to measure the room acoustic responses from 1 kHz upward [12]. According to the results of previous work discussed earlier, those are the main frequencies giving rise to the directional information that might be disturbed by early reflections.

Walker describes the two measures of short-time response and the ways of usefully displaying the early reflection patterns [12]. The first is the energy–time curve (ETC), which is the magnitude of the complex system impulse response, usually taken to represent instantaneous energy. It presents a useful view of the time-domain response in which discrete reflections are easily observed. However, the precise theoretical nature of the ETC is seriously disputed [13]. Olive and Toole's [6] views on ETC measurements of early reflections are far from complimentary. Their experience of the audibility of early reflections compared to ETC is that the most significant visible elements of an ETC are contributed by the high-frequency components in signals. The sharply rising spikes of energy are attenuated rapidly if the high-frequency components are removed from the signal, but they change very little if the low-frequency components are eliminated. Hence the audible significance of reflections cannot be evaluated by means of the height of energy spikes in an ETC as it can be misleading. The practical consequence of this observation is that using the peak level of ETC spikes as a measure of the audible importance of room reflections can lead to a serious underestimation of the importance of reflected

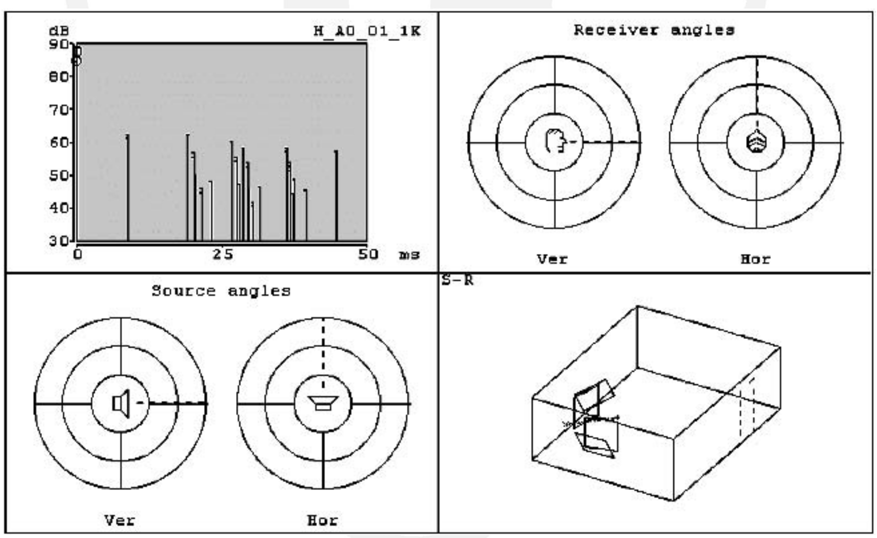


Fig. 9. Time–amplitude plot of predicted early reflections.

Fig. 10. Pilot experiment panel arrangement.

sounds and the normal off-axis response of loudspeakers, which contain less high-frequency energy than direct sound.

The second measure is the three-dimensional energy–time–frequency response (ETF), or the "waterfall" plot, in which the start of the Fourier transform block is progressively shifted in time to produce a series of frequency responses at different times. The time resolution is limited by the length of the transform window. Despite some limitations, this is the most useful representation of reflected sound in time and frequency, and it is used widely in the spectral analysis of reflected sound. For the purposes of this paper, both methods are used.

2.7 Instrumentation

A maximum-length-sequence system analyzer (MLSSA) measurement system was used to measure the early reflection pattern from the source loudspeaker at the given listening position in the experimental setup described in the previous section. MLSSA is a PC-based acoustic analyzer. It generates pseudorandom maximum-length sequences and cross-correlates these with the measured signal to compute the impulse response of a system. A 1-in (25.4-mm) measurement microphone was used at a distance of 1.50 m from the source loudspeaker and at a height of 1.25 m from the floor. The MLSSA output signal was fed directly into the built-in amplifier of the source loudspeaker.

2.8 Sequence of Measurements

The first set of measurements was taken with the panels in a passive mode. This was in order to examine the reflection patterns of the panel arrangement at the measurement position. The second set of measurements was taken with the panels in an active mode. The MLSSA signal was split into two and was fed to the source loudspeaker and one transducer panel in each 1.35 m^2 MDF deflector panel simultaneously. A time delay of 10 ms was used in the transducer signal to separate the impulse of the source loudspeaker and the simulated reflection in time sufficiently. A total of eight measurements were taken, of which only three are presented here and discussed. These are representative of the key findings of this experiment.

The ETF plot in Fig. 11 the spectrum of direct sound and reflected sound energy within a 45-ms time window. No panel was active at this stage, and the panels were acting as deflectors only. The first arrival (the direct sound) is at approximately 5 ms, and the first significant reflection is shown approximately 2.5 ms later, which is from the top edge of the floor deflector panel. The amplitude of this and other subsequent reflections are well below −20 dB relative to the direct sound. This shows a clear reflection-free zone, within the perceptual criteria set out, around the measurement position for up to 30 ms relative to the direct sound.

Fig. 12 shows the spectrum of artificial reflections generated by the flat panels within the reflection-free zone. The panels were activated at different amplitudes to each other with an initial time delay of 10 ms relative to the main source loudspeaker. The ETF plot shows that the first arrival from the main source loudspeaker is at approximately 5 ms, whereas the first and subsequent arrivals of panel-generated reflections have an initial offset delay of 12 ms (electronic delay + physical distance from microphone).

2.9 Conclusion

It is clear from the measurement results that the objectives of this first experiment were achieved as the integrity of the reflection-free zone was preserved. The concept of deflector panels with embedded flat panel loudspeakers in an angular panel arrangement around a source loudspeaker fulfills the fundamental requirements of the proposed simulation setup. The individual flat panel loudspeakers embedded in the deflector panels were also found to be capable of generating discrete artificial reflections without having any adverse effect on the integrity of the reflection-free zone.

3 DEVELOPMENT PHASE OF THE ACTIVE LISTENING ROOM

3.1 Introduction

The expansion and development of the active listening room was undertaken in two phases. First a deflector panel

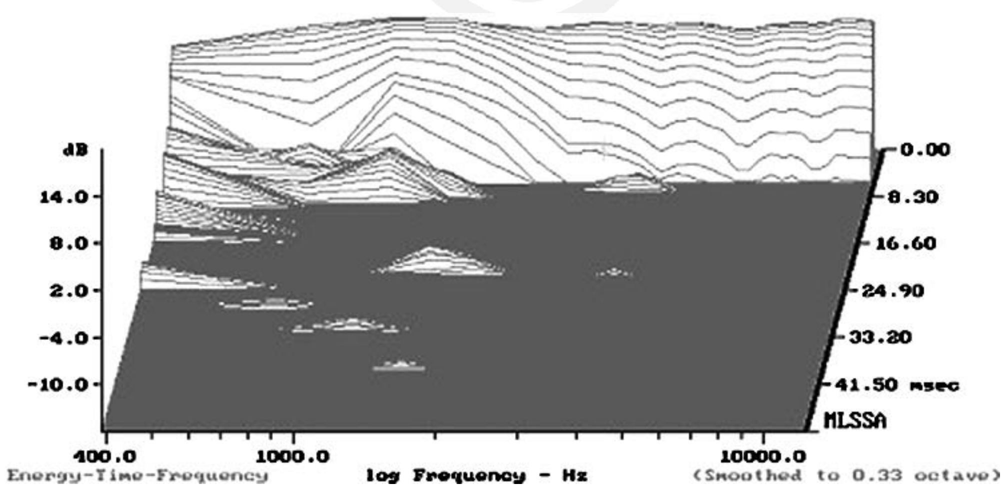


Fig. 11. ETF plot of pilot experiment panel arrangement.

arrangement involving eight panels around a two-channel stereo loudspeaker setup was designed and modeled. The second and final stage involved the design of a five-channel surround simulation setup where a set of deflector panels was clustered around each loudspeaker in a standard ITU surround reproduction loudspeaker arrangement. The flat panels, unlike in the pilot experimental setup, were not embedded in large panels but were used as individual deflectors. The panel sizes used were 600 mm^2. It was calculated that this size panel would be capable of deflecting sound waves above 500 Hz and should therefore create a perceptually reflection-free zone with a low-frequency cutoff of 500 Hz.

In line with findings of previously mentioned research [6], [9], this low-frequency cutoff was considered suitable because the frequency range of interest for investigations into the effects of early reflections on many spatial and timbral effects is between 500 Hz and 2 kHz. With this as the design criterion, the two configurations, namely, the two-channel arrangement and the five-channel arrangement, were developed.

3.2 Two-Channel Arrangement

Before creating the acoustic model a ray-tracing diagram was done to identify the main paths of significant reflections from the source loudspeaker in a symmetrical listening arrangement (Fig. 13). The deflector panels were then arranged in such a way as to deflect the sound away from the listener position to create a reflection-free zone. It was clearly evident that this simple geometric arrangement not only deflected the sound away from the listener but also provided masking of side wall reflections.

A computer model of this panel arrangement around the two source loudspeakers was created with the angular panel settings optimized to create a near-field geometric boundary setting, which forced the early reflections to be directed away from the listening position.

As before, the model was constructed for a closed room with internal elements as floating objects. The absorption coefficients of the walls, ceiling, and floor were adjusted to get a close match between predicted and measured values of the reverberation time. The directivity and the dispersion of the source loudspeaker were modeled accurately. A three-dimensional plot of the deflector panel arrangement is shown in Fig. 14.

Fig. 15 shows the predicted reflectogram as a time–amplitude and path display of significant early reflections up to third-order paths. It should be noted that in this plot the reflection patterns are represented as broad-band reflections containing energy well below 500 Hz. The first

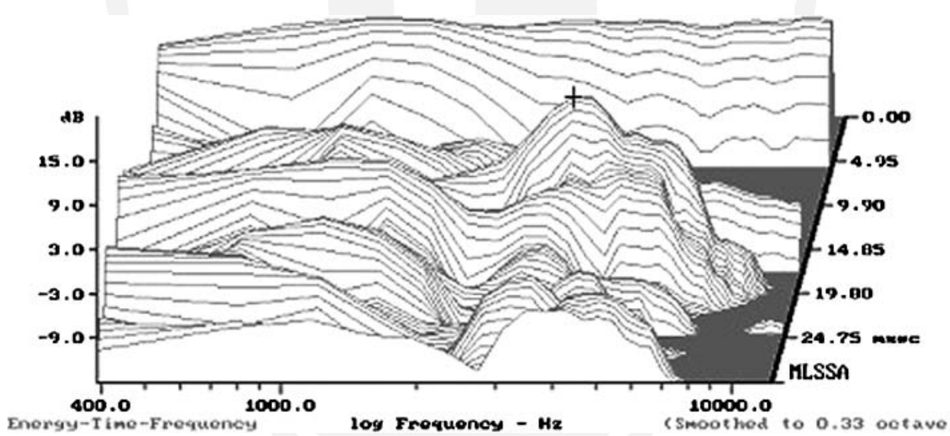


Fig. 12. ETF plot of artificial reflections generated by flat panels embedded in deflector panels.

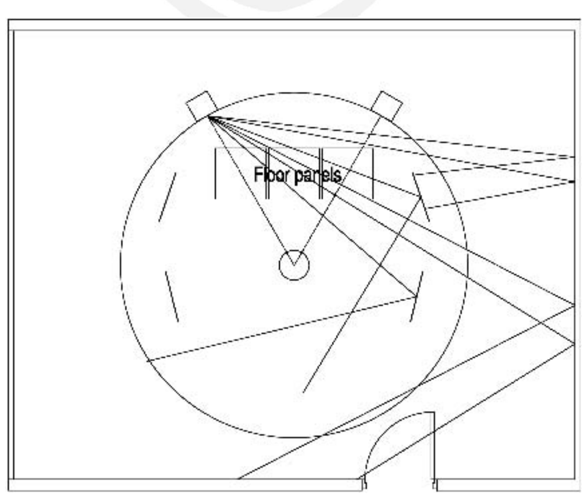


Fig. 13. Two-channel panel arrangement.

reflection, as shown in Fig. 15, is arriving from the back of the source loudspeaker and can only be low-frequency energy that is not sufficiently attenuated. Also, as the loudspeaker dispersion above 500 Hz is mainly concentrated in the frontal hemisphere, it is considered reasonable to ignore this reflection altogether. The remaining reflections are shown to be attenuated by at least 15 dB, which satisfies the design criterion. Furthermore it can be safely assumed that due to the broad-band nature of these predicted reflections, the actual level of any room surface reflections within the panel arrangement above 500 Hz will be further attenuated and would be on the order of −20 dB relative to the direct sound.

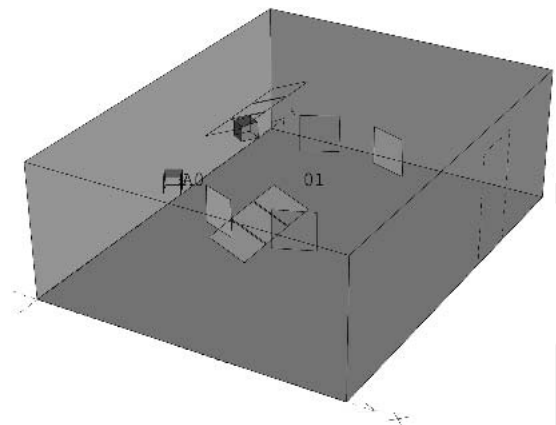


Fig. 14. Three-dimensional view of two-panel arrangement.

As this was an intermediate design stage, rather than acquiring room measurement data, a desktop analysis exercise was used to look at the spectral distribution of the room reflections within this panel arrangement. An impulse response of the room was acquired from the model for postprocessing of time–amplitude–frequency analysis with ETF plots, similar to the ones presented for the pilot experiment.

Fig. 16 is the MLSSA-generated waterfall ETF plot of the impulse modeled, showing all reflections within a 40-ms time window from 400 Hz to 12 kHz. The spectral profile of the reflected energy is noted to be well controlled below critical limits.

3.3 Five-Channel Arrangement

Before creating the acoustic model, an ideal reflection path model was sketched in Fig. 17, in which the locations of the panels were derived from a spherical boundary created around the listening position from which artificial

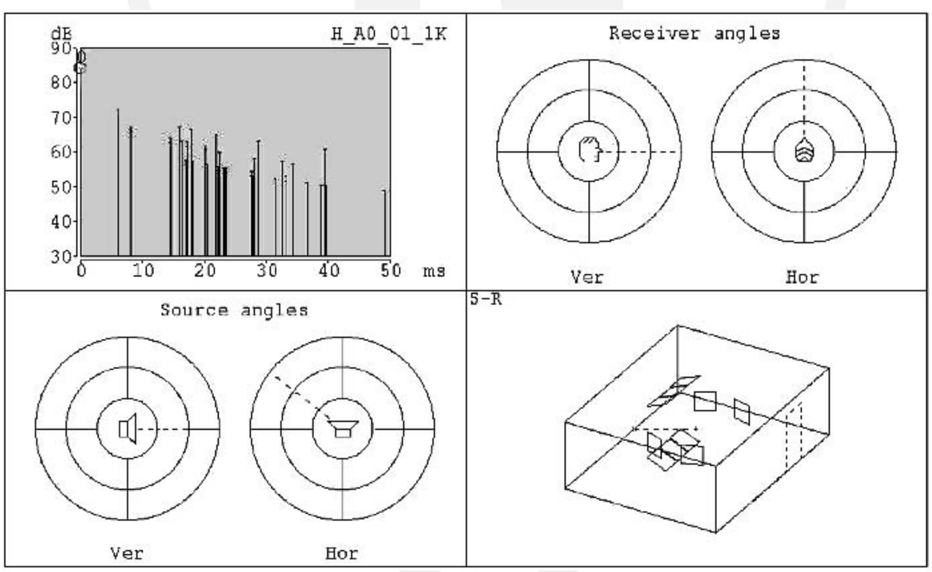


Fig. 15. Time–amplitude plot of predicted early reflections.

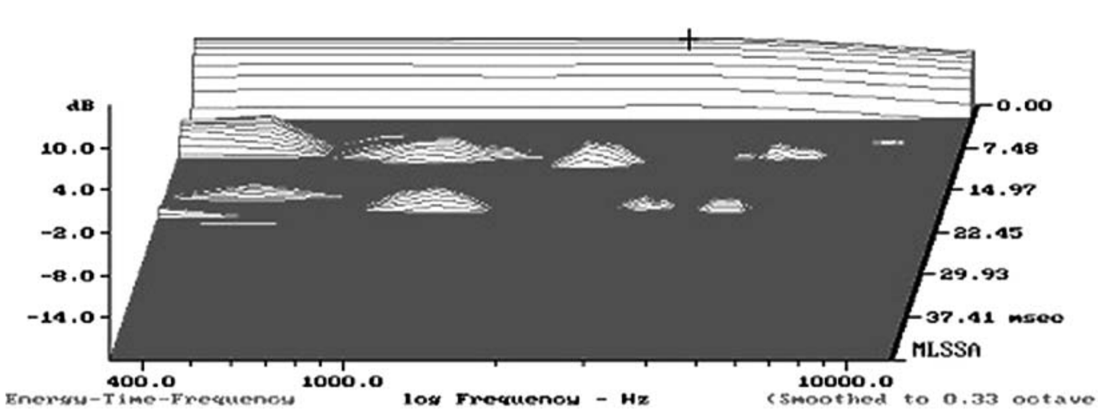


Fig. 16. ETF plot of modeled impulse response.

reflections could be created. Also, this basic model was useful in determining and identifying the main paths of significant reflections from the source loudspeakers in a typical, completely symmetrical listening arrangement. The deflector panels were then arranged in such a way as to deflect sound away from the listener position so as to maintain a reflection-free zone (Fig. 18).

Fig. 19 shows the predicted reflectogram as a time–amplitude and path display of significant early reflections up to third-order paths. As mentioned before, due to the broad-band nature of these predicted reflections we can ignore the first significant reflection from each loudspeaker, and it can be assumed with confidence here as well that the actual level of any room surface reflections within the panel arrangement above 500 Hz will be further attenuated and would be on the order of −20 dB relative to the direct sound. The room arrangement for each source is shown in Fig. 20.

3.4 Room Measurements

Fig. 21 shows the MLSSA ETF plots of the actual room measurements of the five source loudspeakers within the deflector panel arrangement. All plots show the amplitude and time distribution of the early reflection patterns within the panel arrangement up to 30 ms. It is clear that in each case the amplitude of the early reflections is below −20 dB above 1 kHz and −15 dB above 500 Hz, with relation to the direct sound. It can be seen that a fairly uniform reflection-free zone, within the perceptual criteria set out and approaching semianechoic conditions (−20 dB in relation to the direct sound), is achieved for each source loudspeaker.

4 PILOT LISTENING EXPERIMENT

This section describes the electroacoustic simulation arrangement and the results of a pilot listening experiment with a fixed set of artificial early reflections. The objectives and methodology of the listening experiment are discussed along with results and future aims of the project.

4.1 Signal Processing

Fig. 22 shows a complete generic block diagram of the source signal routing and processor layout, capable of generating 16 individual reflections. The processor has exclusive control for each reflection parameter, such as 24-band parametric equalizer, delay, and attenuation. There is a down-mixing stage to generate the late part, that is, control for varying the reverberation of the room.

At this stage our efforts are concentrating only on the generation of a set of early reflections with the active

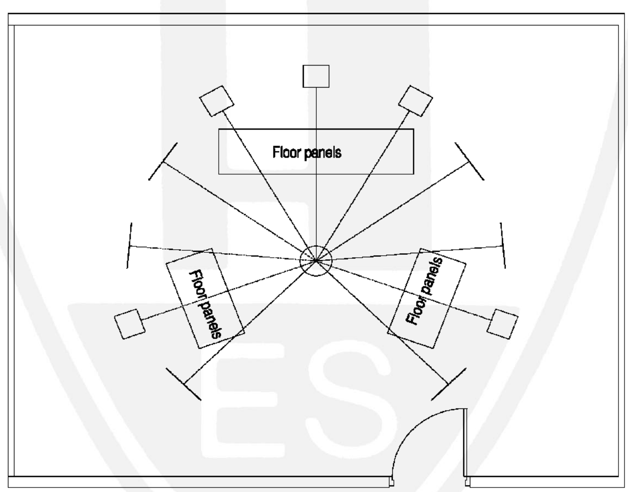


Fig. 17. Floor plan of five-channel panel arrangement.

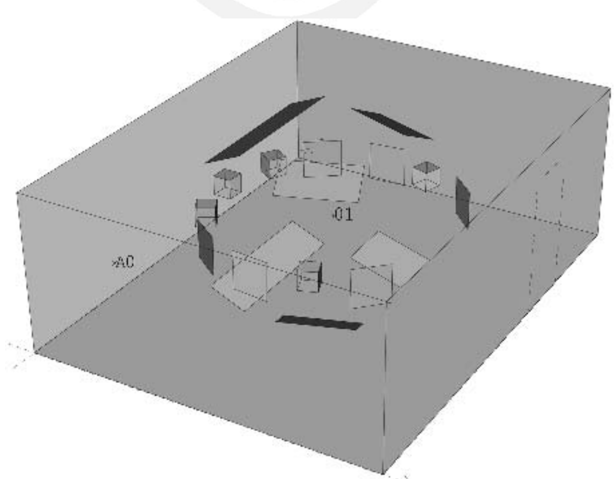


Fig. 18. Three-dimensional view of five-channel panel arrangement.

listening room panel arrangement. In the second stage the late reverberation part will be implemented. In the following section the selection and implementation of a set of early reflections corresponding to a known listening environment are discussed.

4.2 Implementation of Artificial Reflections

As a starting point it was considered appropriate to implement a set of early reflections that corresponded to a room built to a known standard other than ITU-R BS.1116 [1] and to conduct some listening experiments to evaluate the differences.

For this pilot experiment a set of 12 early reflections was derived from a room built to IEC 268-13. Data on the properties of these reflections were acquired from Bech's experimental setup described in [9] and were implemented in our listening room in the manner listed in Table 1.

4.3 Timbral Properties of Artificial Reflections

It is important to stress here that the aim of this simulation was not to be absolutely accurate in the spectral reproduction of each artificial reflection but to emulate a close representation of the spectral profile of each reflection.

A 24-band parametric equalizer was used to achieve a close match of the spectral profile of each reflection based on published data [9] for the IEC 286-13 type listening room. Fig. 22 shows the processor components. Fig. 23 shows the time and spectral distributions of these reflections in an MLSSA ETF waterfall plot.

The reverberation time and the background noise level of the room with these artificial reflections in place was also measured. It was found that there was hardly any difference (less than 5%) in the midband reverberation time of the room with the artificial reflections on and off. This may be due to the fact that the level(s) of artificial reflections were low compared to the direct sound and that these artificial reflections were efficiently absorbed by the relatively absorbent room finishes of the listening room. The background noise level with the reflections off was found to be slightly elevated due the slight hiss from the panels, although the measured background level with the hiss was still found to be below NR15.

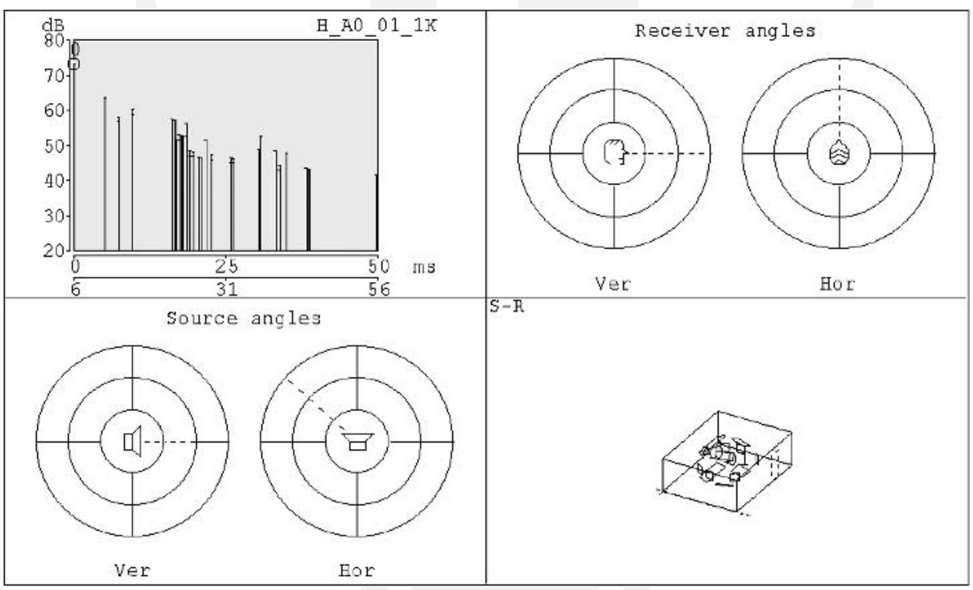


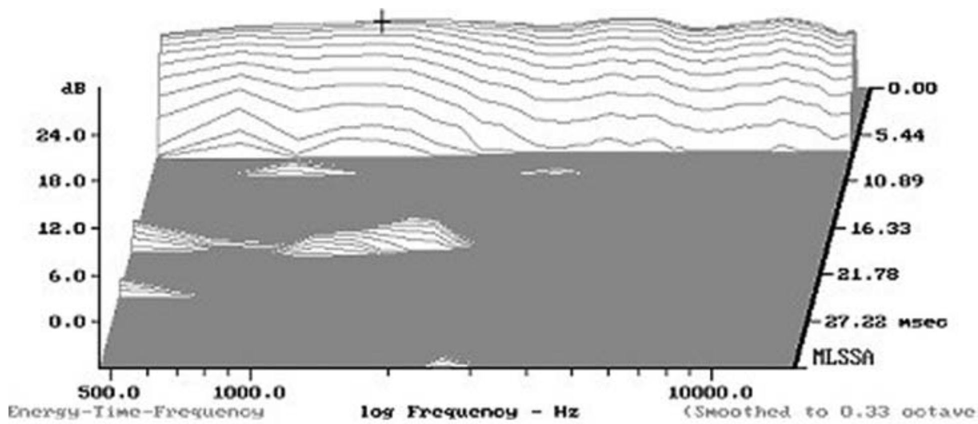
Fig. 19. Time–amplitude plot of predicted early reflections.

Fig. 20. Source loudspeakers and panels for each source.

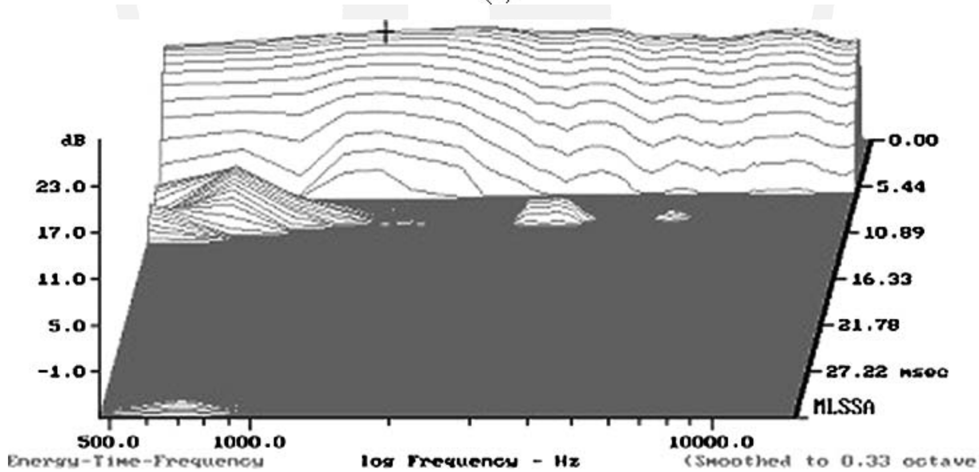
4.4 Stimuli for Pilot Listening Experiment

A relatively simple listening experiment was conducted to evaluate the perceptual effect of the arrangement described. This was in order to determine whether or not the simulated changes in the listening room characteristics were at all detectable on real program material, and to get some idea of the nature of these effects.

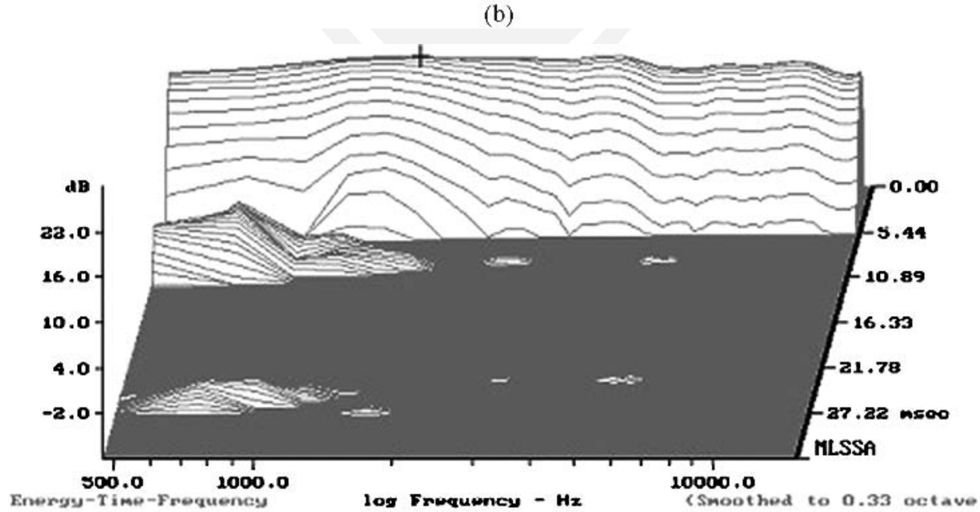
A stimulus was chosen that had been designed to demonstrate the perceptual effect of changes in the source distance to the listeners, and which varied along the perceptual distance dimension in a unidimensional manner [14]. This stimulus itself contained various natural-sounding reflection and reverberation components at different levels, depending on the simulated closeness of the source instrument. This was chosen in preference to anechoic source material (which would probably have been more critical in terms of absolute detectability) in order to test the setup under more ecologically valid conditions involving naturalistic program material. If the changed



(a)

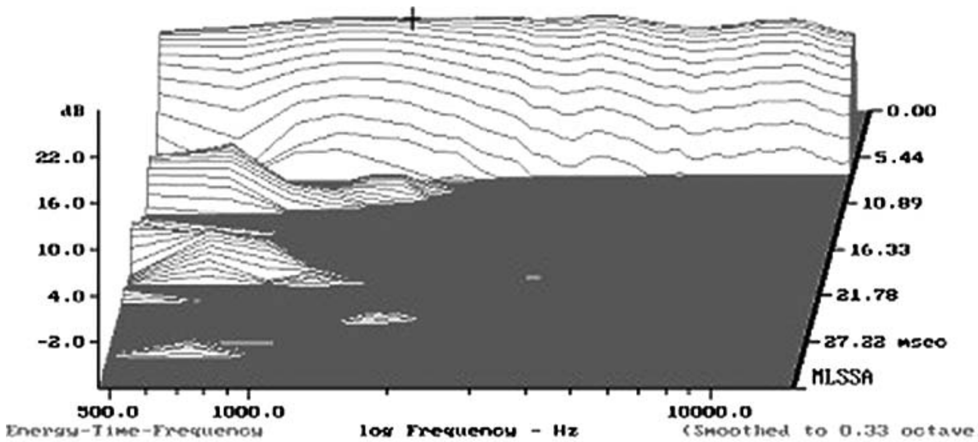


(b)

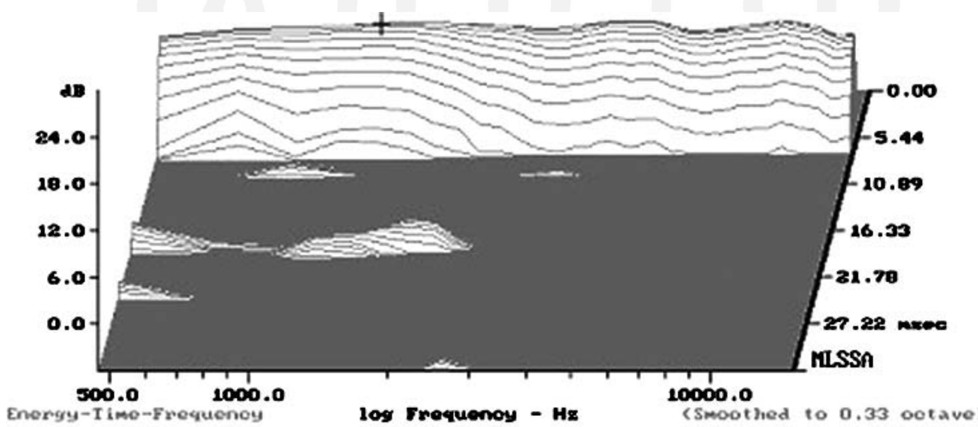


(c)

Fig. 21. ETF plots and reflected energy at listener position with five-channel panel arrangement. (a) Center loudspeaker. (b) Left loudspeaker. (c) Right loudspeaker. (d) Left surround loudspeaker. (e) Right surround loudspeaker.



(d)



(e)

Fig. 21. *Continued*

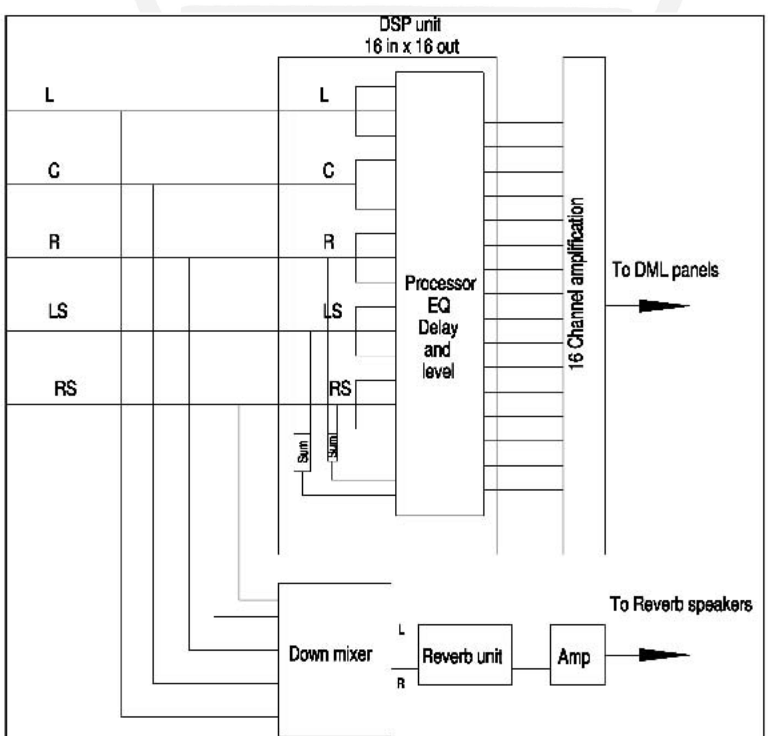


Fig. 22. Block diagram of signal and processor routing.

listening conditions could be detected using source material that was similar to real musical recordings, then there could be some merit in pursuing the use of the system in the evaluation of typical reproduced program material.

For the synthesis of the stimuli, a Lexicon[2] 480L reverberation processor had been used in conjunction with a mixing desk. The reverberation unit is equipped with four digital outputs and offers control over up to six discrete reflections, which are individually real-time adjustable in terms of delay and amplitude relative to the input signal [14]. A generic impulse response was designed consisting of three separate time regions,

1) Direct sound; $t = 0$ ms
2) Early reflections; 15 ms < t < 40 ms
3) Reverberation tail; t > 40 ms.

The source material was taken from the Archimedes CD [15] and comprised a comet, a trumpet, a male voice, and an acoustic guitar. All stimuli were mixed for reproduction over left (L), center (C), and right (R) only, the direct sound being routed to C. The perceptual effect of changes in the closeness of a source with respect to the listening position was produced by varying the relative gains (indicated as x, y, and z in Fig. 24) between the three chosen time regions. For further details see [14].

[2] Lexicon is a registered trademark of Lexicon Pro.

For the purposes of this pilot listening test, three sound samples of varying source distances (extremely close, intermediate, and extremely distant) were chosen for three instruments, namely, cornet, guitar, and trumpet. As the chosen stimuli required only the front three loudspeakers (L, C, R) for the complete reproduction of the stimuli, four of the twelve early reflections (left and right back and rear, as listed in Table 1), were derived from the left and right loudspeaker inputs accordingly.

4.5 Pilot Listening Tests

The objective of this pilot experiment was to investigate whether the artificial early reflection simulation had a perceivable effect. The listening test was based on the A/B/X

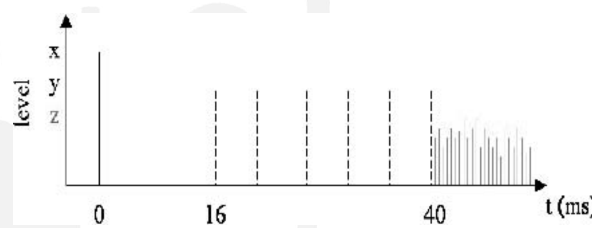


Fig. 24. Generic impulse response of reverberator used for distance processing [14]. --- six early reflections; ··· reverberation tail.

Table 1. Properties of individual simulated reflections.

Reflection Number	Location of Panel	Attenuation Referred to Source Loudspeaker (dB)	Delay Referred to Source Loudspeaker (ms)	Azimuth/ Elevation* (degrees)
1	Center floor	7	1.5	0/−42
2	Center ceiling	9	3.5	0/42
3	Left floor	7	1.5	30/−42
4	Left ceiling	9	3.5	30/42
5	Left front	12	7.5	55/0
6	Left back	12	9	85/0
7	Left rear	10	12.5	135/0
8	Right floor	7	1.5	−30/−42
9	Right ceiling	9	3.5	−30/42
10	Right front	12	7.5	−55/0
11	Right back	12	9	−85/0
12	Right rear	10	12.5	−135/0

* Left-hand side of listener defines positive angle.

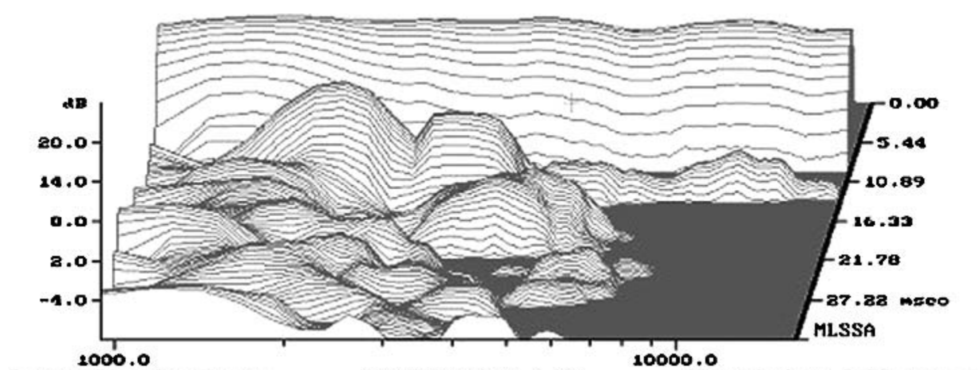


Fig. 23. ETF plot of simulated early reflections with reference to center loudspeaker.

methodology, which is summarized in [16]. In A/B/X tests the listening task for a trial is to compare X to A or B repeatedly until it is determined that X is the same as either A or B. It is unimportant in this method whether or not the listeners are told the content of components A and B because they will not know the identity of X, except by its sound. A series of trials, in which X is randomly reassigned to be the same as either A or B, constitutes a test. The A/B/X test has found considerable use in determining audibility thresholds and whether there is an audible difference between two components. In this experiment X was assigned randomly either to have artificial reflections or not, whereas A and B were the “with reflections” or “without reflections” comparisons, assigned randomly.

The listening panel consisted of eight experienced listeners who had taken part in training experiments with the original stimuli as part of the research project described in [14]. After each listener had completed the test, some informal questions were asked to acquire some indication of what perceivable differences they had experienced, if any. Comparisons were made for each musical instrument at each of the three simulated source distances. Listeners were asked to familiarize themselves with the differences in source distance between sounds by playing each sound as many times as necessary prior to commencing the test.

4.6 Results of Listening Experiment

The results of the A/B/X test are shown in Fig. 25. In order to have reasonable confidence in the results being other than chance, it is common to look for a proportion of correct identifications above 75%. It is clear from the results that a very high proportion of subjects detected the difference in the sound field of the room with reflections on and off. It was interesting to note that the highest percentage of correct results was for sounds with a pronounced transient nature and with a mainly mid- and high-frequency spectrum; in this case for cornet and trumpet.

The guitar sounds, which had a long sustain relative to the other two sounds, yielded slightly less promising results for detection, although still above the threshold. This suggests that the early reflection detection threshold is related to the spectral and dynamic profile of the stimuli. Similar findings were also reported by Olive and Toole [6] and Bech [9].

Informal comments suggested that 85% of the subjects mainly experienced the change as one of source width or source focus. According to most listeners, the source width increased and the source focus decreased in the presence of early reflections. Some listeners (20%) experienced changes in the environment width, timbral differences, and changes in loudness.

4.7 Discussion

In the context of the pilot listening experiment, which was carried out using three-channel stereo (frontal image plane), it is clear that the listeners did perceive a difference in the spatial characteristics of the reproduced stimuli with and without the early reflections. The results of the informal survey confirm that the listeners experienced changes in source width and source focus in the presence of relatively low-level reflections (maximum amplitude of −7 dB with reference to the direct sound and maximum delay of 12.5 ms relative to the source stimuli). It is interesting to compare the reflection distribution of the simulated distance stimuli used as source material with that of the artificial room reflection. As shown in Fig. 24, all significant and discrete reflections in the recorded source stimuli came after the first 15 ms with reference to the direct sound, and all artificially created room reflections were simulated within the first 15 ms, as listed in Table 1. Therefore no interaction between the two sets of reflections was expected.

It should also be noted that the experimental setup was a symmetrical one where the properties of the reflections coming from the left-hand side of the listener were mirrored to the right-hand side in all planes. Therefore it is no surprise that the perceived effects of these reflections resulted in a broadening of the source and the environment. If, however, the reflection arrangement had been unsymmetrical, then an image shift would probably have been perceived toward the strongest reflection plane or hemisphere.

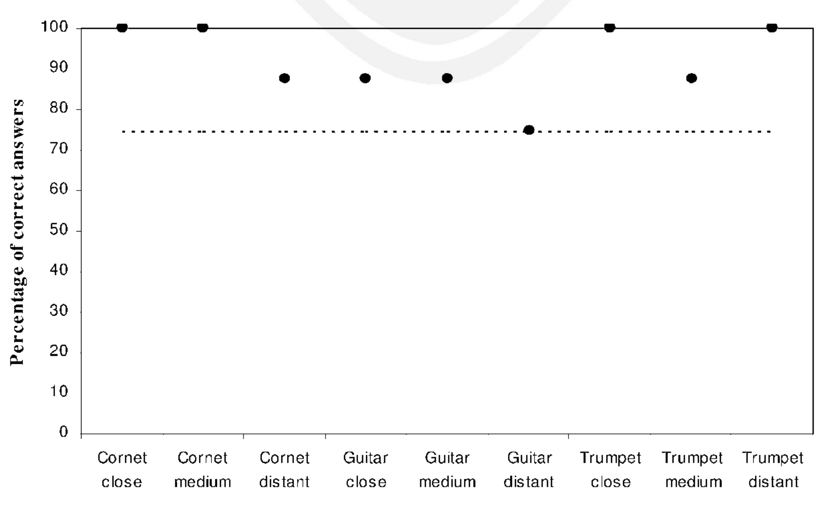


Fig. 25. Results of A/B/X listening test (reflections on/off); percentage of correct answers plotted against stimuli.

5 CONCLUSION

Simulations undertaken in a computer-based acoustic design package enabled the accurate modeling of the physical arrangement that was subsequently used to control reflections in the physical setup. The measured results in the physical setup turned out to be surprisingly good and within the perceptual limits required to create a reflection-free zone. The use of active flat panels that also acted as deflectors was found to be a valid way of modifying the early acoustical characteristics of a listening room in such a way as to enable the simulation of one type of listening room within another while preserving the low-frequency modal response of the original room.

It was established that the proposed simulation setup is capable of emulating a different but known early reflection pattern in controlled listening conditions without having any noticeable effect on the remaining acoustical conditions of the room. The findings of the pilot experiment were found to be in close agreement with the findings of Olive and Toole's [6] and Bech's [9] experiments in terms of the threshold of image shifts in IEC standard listening rooms.

6 REFERENCES

[1] ITU-R Rec. BS.1116, "Methods for the Subjective Assessment of Small Impairments in Audio Systems, Including Multi-Channel Sound Systems," International Telecommunications Union, Geneva, Switzerland (1998).

[2] IEC Rec. 268-13, "Sound System Equipment, Part 13: Listening Tests on Loudspeakers," International Electrotechnical Commission, Geneva, Switzerland (1985).

[3] EBU Rec. R22, "Acoustical Properties of Control Rooms and Listening Rooms for Assessment of Broadcast Programmes," European Broadcast Union, Geneva, Switzerland (1985).

[4] OIRT Rec. 86/3, "Technical Parameters of OIRT Reference Listening Rooms," OIRT Doc. TK-29-116, Prague, Czech Republic (1990).

[5] Nordic Rec. N 12-A, "Sound Control Rooms and Listening Rooms," Nordic Broadcast Corporations (1992).

[6] S. E. Olive and F. E. Toole, "The Detection of Reflections in Typical Rooms," presented at the 85th Convention of the Audio Engineering Society, *J. Audio Eng. Soc.* (*Abstracts*), vol. 36, p. 1029 (1988 Dec.), preprint 2719.

[7] S. Bech, "Timbral Aspect of Reproduced Sound in Small Rooms, I," *J. Acoust. Soc. Am.,* vol. 97, pp. 1717–1726 (1995 Mar.).

[8] S. Bech, "Timbral Aspect of Reproduced Sound in Small Rooms, II," *J. Acoust. Soc. Am.,* vol. 99, pp. 3539–3549 (1996 June).

[9] S. Bech, "Spatial Aspect of Reproduced Sound in Small Rooms," *J. Acoust. Soc. Am.,* vol. 103, pp. 434–445 (1998 Jan.).

[10] R. Walker, "A New Approach to the Design of Control Room Acoustics for Stereophony," presented at the 94th Convention of the Audio Engineering Society, *J. Audio Eng. Soc.* (*Abstracts*), vol. 41, p. 400 (1993 May), preprint 3543.

[11] A. Naqvi and F. Rumsey, "The Active Listening Room Simulator: Part 2," presented at the 111th Convention of the Audio Engineering Society, *J. Audio Eng. Soc.* (*Abstracts*), vol. 49, p. 1217 (2001 Dec.), convention paper 5425.

[12] R. Walker, "Early Reflections in Studio Control Rooms: The Results from the First Controlled Image Design Installations," presented at the 96th Convention of the Audio Engineering Society, *J. Audio Eng. Soc.* (*Abstracts*), vol. 42, p. 408 (1994 May), preprint 3853.

[13] J. Vanderkooy and S. P. Lipshitz, "Uses and Abuses of the Energy–Time Curve," *J. Audio Eng. Soc.,* vol. 38, pp. 819–836 (1990 Nov.).

[14] T. Neher, F. Rumsey, and T. Brookes, "Training of Listeners for the Evaluation of Spatial Sound Reproduction," presented at the 112th Convention of the Audio Engineering Society, *J. Audio Eng. Soc.* (*Abstracts*), vol. 50, pp. 518, 519 (2002 June), convention paper 5584.

[15] V. Hansen and G. Munch, "Making Recordings for Simulation Tests in the Archimedes Project," *J. Audio Eng. Soc.* (*Engineering Reports*), vol. 39, pp. 768–774 (1991 Oct.).

[16] D. L. Clark, "Ten Years of A/B/X Testing," presented at the 91st Convention of the Audio Engineering Society, *J. Audio Eng. Soc.* (*Abstracts*), vol. 39, p. 1003 (1991 Dec.), preprint 3167.

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