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The Active Listening Room: Part 3 A Subjective Analysis

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ABSTRACT
This paper presents the results of computer simulation and the actual room measurements of reflection patterns from the active reflector arrangement in a reference listening room which will be used to create artificial reflections in a five speaker, surround listening configuration. This formulates the third and final phase of design experiments relating to the panel arrangement to create a perceptually reflection free zone involving the analysis of computer modelling and room measurement results. Results of a pilot listening test using a well defined stimuli with artificial reflections generated by the DML loudspeaker arrangements are also presented.

INTRODUCTION
In the preceding publications about this project [1] [2], the key aims of the project were clearly listed. Here is a brief summary. Traditionally, experiments in the field of simulation of sound fields are carried out in anechoic chambers where 95% of the source sound energy is absorbed and is considered a completely inert acoustic environment. 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 rooms, has its own acoustic characteristics, which makes it subjectively quite different from any other. Although complying with a given standard, the sound field of a reference listening room is far from being considered acoustically inert. This variation in the subjective and objective domains is the basis of the active listening room where the key acoustic features such as the reverberation time, early decay time and the early reflection pattern can be varied during specific listening tests to subjectively assess the effect of change in listening conditions on the results of the tests.

The experiments were set up in a ITU-R BS1116 [3] specification listening room at the University of Surrey. The approximate internal dimensions of the room were 7.35 x 5.33 x 2.50m. Internal room finishes were carpet on floor, lay-in grid tile absorbent ceiling and full range acoustic absorber boxes on the walls. The measured reverberation time of this room was 0.245sec at 1KHz and was found to be within the specified reverberation time window for upper and lower limits in all relevant 1/3 octave bands. The measured ambient background noise level with the ventilation system and technical power switched corresponded to NR12. The chosen source loudspeakers were an active integral amplifier, full range infinite baffle, wide dispersion, medium size studio monitor with an operational bandwidth of 60Hz – 18KHz +/-3dB. The loudspeakers were mounted on a speaker stand and the centre of the cabinet was 1.25m from the floor. The deflector panels around the source loudspeaker were angled in such a way that any sound hitting these panels was forced away from the listening position. The physical size of the panels determines the wavelength of sound waves which can be reflected, therefore the lower frequency sound waves with greater wavelength compared to the panel size will not be deflected by this arrangement. The lower frequency cut-off limit was calculated to be approx. 500Hz.

The DML panels were chosen because they are rigid flat panels, which can be used as deflectors to deflect sound from the source speaker away from the listener. Also, they have wide dispersion characteristics and their on-axis and off-axis responses are favourable to create artificial reflections in an angular panel arrangement.

COMPUTER MODELING

In this section the computer modelling results of a five channel source loudspeaker involving sixteen DML panel arrangement is discussed. The experimental DML panel arrangement set-up for a five channel surround configuration, was based on a computer model of the panel arrangement around the five speaker i.e. L,C,R LS and RS source loudspeaker, positioned as shown in figure below. The angular panel settings was optimised to create a geometric boundary setting which forces the early reflections to be directed away from the listening / measuring position. The computer model of this experiment was created in the commercially available software CATT Acoustic®. The main purposes of creating the model were:

1 - To optimise the position of the experiment set-up in terms of angles and positions of the panels to create a reflection free zone at the listener / measurement position.

2 - To predict the reflection patterns of the set-up as a whole, and from the panel arrangement in particular, within the first 50ms time window.

The model was constructed for a closed room with internal elements as floating objects. The absorption coefficients of walls ceiling and floor were adjusted to get a close match between the predicted and measured values of reverberation time. The directivity / dispersion of the source loudspeaker was accurately modelled. The positions of the source loudspeaker and the deflector panels were adjusted for maximum deflection of early reflections away from the listening / measurement position.

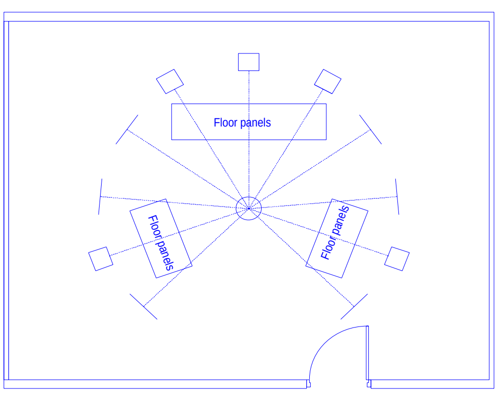

Figure 1
Ray tracing diagram of panel arrangement

Before creating the acoustic model an ideal reflection path model was sketched in which the locations of the DML panels was derived from spherical boundary created around the listening position from which artificial reflection can be created. Also, this basic model was useful in determining and identifying the main paths of significant reflections from the source speakers in a typical, completely symmetrical, listening arrangement. The deflector panels were then arranged in such a way to deflect sound away from the listener position to create a perceptual reflection free zone. As the selected panel size was 600x600mm, a reflection free zone with a low frequency cut-off of 500Hz was expected. The predicted early reflection pattern is shown in the CATT Acoustic ®prediction plots below.

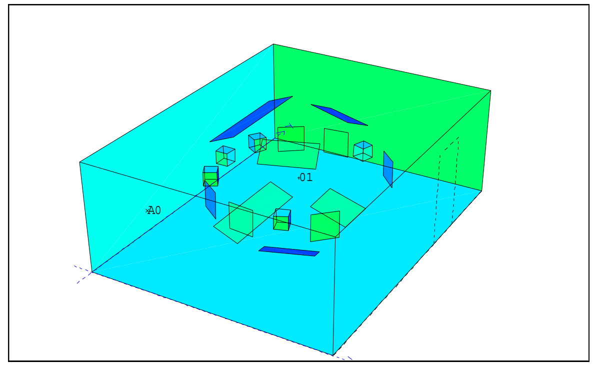

Figure 2
3D view plot of the panel arrangement in LR1
A0 = left speaker 01 = receiver

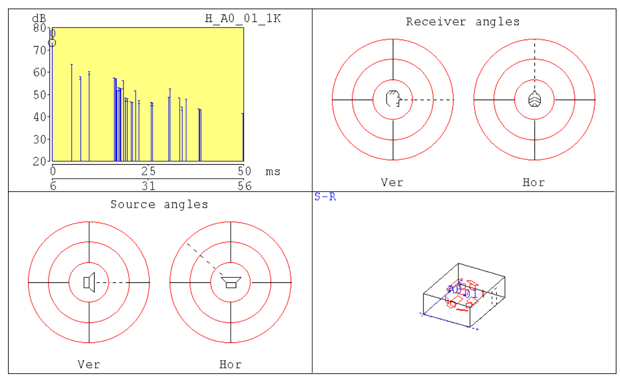

Figure 3
CATT Acoustic® model results, path and amplitude of first order reflection

Figure 3 shows the time amplitude and path display of the most significant, first order reflection from centre speaker within the panel arrangement arriving at the listener position. This plot shows that the amplitude of the second, first order, reflections and there on is below the 14dB relative to the direct sound. We can ignore the first reflection, as shown above, as the predicted path of this reflection is from behind the source loudspeaker. It should be noted that the above plot is based on broad band spectrum of reflections. Our aim to create a perceptually reflection free zone approaching semi anechoic conditions between 500Hz and 20kHz within the first 22ms. Therefore, it is predicted that the level of early reflections around the listening position within our panel arrangement would be around –20dB ref. direct sound. This is proved in the following sections where the actual room measurements of our panel arrangements are discussed.

ROOM MEASUREMENTS

For the purposes of this experiment it is assumed that a measurement system with well-defined Fourier-Transform windows with time resolution of 1–2 ms and a frequency resolution of about 500Hz will be adequate to measure the room acoustic responses. These are the main frequencies bands which we are interested in as they give rise to the directional information which might be disturbed by early reflections[4]. All room measurements presented here are 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 is widely used in the spectral analysis of reflected sound.

Figure 4
Picture showing the source loudspeakers and the DML panels for each source

Below are the MLSSA ETF of the five source loudspeaker within the deflector panel arrangement. All plot show amplitude and time distribution of early reflection patterns within the panel arrangement up to 30ms. It is clear that the amplitude of the early reflections is below –20dB ref. direct sound above 1kHz and
–15dB ref. direct sound above 500Hz, in each case.

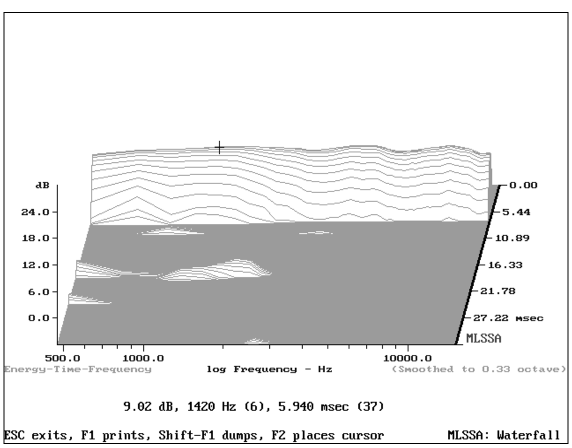

Figure 5
ETF plot of centre loudspeaker and the reflected energy at the listener position within the five channel panel arrangement

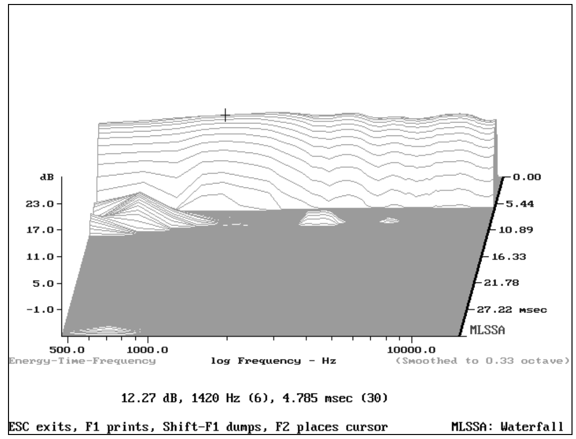

Figure 6
ETF plot of centre left loudspeaker and the reflected energy at the listener position within the five channel panel arrangement

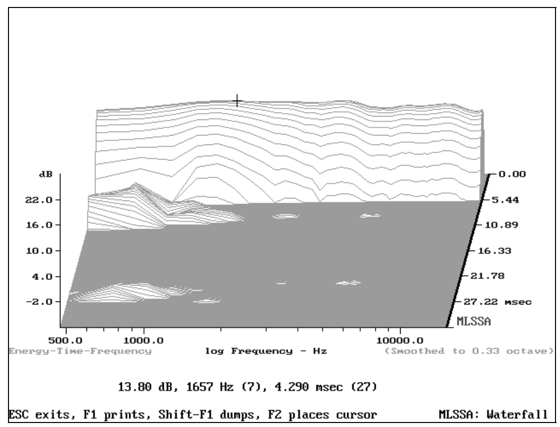

Figure 7
ETF plot of right loudspeaker and the reflected energy at the listener position within the five channel panel arrangement

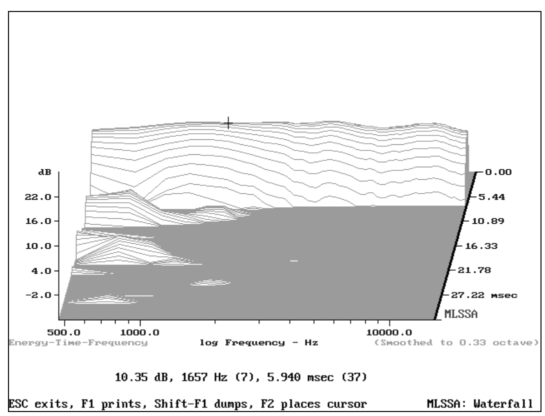

Figure 8
ETF plot of left surround loudspeaker and the reflected energy at the listener position within the five channel panel arrangement

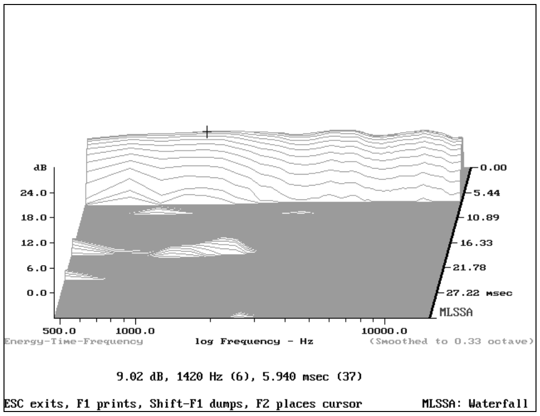

Figure 9
ETF plot of right surround loudspeaker and the reflected energy at the listener position within the five channel panel arrangement

The above plots show that a fairly uniform reflection free zone, approaching semi anechoic conditions (-20dB ref direct sound) is achieved by the DML panels acting as deflector panels for each source loudspeaker. It was considered that this arrangement was fit to be used for the second stage, which involved excitation of these panel to generate artificial reflection and simulate the early and late part of a different, but known, sound field within this listening room. The following section will discuss the signal processing required to achieve the above objectives.

SIGNAL PROCESSING FOR ARTIFICAL REFLECTIONS

Below is a complete generic block diagram of the source signal routing and processor capable of generating 16 individual reflections by means of the DML panels with exclusive controls for each reflection parameters such as 24 band parametric equalizer, delay and attenuation. There is a down mixing stage to generate the late part i.e. control for varying the reverberation in the room. The reverberation time is varied by four conventional loudspeakers located in each corner of the room.

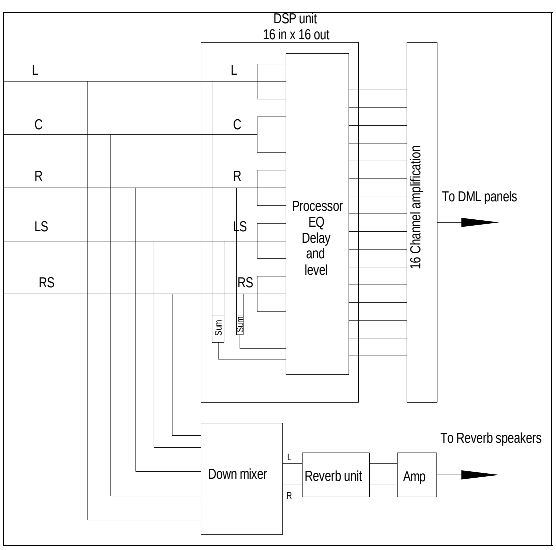

Figure 10
Block diagram of signal and processor routing

At this stage, our efforts are concentrated in only generating a set of early reflections with our panel arrangement and this processor layout, corresponding to a known room for its early reflection patterns. In the second stage, the late reverberation part will be implemented.

In the following section, selection and implementation of a set of early reflections corresponding to modelled room is discussed.

IMPLEMENTATION OF ARTIFICIAL REFLECTIONS

One of the specific aim of this project is to design a listening area where the acoustic parameters can be varied to simulate a slightly different listening conditions for specific listening tests. This is in recognition of the fact that there are slight differences in the various standards such as the ITU-R BS 1116, IEC 268-13, EBU R22, OIRT recommendations and the Nordic recommendations. For this reason, it was considered appropriate, as a starting point, to implement a set of early reflection pattern which corresponded to a room built to one of these standards other than the ITU-R BS 1116 (as our experiment room is built to this standard) and conduct some listening experiments to subjectively appraise the differences.

For initial the experiments, a set of 12 early reflections from a room built to an IEC 268-13, data acquired from [5], was implemented in the following manner.

Reflection no.	Location Of DML panel	Attenuation ref. source speaker	Delay Ref. source speaker	Azimuth / Elevation [degrees]
1	Centre floor	7 dB	1.5 ms	0 / - 42
2	Centre Ceiling	9 dB	3.5 ms	0 / 42
3	Left floor	7 dB	1.5 ms	30 / - 42
4	Left Ceiling	9 dB	3.5 ms	30 / 42
5	Left side front	12 dB	7.5 ms	55 / 0
6	Left side back	12 dB	9 ms	85 / 0
7	Left rear	10 dB	12.5 ms	135 / 0
8	Right floor	7 dB	1.5 ms	- 30 / - 42
9	Right ceiling	9 dB	3.5 ms	- 30 / 42
10	Right side front	12 dB	7.5 ms	-55 / 0
11	Right side back	12 dB	9 ms	-85 / 0
12	Right rear	10 dB	12.5 ms	-135 / 0

** Left hand side of listener defines positive angle*

Table 1
Properties of individual reflection

As described in detail in [2], it was considered necessary to investigate into the spectral properties of real early reflections in listening rooms before setting up the simulator for listening tests and also evaluate the performance of the DML panels to establish their suitability in terms of timbral properties. It was apparent from the DML panel laboratory measurements [2] that they will require careful equalization to match the spectrum of this given reflection.

A 24 band parametric equaliser, see figure 10 for processor components, was employed to achieve the spectral profile of each reflection based on published data [5] for a IEC 286-13 type listening room. Figure 11, below shows the time domain and spectral distribution of these reflections in an ETF plot.

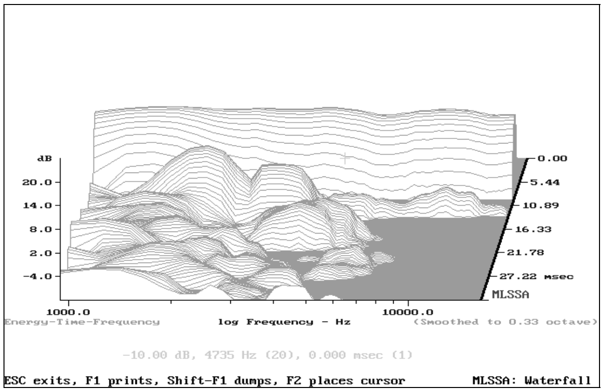

Figure 11
ETF plot of simulated early reflections ref to centre loudspeaker

STIMULI FOR PILOT LISTENING TEST

It was considered necessary, for the purposes of a pilot listening test, to chose a stimuli which had some inherently well defined composition of early and late reflection. For this reason a stimuli was chosen which is designed to be used to demonstrate the perceptual effect of changes in source distance to listeners and varies along the distance dimension in a uni dimensional manner [6].

For the synthesis of the sounds, a Lexicon® 480L reverberation processor was used in conjunction with the 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 [6]. A generic impulse response was designed consisting of three separate time regions, i.e.:

1. Direct Sound: t = 0ms
2. Early reflections: 15ms < t < 40ms
3. Reverberation tail: t > 40ms

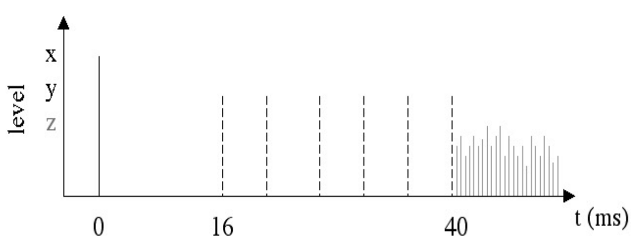

Figure 12
Generic impulse response of the reverberator used for distance processing [6]. The six early reflections are shown in dashed lines and the reverb tail in grey.

The source material was taken from the Archimedes CD and comprised a cornet, trumpet, male voice and acoustic guitar. All stimuli were mixed for reproduction over Left (L), Centre (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 the above figure) between the three chosen time regions[6].

For the purposes of a pilot listening test, three sound samples of varying source distance (extreme close, intermediate and extreme distant) were chosen for three instruments namely Cornet, Guitar and trumpet.

As the chosen stimuli required only front three loudspeakers (L,C,R) for the complete reproduction of the stimuli, four out of the twelve early reflections i.e. left and right back and rear, as listed in table 1, were derived from the left and right loudspeaker inputs accordingly.

The following sections describes the listening experiment design.

PILOT LISTENING TEST

The pilot listening test was based on the ABX type detection experiment methodology. The object of this experiment was to investigate whether the artificial early reflection simulation had an perceivable effect. It was considered appropriate to first establish if the listeners could perceive “any difference” in the sound from the source loudspeakers, with and without the early reflections. Depending on the results of this basic experiment, the next stage experiments will investigate into “what” perceivable differences are experienced, in detail.

The listening test was set up with two screens. On screen 1, there were three sets of source stimuli, each set containing three versions of the same sound with different source distance - extreme close, intermediate and extreme distant. Listeners were asked to familiarise themselves with the differences in source distance between sound by playing each sound as many times as necessary, before commencing the test.

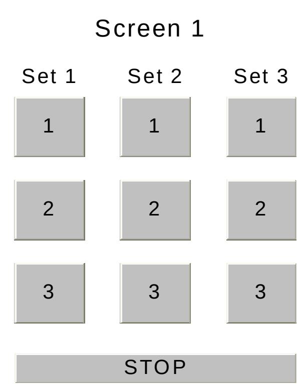

Figure 13
Image of screen 1 with three sets of sounds with varying source distance

On screen two, there were three buttons (Ref, A, B) corresponding to each sound on screen 1. Listeners were asked to play a sound from screen 1 and compare any perceived difference between Ref, A and B by clicking the corresponding buttons on screen 2. The early reflection settings (ON or OFF) for Ref was identical to A or B. They were asked to mark if Ref was identical to A or B for each sound.

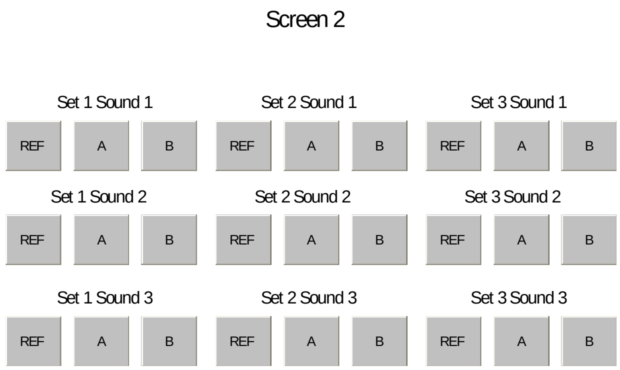

Figure 14
Image of screen 2 with three buttons Ref, A and B (reflection on / off) for each sound on screen 1

The listener panel comprised eight experienced listeners who had taken part in training experiments with this stimuli as part of a research project mentioned in [6].

After each listener had completed the test, some informal questions were asked to acquire some indication of what perceivable difference they had experienced, if any.

RESULTS OF THE LISTENING TEST

The results of the ABX detection test are shown in figure 15 with the line of significant difference from chance plotted at shown at 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 are for sounds with high degree of attack component with a mainly mid and high frequency spectrum; for e.g. Cornet and Trumpet.

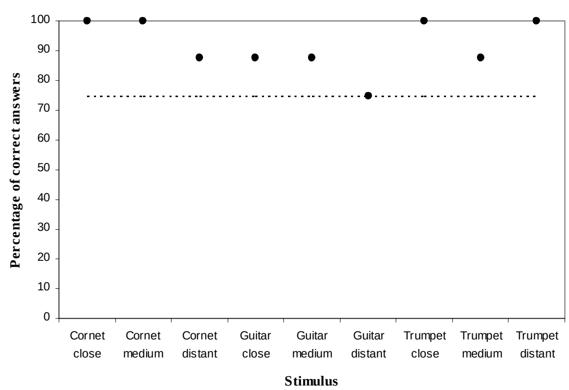

Figure 15
Results of ABX listening test (reflections on / off), percentage of correct answers plotted against stimuli

The Guitar sounds, which had a long sustain relative to the other two sounds, yielded relatively poor results for detection of reflections which indicated that the early reflection detection threshold is related to spectral and dynamic profile of the stimuli. This was also reported by Olive and Toole [4].

The informal comments suggests that 85% of subjects mainly experienced the change in source width or source focus. According to most listener's, the source width increased with less focus in the presence of early reflections. Some listeners (20%) experienced changes in the environment width, timbral differences and changes in loudness.

FURTHER RESEARCH

The next stage of experiments will involve the implementation of the late part i.e. variation of reverberation in conjunction with the early reflections which will complete the synthesis of the artificial sound field in the listening room. Experiments will be designed to look into the effects of room related changes on source width, depth and distance in multi-channel reproduction systems along with spatial effects such as envelopment and spaciousness.

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