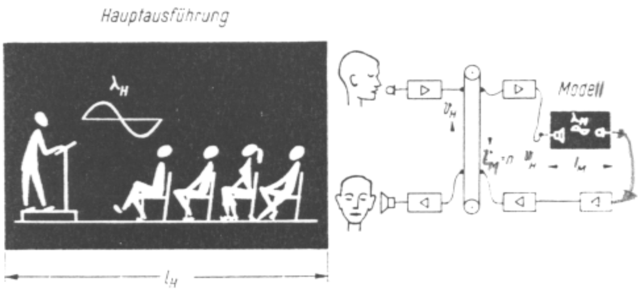

„Das Tonband läuft zehnmal so schnell
Für Schallaufnahmen im Modell;
Um die Akustik zu erfassen,
Muß man es langsam laufen lassen!"[1]
(F. Spandöck)

Auralisation of a Scale Model of the Royal Albert Hall

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Abstract

Following the law of acoustical similarity, the improvement of existing and design of new enclosures have been studied since Spandöck by means of scale model measurements.
Auralisation, making measurements audible, became possible by measuring impulse responses in model frequency range (e.g. 600 Hz-92 kHz, scale 1:12).
Main problem is absorption by oxygen and water molecules leading to energy dissipation increasing with time and frequency. Instead of using a closed cavity filled with nitrogen or dried air the correction of model impulse responses can be done by computer equipment.
Here, a time- and frequency varying filter, computed in MATLAB, supplies equalisation of the speaker and model dummy head, compensation of high frequency attenuation and correction of the decay for noise.
The auralisation of the corrected and compensated binaural impulse responses is done by convolution with anechoic music recordings and special HRTF filters, using the difference of a special real scale dummy head and the test person to get individual frequency correction spectra for each listener.
The application to a scale model of the Royal Albert Hall in different stage and reflector setups to reduce the famous echo further and to make improvements audible is presented with examples of convolved music.

1. Introduction

The quotation above of the pioneer of room acoustic model measurements F. Spandöck leads directly to the problems of this subject: the modelling of sound fields in enclosure scale models has its base in the laws of acoustical similarity [1]:

$$\frac{\lambda_M}{l_M} = \frac{\lambda_R}{l_R}; M = \frac{\lambda_M}{\lambda_R} = \frac{l_M}{l_R}; f_M = \frac{1}{M}\frac{c_R}{c_M} f_R \; . \tag{1}$$

The correct modeling of halls and refraction phenomena requires the wavelengths in the Model $\boldsymbol{\lambda_M}$ to have the same relation to the dimensions of the model $\boldsymbol{l_M}$ as the wavelength $\boldsymbol{\lambda_R}$ to the dimension $\boldsymbol{l_R}$ in the real enclosure. The interesting frequency range for models is therefore by the scale factor higher than in reality: in real halls the range from 50 Hz to 8 kHz is the most important, in 1 by 12 scale model like that of the Royal Albert Hall the interesting frequency range is 600 Hz to 196 kHz. This indicates the need to use loudspeakers and microphones that are capable to operate in ultrasonic frequency range.
If the walls in the model have similar absorption properties for the corresponding frequency range like the real walls in the real frequency range, the sound fields in the model should be similar to the sound fields in the real hall in the limit of how good the details are modeled – that is the idea.
A major problem in model measurements and auralisation is the fact that the in the filling gas air the oxygen molecules are influenced in their relaxation behavior by water molecules. This

leads to a dissipation of energy increasing with traveling time and frequency which is highly nonlinear. The dissipation coefficient $\boldsymbol{\delta}$ is a function of frequency **f**, temperature $\boldsymbol{\vartheta}$ and relative humidity $\boldsymbol{\varphi}$ of the air and can empirically be estimated to the rule-of-thumb [1]:

$$\delta\ (\vartheta,\varphi,f) \approx \frac{85}{\varphi\ /\%}\left(\frac{f}{kHz}\right)^2 \cdot 10^{-4} \cdot \frac{1}{m}, \vartheta\ = 20°C\ . \qquad (2)$$

Former solution for smaller models was the use of Nitrogen as filling gas, which does not show this high frequency attenuation effect – but filling a 5 m long and 2.5 m high 1 by 12 scale model like the one of the Royal Albert Hall would require an even bigger anechoic vacuum vessel of about 50 m^3 volume which had to be filled with Nitrogen!

To overcome this rather uneconomic solution formerly also a drying up of normal air was applied, which was also effective because dry air without water molecules behaves much less absorbing; but this method suffered from the long drying times before it was possible to measure [1]. But this method, too, was only practicable for small models.

The aim of these methods was to get a valid modeling of the sound-field of the hall in the scale model. For room acoustic purposes reverberation times and ETCs of the first hundreds of milliseconds may be sufficient with the use of correction formulas, but for auralisation valid impulse responses have to be measured.

2. Model Measurements – Equipment and procedure

The first step in auralisation of model impulse responses is the measurement of valid and highly repeatable model room impulses. Validation measurements were made in the real Royal Albert Hall. Then impulse responses were measured in the 1:12 scale model of the RAH of the company Peutz nearby Nijmegen at the same positions as before in the real hall:

Figure 1: 1:12 Scale Model of the Royal Albert Hall

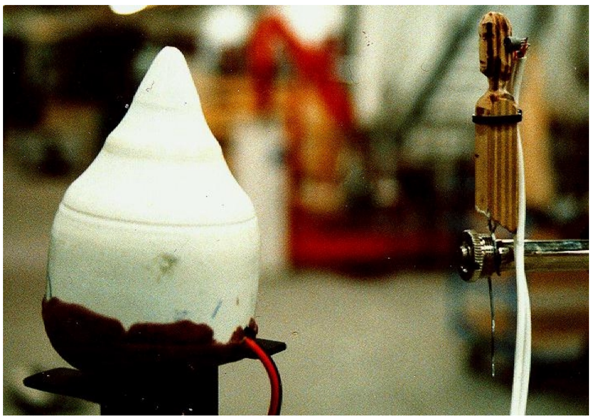


Figure 2: Model Source and Dummy Head

The impulse responses were measured using the maximum-length-sequence based program WINMLS 2.0 by the SEKD Prodif 96 Pro card with a MLS sequence order of 16 at a sample frequency of 96 kHz. This fact restricted the frequency range of the result to 4 kcps maximum. This seems to be low, but is perhaps enough for a realistic impression.

The microphones used were Monacor MCD 2500 capsules for the monaural, and the model dummy head with built-in Sennheiser KE 4-211-2 capsules for the binaural impulse responses. The used sound source was a Motorola piezo-effect high frequency loudspeaker, which was covered by a special developed symmetrical cone out of hard foam material. The source and receivers are the critical part in this system, as the calibration measurements will show below, due to the limited and spiky frequency responses. On a scale 1:12 even more than in reality all possible things are obstacles in the sound field so that refraction may occur anywhere. Furthermore special microphones and power amplifiers were used to ensure a flat frequency response up to 48 kHz.

3. Auralisation of Model Impulse Responses

The task was to find a way to correct the more than proportional air absorption of high frequencies in measured impulse responses of a model of the famous Royal Albert Hall in London by use of a computer. The way to do this was to implement a module in the programming platform MATLAB, which solved all the parts of this problem in one:
to correct the frequency responses of the combination of source and microphone used,
to compensate for the air absorption of the model impulse responses, to remove the amplified noise background, and to correct the tail of the responses.
The aim was to convolve these corrected impulse responses with music recorded in an anechoic room and to compare the results to the original convolutions of the Royal Albert Hall measurements regarding their frequency-limited bandwidth.

2.1 Air Absorption in Scale Models – Ways of Solving

Measurements in room models filled with air generally suffer from high-frequency-absorption which is much higher than the law of acoustical similarity predicts. By traveling through the medium air, sound waves are damped by a frequency- and traveling-time-dependent attenuation due to dissipation of their energy, following the Law of Lambert-Beer:

$$I(t)=I_0 \bullet \exp(-\delta t). \qquad (3)$$

For the special conditions of 293K temperature, normal pressure of p=1013 H Pa and a relative humidity of 55% the following figure 3 (taken from [2]) shows the enormous effect on to high frequencies of sound in air and the need to correct model room impulse measurements for this effect, which seems to be rather small, but turns out to be in the range of dozens of dB in a several m large model and several hundred ms traveling times. Apart from the expensive filling of a scale model with nitrogen one can use the capabilities of modern computing power to calculate the reduction in the amount of absorption and to correct the measurements by applying digital audio processing.
The basic principles of this process were developed and described by [2].

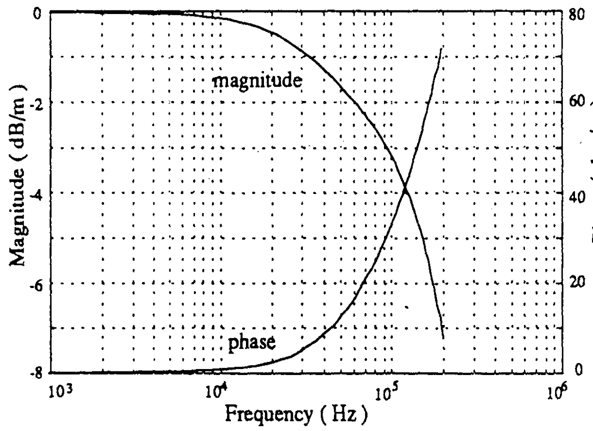


Figure 3: Magnitude and Phase of δ (from [2])

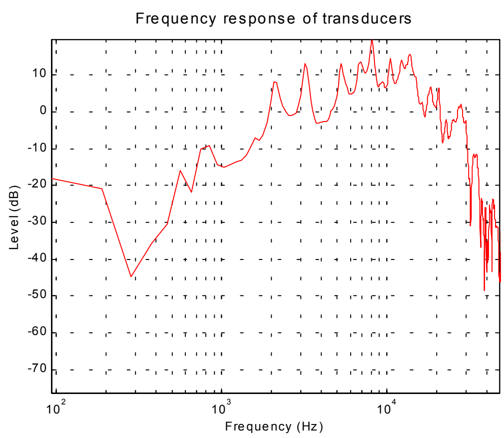


Figure 2: Frequency Response of Model Transducers

2.1.1 Problem 1: Equalisation of Source and Microphone

For the fact, that both source and microphone work in the region of ultrasonic and the most important aim for their development was to reach the best non-directivity possible, they do not have a flat frequency response, and for good measurements, this has to be as flat as possible without spoiling the time-response.

So using the efforts of the digital domain we built a module to equalise the frequency response of a given pair of speaker and microphone by means of a given free-field reference measurement, say h_{ref}. Unfortunately the direct inverse filtering of the reference measurement - which would mean to calculate the discrete Fourier transform:

$$H_{ref}=DFT\ (h_{ref})\ ; \qquad (4)$$

and calculating $1/H_{ref}$ - leads to a general unstable filter for the transfer function; H_{ref} is not of minimum phase and the time sequence:

$$h_{inv}=IFT\ (1/H_{ref})\ ; \qquad (5)$$

is not necessarily a causal filter any more and the convolution $h_{rev} * h_{inv}$ may not result in the unit sample response $\delta\ (t)$ (see [3]). As a correction filter we used the normalized conjugate complex function of the transfer function, like it is described in [4]: while the convolution of h_{rev} and h_{inv} may not result in a stable filter, the least-squares criterion is applied:

$$<\{[h_{filter} * (h_{model} * h_{ref}+h_{noise})-h_{model}]^2\}> \ , \qquad (6)$$

i. e. the ensemble average over all possible realizations in the model has to be minimum for a given length of the applied filter. The optimum filter satisfying eq. (6) is:

$$F(w)=\frac{W^*(w)}{[|W(w)|^2+R_{n,n}(w)]}, \qquad (7)$$

where the superscript * denotes the complex conjugate and $R_{n,n}$ the autocorrelation function of h_{noise}, which is assumed to be constant over frequencies, i.e. broad band noise. The auto-correlation of broad band noise has only a contribution near dc, so it may be neglected here.

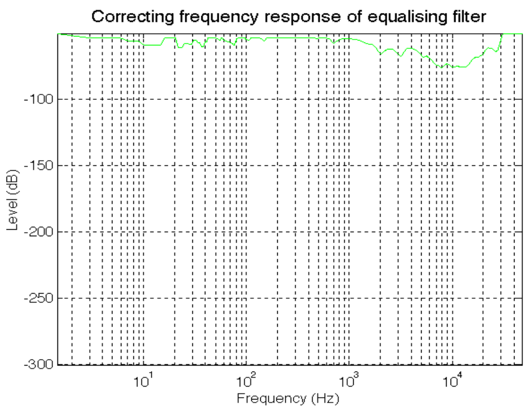


Figure 5: Magnitude of Equalising Filter

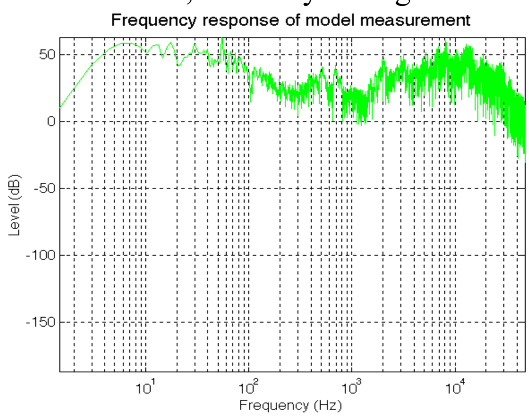
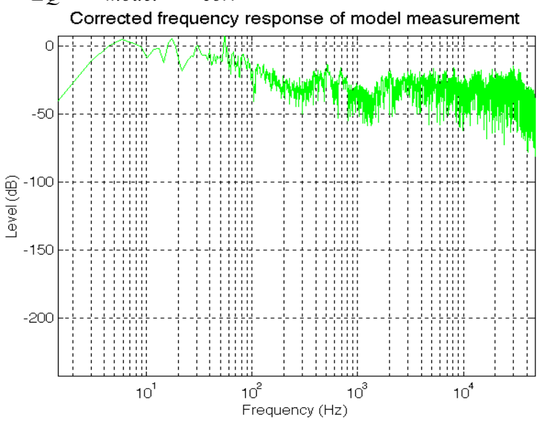


Figure 6: Frequency Response of Model Measurement

This filter makes a nearly perfect equalisation of the source to microphone measurement. Unfortunately this free-field reference measurement itself suffers from several errors due to the dimensions and shaping of the transducers: In all the reference free field measurements very narrow and sharp dips in the frequency curve occur due to interference cancellation. If these dips – which are due to measurement artifacts and not to the transfer functions of the transducers - are compensated perfectly, like this least-squares filter it provides, any noise that exist in the measurement of the model will be amplified in the narrow dips; by the correction filter these turn into sharp high peaks. Amplifying broad band noise with a filter containing frequency peaks will lead to narrow band noise of high amplitudes, and these narrow noise bands are audible as *tones* and especially in high frequencies, as a constant ringing. So this effect had to be suppressed by making the frequency properties of the correcting filter less accurate again especially in the higher frequencies above 1 kHz. This was done by applying an fractional-octave averaging to the magnitudes of the filter to smooth it out. Afterwards the filter-FFT was recomposed using the unchanged phases. Now the filter was ready to work as a correction filter for the model impulse responses measured with the same microphone-speaker combination to give a frequency-corrected impulse response:

$$h_{EQ} = h_{model} * h_{corr}. \tag{8}$$



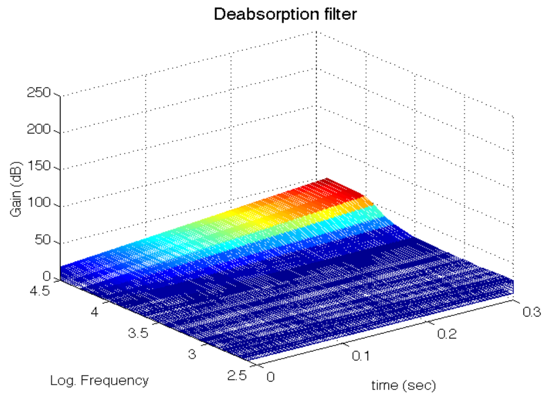
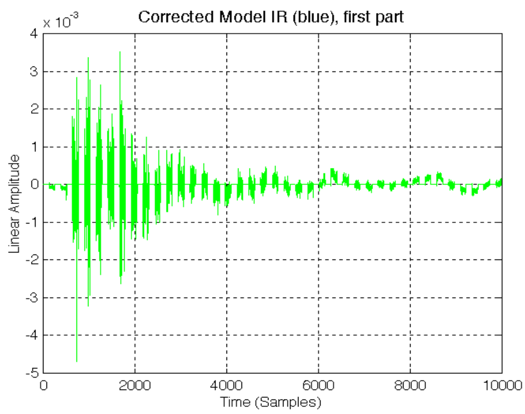


Figure 7: Frequency Resp. of Equalised Model Measurement **Figure 8**: Deabsorption Filter

2.1.2 Problem 2: Compensation of the Sound Attenuation in Air

The solution is basically to build up a time- and frequency-varying filter which amplifies the room model impulse responses linearly in dB and even more the higher the frequency according to the values of the damping coefficient **δ** mentioned before. Such a filter shape is depicted in figure (8). We applied an overlapping series of time-windows to the measured sample sequence and filter the windowed time sample sections with stepwise changing coefficients calculated out of the damping values. In a last step the overlapping filtered time windows are added back together to get the absorption-corrected model impulse response. The corrected room response h_{EQ} was split up into two shifted series of boxcar-windowed time slices:



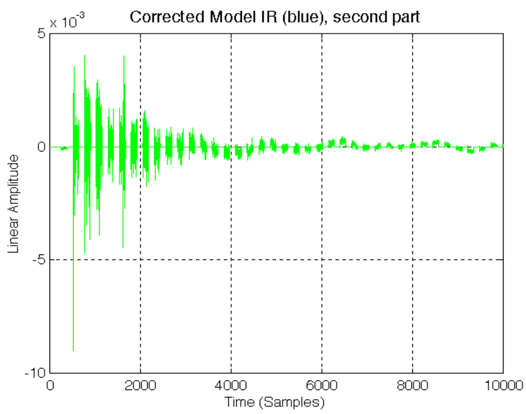


Figure 9,10: Boxcar windowed Time Slices of Corrected Model Impulse Response, 1st and 2nd Series

which were long enough to provide the lowest needed frequency, in real world the corresponding largest wavelength had to fit into the length of one slice.

Then we took the values for the absorption-correcting amplification [5] and made tables out of it ready to apply in filtering later on and selectable in 10 % steps of relative humidity of air at room temperature – the example shape shown in figure (8) is e.g. valid for 60 % r. H.

Then the final gain of the filter and the gains for the different slices in between have to be calculated by the time in milliseconds the sound is traveling, for the absorption in the table given as coefficients of the dimension Level/time are in dB per second, minus the amount of damping in high frequencies is normal in real scale. The gain of the filter has to be interpolated for the different frequencies out of the edge-values of the tables mentioned above. So these values are estimated by a linear stepwise interpolation per octave frequency. One further transforms the coefficient value in dB/s to the multiplication factor to apply to calculate linear amplitude values. The single time slices of the two series are now time-windowed by multiplying with a hanning window, which has the very nice property, that an overlap of half the window length and a subsequent adding of the time slices reveals the total energy unchanged. The calculated values of $\boldsymbol{\delta}$ serve as coefficients to build up a FIR-filter which provides the wanted slice-wise changing amplitude gain values to the recent frequencies. The result now is achieved by filtering the windowed time parts of the two sequences h_{win} with the filters h_{fil} to get $h_{filtered}$:

$$h_{filtered} = h_{fil} * h_{win}. \quad (9)$$

While the high frequency absorption is linear per time and distance in level, the filter turns out to of be highly exponential shape. After the filtering process the filtered time window series are added together to get the complete time data again. This time data is now corrected for the high frequency absorption in the useful data parts, but unfortunately every high frequency noise content is also amplified; and nevertheless, how small the amount may be, after an amplification of e.g. 60 already dB it sooner or later becomes audible – and visible. And this gives a chance for the next step of the process.

2.1.3 Problem 3: Correction of the Decay for the Background Noise

The model room impulse response, which is now corrected for the air absorption effect, now suffers from the limited signal to noise ratio of the measurement, for all noise in the higher frequency-bands which is in the original measurement, will be amplified by the absorption correction filter with some dozen of dB per second, i.e. linear amplitude factors of 10^5 up to 10^{25}. Like the following figure shows, the result is that the tail of the impulse response, that should go down with the now absorption corrected slope in each frequency band turns its direction after a certain time and starts to grow in magnitude again. To get reasonable results which can be convolved with music recorded in an anechoic chamber in order to auralise the model and changes in it, this growing part of the decay curve has to be corrected somehow in order to get the right slope of decay. Due to the fact, that it is only the noise of the measurement and no real sound information, this late part of the impulse response could be discarded and replaced by a signal that has the desired decay slope and no disturbing effects. We took this tail as it results from the process and treated it in a different way: after an octave band filtering in every octave band the turning point of the logarithmic envelope is detected by the supposition, that this turning point is near to the local minimum level of the smoothed envelope. Therefore the octave bands are low-pass filtered with a first-order butterworth-filter, which does the job perfectly if a fitting cut-off frequency is chosen, e.g. 17 ms for the 63 cps octave. In the smoothed logarithmic level-time curve the local minimum is chosen as the turning point after that a complete de-trending operation removes the growing trend of the octave band noise to a constant level. This constant tail is now multiplied with an octave depending factor in dB/s, which is derived from the real impulse response of the hall, so that the ‘right’ reverberation time is multiplied to the model octave bands. This, of course, could be replaced by the EDT-values out of the model decay. Then the octave bands are added together out of the beginning of the absorption corrected ones and the de-noised tail after the turning point, and in a last step the equalized, dissipation- and noise-corrected model impulse response is summed up from the different octave bands.

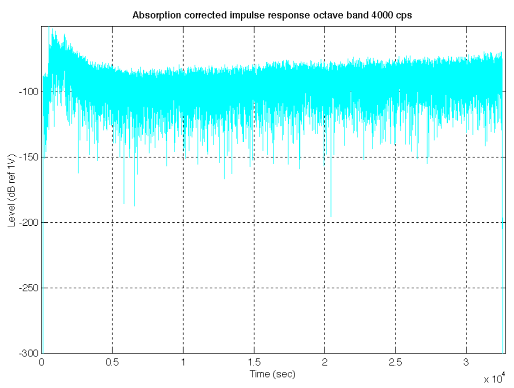


Figure 11: Absorption Corrected Octave Band

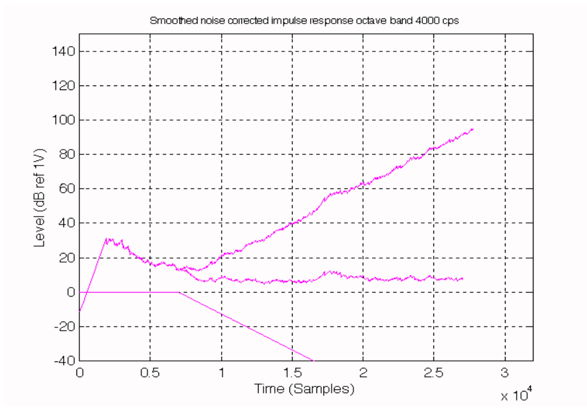


Figure 12: Tail Noise Correction Process

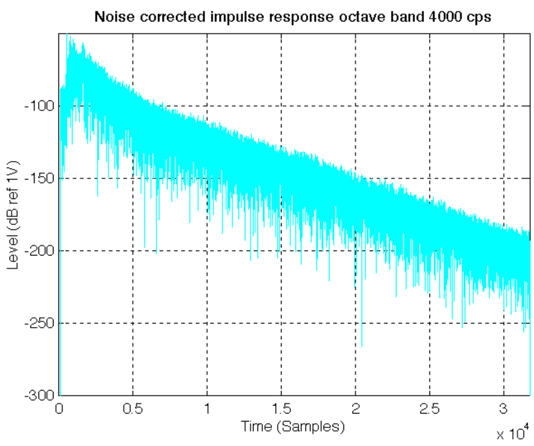


Figure 13: Absorption and Tail Corrected IR

Figure 14: New Dummy Head, Model Dummy Head

3. Building up personal HRTF correction filter

3.1 Difference Measurements of Two Transmission Chains

This test was to show that a compensation of the differences in frequency response and directionality patterns of the two system chains:
a) Dodecahedron source – enclosure – dummy head – amplifier – headphones – ear of listener; b) Dodecahedron source – enclosure – ear of listener
can be corrected for each another. For this test the Transmission test room at the Peutz laboratory was chosen with a T_{60} of about 3 sec. instead of the dry free-field laboratory, because the sound field in there can be assumed to be a sufficiently diffuse field one, and the correction should not depend on random dips in free field frequency response curves, which occur often due to interference or comb-filter effects. The desired correction was to correct the power spectra. The experimental setup was the following: The additional transmission system consisted of the dummy head, one Tascam DAT-tape deck DA-P1 working as microphone pre-amplifier and headphones amplifier, followed by a pair of Beyerdynamic DT 550 headphones. In the case with additional transmission system the impulse response of enclosure and system was measured at the test person's ear, in the second case only impulse response of the enclosure, now with test person at the same spot as the dummy head before. All measurements were done with the MLSSA system, with MLS of Fs 80 kHz, the sequence order and averages were 16 to enhance the S/N ratio by 12 dB.

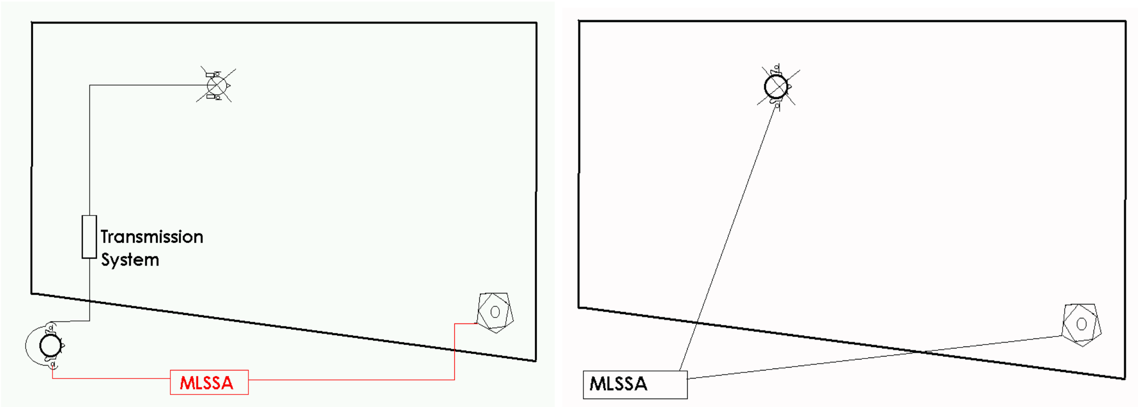


Figure 15: Different Experimental Setups for Measuring Transfer Functions of the Two System Chains

These MLS were amplified by the Quad 303 Amp and fed into the dodecahedron. The systems responses were measured at the ear of the listener by use of a Monacor MCE 2500 capsule, set into the pinnae right 1 cm in front of the entrance of the ear channel. Some experiments had to be made with the exact position and direction of this capsule, for if it was to near to the ear channel, deep dips occurred in the frequency response of that measurement, too far away from it (above 1.5 cm) strong reflections in the resulting impulse response could be detected. So there is an optimum at about ¾ cm distance to the entrance of the ear channel, exactly in the nice little bow which exists in the fine structure of the pinnae; this position also has the advantage that it can be used unchanged with headphones on the head. The capsule was taped directly to the skin facing upwards to ensure, that at least the high frequency directionality in the horizontal plane changes only due to the fine structure of the pinna, not due to the microphone properties.

3.2 New dummy head

A former used Sennheiser dummy head turned out to be too specific, like any human being's ear would be: the pinnae are much too detailed, so that the resulting directionally dependent frequency curves have very serious dips and peaks [6]. One can see: the measurement conditions for head related transfer functions (HRTFs) are very critical for such measurements; moving the microphone a little out of the optimal position will result in narrow and steep dips due to interference cancellations. The structure of the pinna and its effect on the fraction of sound due to the of any direction different dimensions allows it to locate sound sources even if the person is able only to hear with one ear, where level- and time differences are not important. We all are used to our own ears, we grew up with them and therefore it is a difficult task, to apply a specific personal HRTF to a sound due to the sharp dips in frequency response. This task comes even harder, if a correction is tried, where for the measurement a dummy head is used, which is almost as specific as any individual, because there these dips have to be corrected before one can apply the wanted HRTF. With the sharp peaks of 35 dB height any noise in signal or measurement will be a possible source of ringing sounds or narrow band noise, which is perceived as similar to tones. Another problem about this way of correction is, that in principle the "personal dips" of the dummy head have at first to be removed and than at other frequencies applied again. The aim was, to get a correcting FIR-filter with less than 500 points of length to be able to feed for presentations a real-time convolution machine. So it was decided not to take the specific dummy head and instead of the new unspecific one with a smooth frequency curve and about the same directional characteristics in lower and middle frequency bands as a natural head. In 1:12 a dummy head can not be too detailed. So the used dummy head had to be modified to be similar to the model dummy head: the original, much too specific pinna imitations were replaced by halved cylinders of wood, 5.5 cm in diameter and 4.4 cm long - a very simple shape. The aim of the modification was not to simulate one specific head or not even an average head exactly, but to ensure localisation and perception of spaciousness in diffuse sound fields like concert halls. The dummy head now is not of a very specific shape apart from the slice structure of the head which should make effects in the range of 10 kHz corresponding to their dimension of about 1 cm in step size. But this frequency range is for the auralisation at least of model measurements not too important any more. In front of the cylindrical pieces the microphone capsules KE-4-211-2 have been placed with a impedance matching circuit in the head, facing towards the sides. The pieces of wood ensure the main functions of natural pinnae: to build a shield to damp the high frequencies falling in from behind and to emphasize that frequencies coming from the front, without adding too much details to the frequency response. This simple modification of the AKG dummy head showed quite remarkable results at once: when only listening by headphone to the uncorrected signals of this dummy head the in-head-localisation vanished and a front localization was perceptive although being a little elevated. The next figures show the 50° angle-of-incidence frequency responses and the directional pattern of the modified dummy head, the second pair of figures is a natural HRTF and pattern just for a comparison:

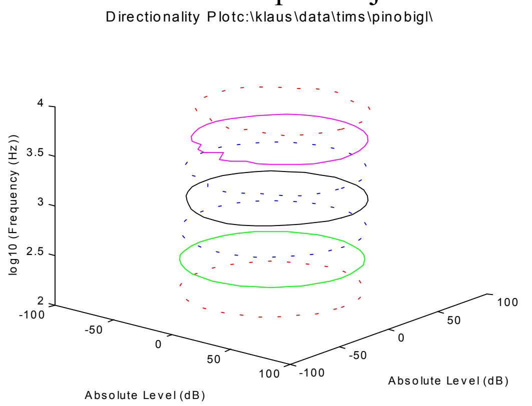


Figure 16: Directional Pattern of Dummy Head

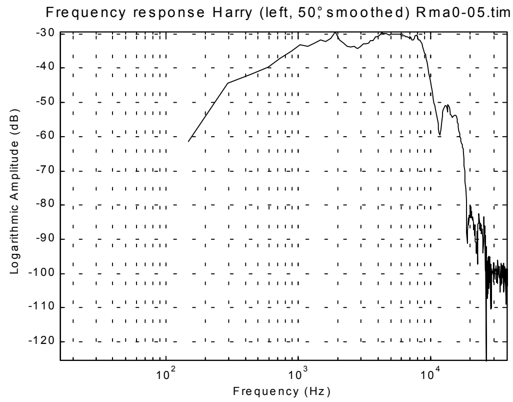


Figure 17: Frequency Response Dummy Head, 50º

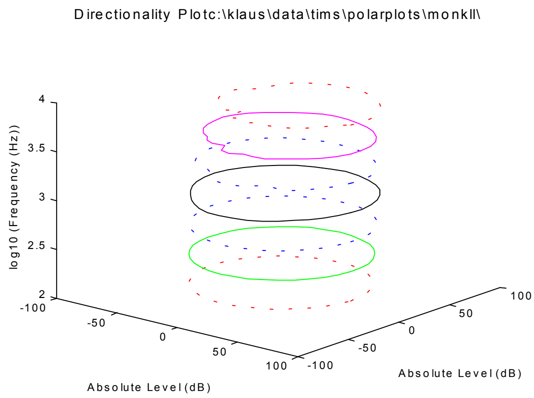


Figure 18: Directional Pattern of Natural Head

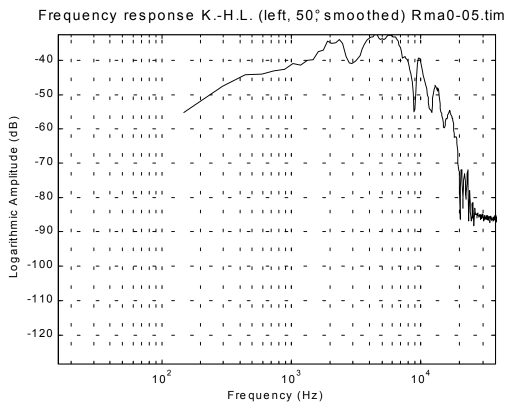


Figure 19: Frequency Response Natural Head, 50º

Obviously the signals recorded with this dummy head are much easier to filter to an individual HRTF than a complicated response. But it had to be proved, that the directional characteristics are also almost unaltered by this special dummy head. MLS measurements on the rotating desk were made: in 10 degree steps the impulse responses were measured and time-windowed, the frequency responses calculated by Fourier-transforms, the magnitudes averaged in every third-octave-band and the results plotted versus the angle of incidence.

Figure (15) shows the directionality patterns for the modified new dummy head. The averaged octave band directionality patterns show a reasonable good matching of the modified dummy head to the real head, better than the former used old Sennheiser head.

The transfer functions with and without additional transmission system consisting out of new dummy head, amplifying Tascam DA-P1 and headphones Beyerdynamic DT550 were measured for three individuals. With these data sets the calculation of the correcting filter, which corrects for an individual difference of hearing with additional transmission system and without, could be done in MATLAB. In the program the two different impulse responses are loaded in, Fourier-transformed, and the amplitude magnitudes averaged in fractional octaves. The octave fraction can be chosen, a sixth of an octave was judged to be of sufficient smoothing and accuracy. After that the two smoothed magnitude spectra were divided and this difference spectrum delivered the magnitudes for a set of logarithmically spaced frequencies. These frequencies divide half the Nyquist frequency into logarithmically equally spaced 500 frequency points, because the filter to be built should meet the requirement to stay with its length below 500 points, i.e. below 500 samples to make online convolution possible with very small latency time inside an available 48-channel convolution box. And the logarithmic distribution of frequency points off the axis is necessary because it fits as a first approximation the frequency resolution of the ear, which gives more weight to low frequencies. The perception of pitch is in a first approximation a logarithmic function of the frequency. In the next step the magnitude values of that frequency bins, which stand for the desired frequencies are extracted out of the differences of the frequency responses with additional transmission system and without. Together with the relative frequencies the amplitudes were used to built a finite impulse response (FIR)-filter in MATLAB.

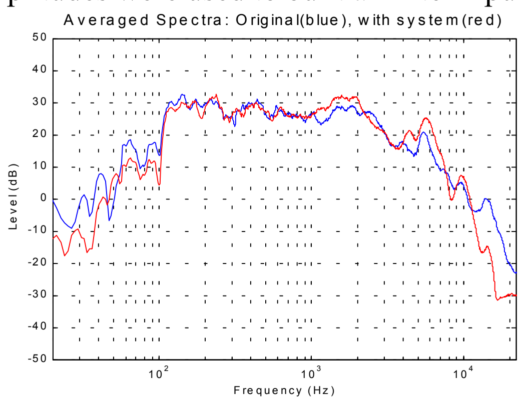


Figure 20: Averaged Spectra of the Two Systems

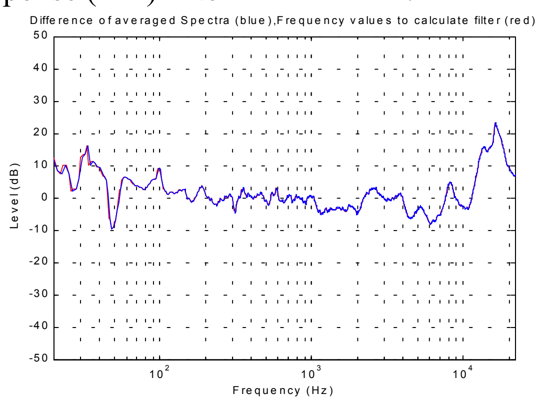


Figure 21: Difference Response for Calculation

The longer the impulse response of a filter, the more accurate is the result to low frequencies, but the longer the calculation time becomes, too, because if there are m samples in the signal and n samples in the filter, the calculation expense is m*n multiplications and additions.
These fir-filters are calculated for both ears of all test persons and tested in situ in the auralisation test in the real enclosure “De Vereeniging” Nijmegen showing that the test persons could hardly distinguish the sound in timbre between the original sound and that corrected one with additional transmission system.

3.3 Auralisation of a real hall in situ with personal HRTF filters

This was done in the real concert hall “De Vereeniging” in Nijmegen. For three experienced listeners dummy-head-headphone correction filters had been made for both ears as described. Different anechoic monaural samples of speech and music from CD were amplified by the SA 900 C amplifier and sent out by the dodecahedron loudspeaker placed on the stage to the enclosure. These signals were recorded by the Sennheiser capsules in the dummy head. Their signals were amplified by the Behringer Ultragain Mic 2002 by 32 dB and sent to a computer-controlled 48 channel convolution box which had the 6 personal correction filters pre-loaded. This convolution box convolves with minimum latency (therefore the filter length had to be reduced to 500 samples, i.e. 11 ms) these fir-filters and the input from the preamplifier. The resulting compensated signals were recorded by the Tascam DAT-tape deck DA-P1 and passed amplified through to the Beyerdynamic DT 550 headphones. For two listeners the samples were recorded on DAT with the system in between. The test-listeners compared in situ localisation, timbre, reverberation, spaciousness of their natural hearing directly besides and behind the dummy head to their personally compensated signals, as the picture below shows. The impression was of course not completely identical - but very close to the original in all of these properties. The difference was not in the timbre or reverberation nor even the localisation; only the spaciousness of some single reflections was different. But in total it can be stated that the smoothed frequency curve correction of the calculated filters works very well in fitting the personal impression of this hall. An in situ auralisation without almost no loss is possible with this way of compensating the transmission system.

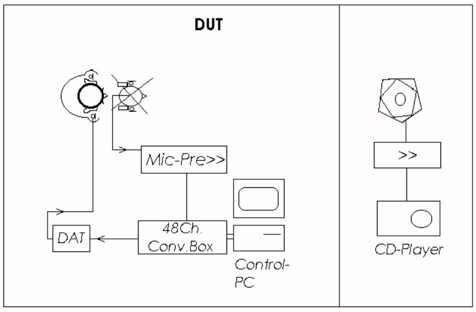
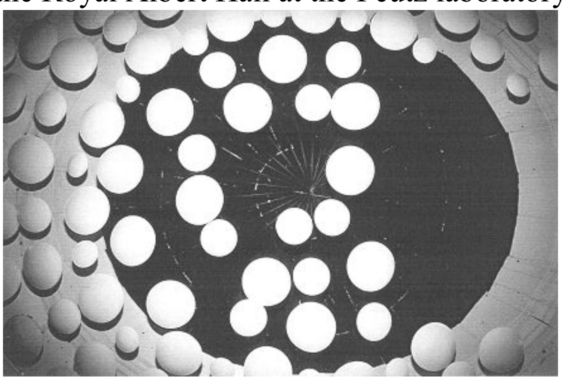


Figure 22: Experimental Setup for Auralisation in situ **Figure 23**: Auralisation in situ in Real Concert Hall

Conclusion: Results of the Model-Auralization and Limitations

Convolutions were made with music that was recorded without reverberation (Denon PG 6006) and convoluted with stereo room impulse responses measured in the 1:12 scale model of the Royal Albert Hall at the Peutz laboratory and processed in the way described above.



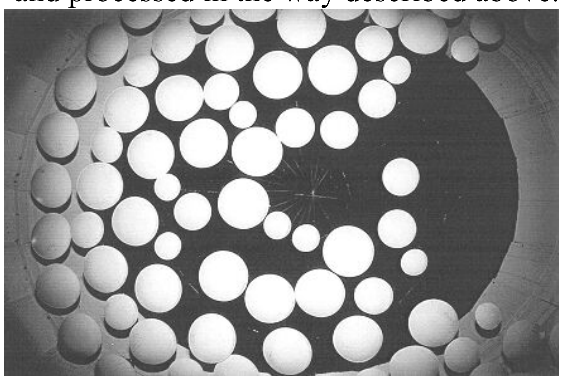


Figure 24: Old Setup of Mushrooms **Figure 25**: New Setup of Mushrooms

The convolutions revealed a relatively natural sounding picture of a reverberation and sound quality similar to the real hall. So this approach to solving the problems with these kind of measurements turned out to be a valid and successful tool in the auralization of model room impulse responses. It also can reveal, that the quality of auralization models and their materials, too, for there remains little difference between model and real impulse response in time response, and a more audible difference in frequency content or perceived timbre and spaciousness. To make this audible, impulse responses of the model in two different states to listen to the changes of a modified stage setup and a new arrangement of the famous "mushroom" reflectors. To make the difference visible, pseudo-three dimensional sonagram pictures are made from the pure, but band-limited impulse responses of the real hall, the unchanged and the corrected model measurements in the two states of the model:

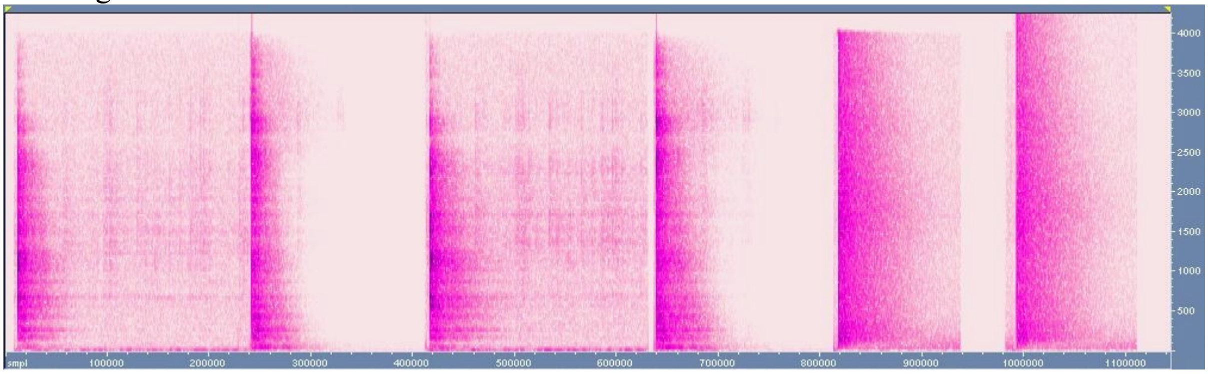


Figure (26): Model in old state (corrected), in new state (corrected), real hall bandlim., real hall

In further evaluations it has at first to be shown why the difference exists, e.g. because the model itself may suffer from some errors in the frequency-shifted absorption coefficients of the materials used in the model, which may behave additive in the audible differences and lead to too smaller reverberation times in the beginning of the reverberation process.

The limitations in our approach to improve are:

- limited frequency bandwidth up to 4000 Hz, which could be pushed up by other measurement equipment, which would make reference and dissipation correction harder;
- limited signal-to-noise ratio in the model room impulses, which could be improved by using MLS of a higher level, order and a larger number of additions,
- do the correction of air absorption and noise tail smaller time- and fractional frequency bands: e.g. third-octave bands would approach the ideal better and have less disturbing effects due to sharp edges in the level at the frequency band boundaries,
- replace the reference free-field measurement of the combination of loudspeaker and microphone by a diffuse-field reference measurement to cancel out comb-filter peaks.

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