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AURALIZATION TECHNIQUES USING BINAURAL IMPULSE RESPONSES FROM SCALE MODELS

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1. INTRODUCTION

Before discussing the technical side of auralization it is necessary to define what we can expect from auralization in the case of concert hall design.

We would like to hear the sound of the hall before it is built or altered. The general technique is to use binaural impulse responses, either calculated or measured to be convolved with anechoic music for speech.

Impulse responses are usually acquired, using one omnidirectional source. A real orchestra is neither one source or omnidirectional.

The "reality" we want to hear is therefore far from the real thing. This means that it is highly questionable that we will be able to judge many of the parameters in an absolute way, like orchestral balance, apparent source width, quality of string tone. Even definition or clarity will be different due to the difference between sound radiation of the point source and the musical instrument. Therefore auralization cannot be considered as an instrument for absolute judgement of a hall, in the optimal case one can judge the hall as if music was played over an omnidirectional loudspeaker on stage. One has to learn to listen to this type of music played in real halls to be able to judge auralized music from computer models or physical scale models.

However one can use auralization to make comparisons and if one wants to judge irregularities in impulse responses, like echoes or gaps in responses. In our work on the Royal Albert Hall's 1 : 12 scale model, where echo suppression and in some arrangements long time delays between (clusters of) reflections are of importance, auralization proved to be very useful to get an impression to which extent these irregularities were acceptable.

Because the shape of the Royal Albert Hall, where focusing originates from elliptical surfaces, a computer model based on geometrical acoustics could not be used for echo evaluation or auralization purposes because its inability to quantify echo strength. But using the responses from a scale model gives technical problems that have to be overcome to attain sufficient quality of the auralization signals.

These problems are:

- sufficiently "flat" response of the source/artificial head microphones to a reasonable upper frequency;
- good omnidirectionality of the source;
- 100% reproducibility of source (no sparks allowed);
- sufficient signal to noise ratio in the impulse responses up to at least the first 300-400 ms (real scale);
- elimination of effects of air absorption;
- getting RT's in all frequencies right, including controlled creation of reverberation tails from any chose point of the impulse response in order to get an usable S/N ratio in the auralized signal.

2. SOURCE AND RECEIVER

For reproducibility MLS or TDS type methods were chosen, which means that an electroacoustic transducer has to be used. Proper omnidirectionality of a source can only be achieved in practice if a very small source is used, in our case 4000 Hz at 1 : 12 scale means that the size of the source should be in the order of 1.5 mm. We make this by putting a slightly dampened reverse cone on a tweeter.

For reproducibility and omnidirectionality this is fine, the frequency (and phase) response are quite horrible.

The small microphones, used in the 13 mm wide artificial head are not as bad, but also start to give some frequency response problems above 35 kHz.

Therefore the response of the source combination is measured at a short distance. The inverse of this response is used as the filter for the measured impulse responses in the model to attain flat response, but at the cost of signal to noise ratio, which means that in the highest and lowest frequency range the signal to noise ratio drops, so the length of the useable part of the impulse response shortens in these frequency ranges (see figures 1-3).

3. IMPULSE RESPONSE PROCESSING

After inverse filtering to cancel the source/receiver imperfection, processing is necessary at the following points:

a. Compensation for air absorption (frequency dependant).
b. Correction of general slope (□ RT) due to the scale modell not being perfect. Our opinion is that scale models do not have to be exactly right for general reverberation, this can easily be corrected in the software. It is very time consuming and sometimes impossible to get the RT's right for all frequencies. Other tools, like Sabine, ray-tracing models and the like are more

suitable for reverberation time design. (The scale model can however sometimes help to get indications of the effectivity of absorption in slightly "hidden" places.)

c. Addition of reverberation tails where the (useable) length of the binaural impulse responses is too short (frequency dependant).
d. The possibility to shape the responses to reduce or increase certain parts of it to make it possible to find "thresholds" of audibility of echoes or gaps in the responses.

For a, b and c it is necessary to divide the impulse responses by filtering into (in our case octave) frequency bands in such a way, that in the case of no further processing the original impulse response is regained after recomposition (see lit. [1]). On the filtered impulse responses all the necessary processing is done, like:

- multiplication with $e^{\forall t}$ where $\forall$ is a parameter calculated from the compensation for air absorption and the ratio between the model's reverberation time in that frequency band and the reverberation time the real hall will have. It is clear that due to noise in the measurement the response will tend to start rising for higher values of t. Either by looking at it or by an algorithm the point t_c is found where the impulse response is replaced by an artificially generated "tail". The response is cut at t_c.

- This tail is constructed by filtered pink noise modulated in time by:

$$e^{-\beta(t-t_c)^2} \; for\, t < t_c$$
$$e^{-\frac{RT}{13.5}(t-t_c)} \; for\, t \geq t_c$$

The parameter $\exists$ is more or less arbitrary and defines the "rise time" of the tail.

The amplitude of the tail (which is important in the case that the useable time of the impulse response is short) is to be determined by least square regression from the peak – 5dB of the pulse response after the rise time of the hall.

- Shaping the impulse response to modulate echoes and gaps is done by applying a "filter" in time defined by a time parameter (where the peak or gap is maximally affected and a "Q" parameter to define the relative effect and duration. This tool turns out to be sufficiently effective to smoothly modify the curves to be able to judge the effects of reduction of echo strength or "filling" gaps.

After all this processing the octave band impulse response are added to produce the broad bandwidth impulse response to be used for convolution.

4. CONCLUSION

Although, as stated in the introduction, auralization by convolution of the binaural impulse response from a single, omnidirectional source to a dummy head, is a limited tool concerning the type of reality, it has proven to be a very effective tool to assess modifications in a designed or existing hall. In the case of the scale model of the Royal Albert Hall the proposed alterations of the upper part of the auditorium (dome, mushrooms) effects of a great number of possible changes could be better judged than looking to the impulse response only, and could be compared with the existing hall as the reference object.

The quality of the signals after the processing as described, is sufficient to judge the effect of alterations despite the limited bandwidth. It takes however some time to learn to listen to these signals. Further work is in preparation in order to do 1 : 1 scale auralizations of existing halls based on the same principle to improve the understanding of the acoustics heard by playing anechoic music over a single point source.

Literature:

1. Polak, J.D., X. Meynial and V. Grillon: Auralization in scale models: Processing of Impulse response.

RM/TSt130/RT101RM.DOC

Figure no. 1

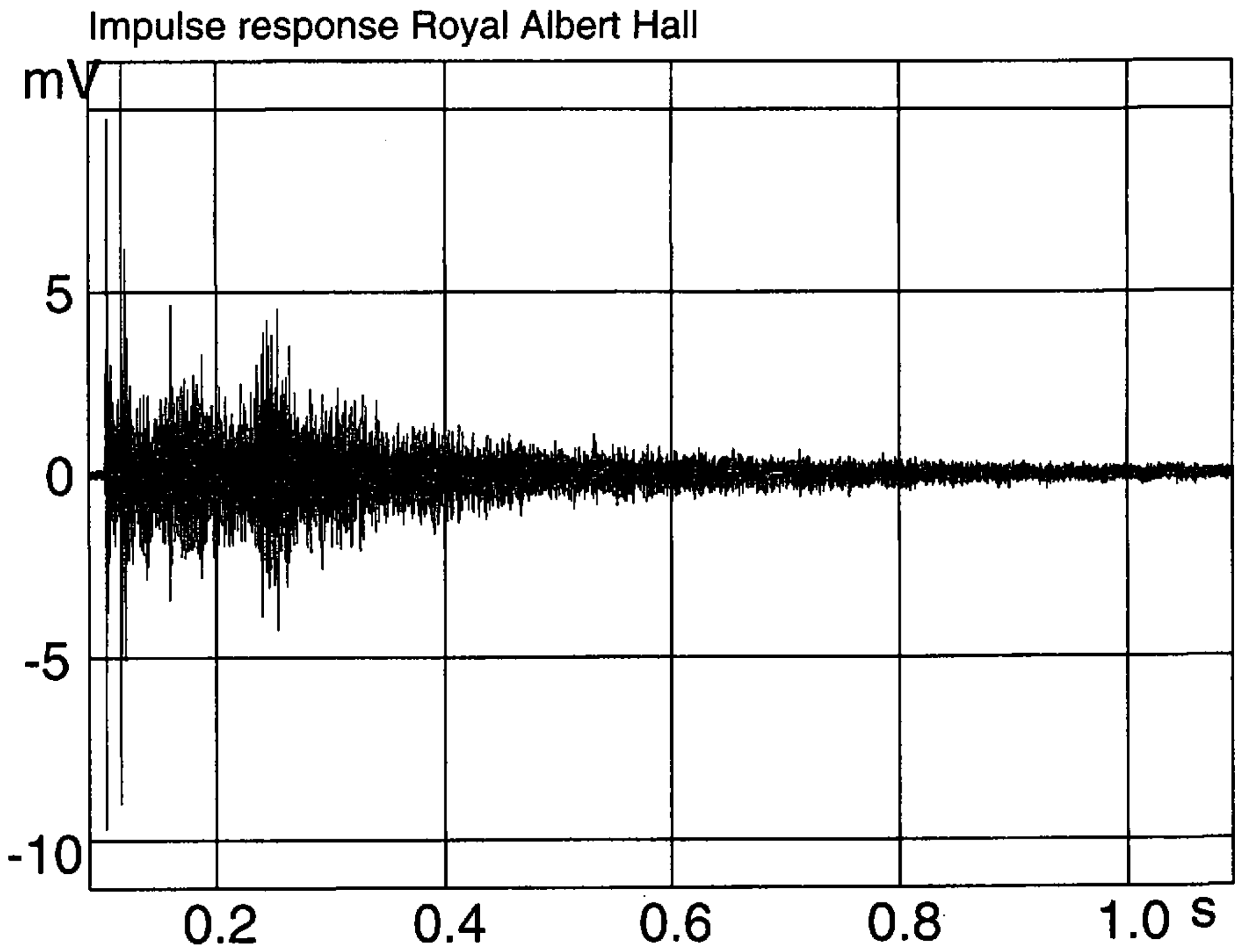


Figure no. 2

Air Absorption Correction Filter Shape (for 60 % relative Humidity)

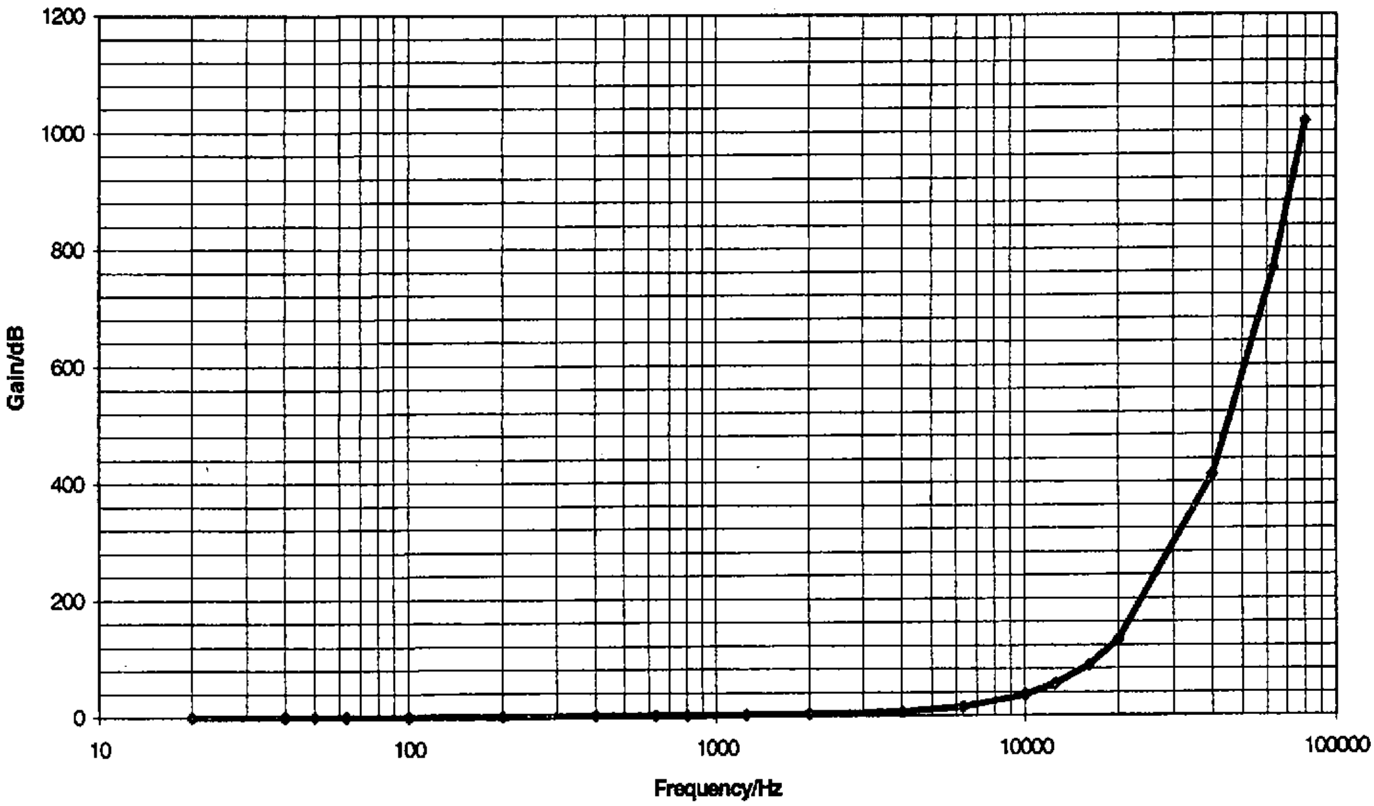


Figure no. 3a

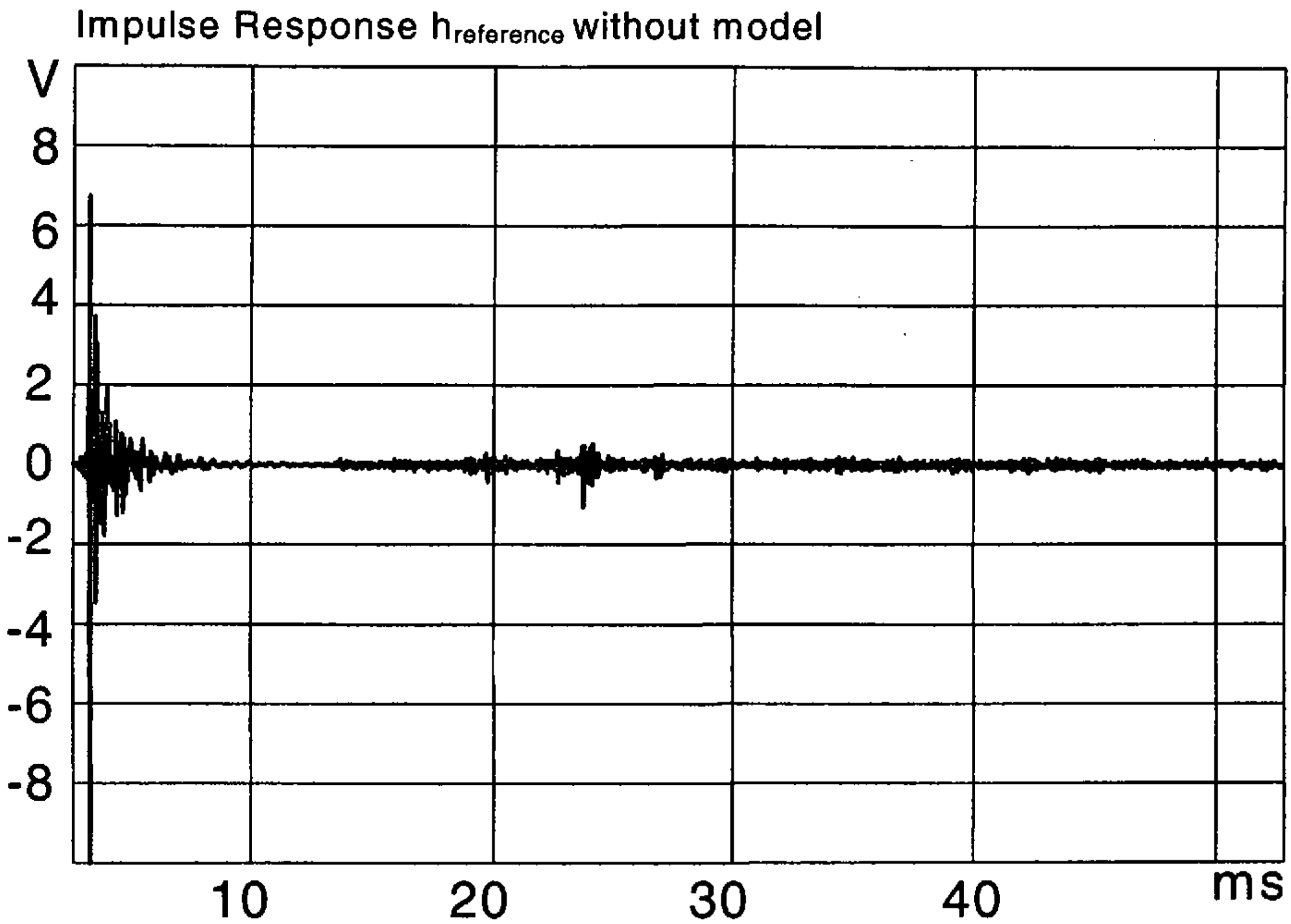

Impulse Response h reference without model
V
8
6
4
2
0
-2
-4
-6
-8
10
20
30
40
ms

Figure no. 3b

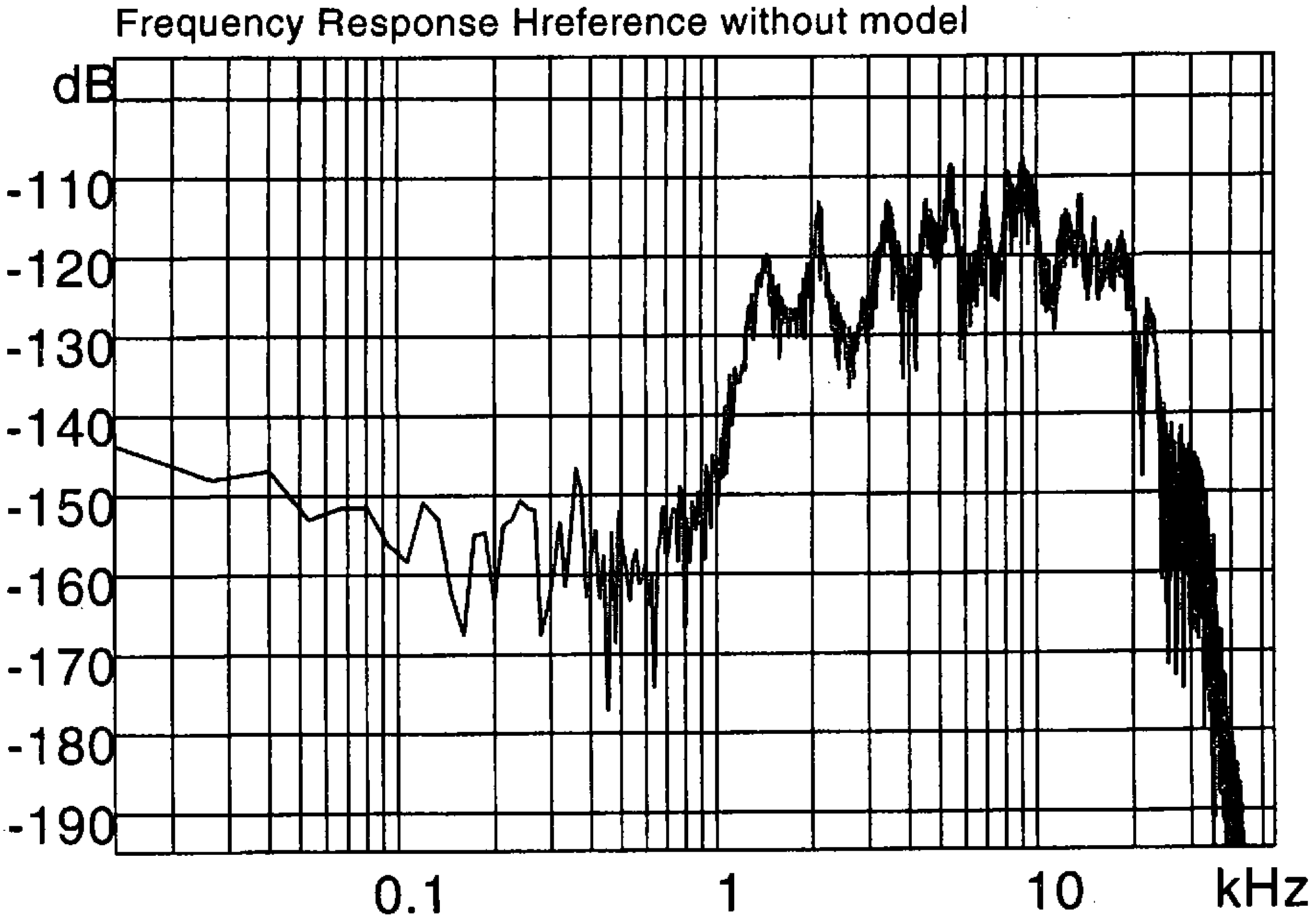

Frequency Response Hreference without model
dB
-110
-120
-130
-140
-150
-160
-170
-180
-190
0.1
1
10
kHz

Figure no. 3c

Frequency Response $H_{inverse,minimal\ phase}$ without model

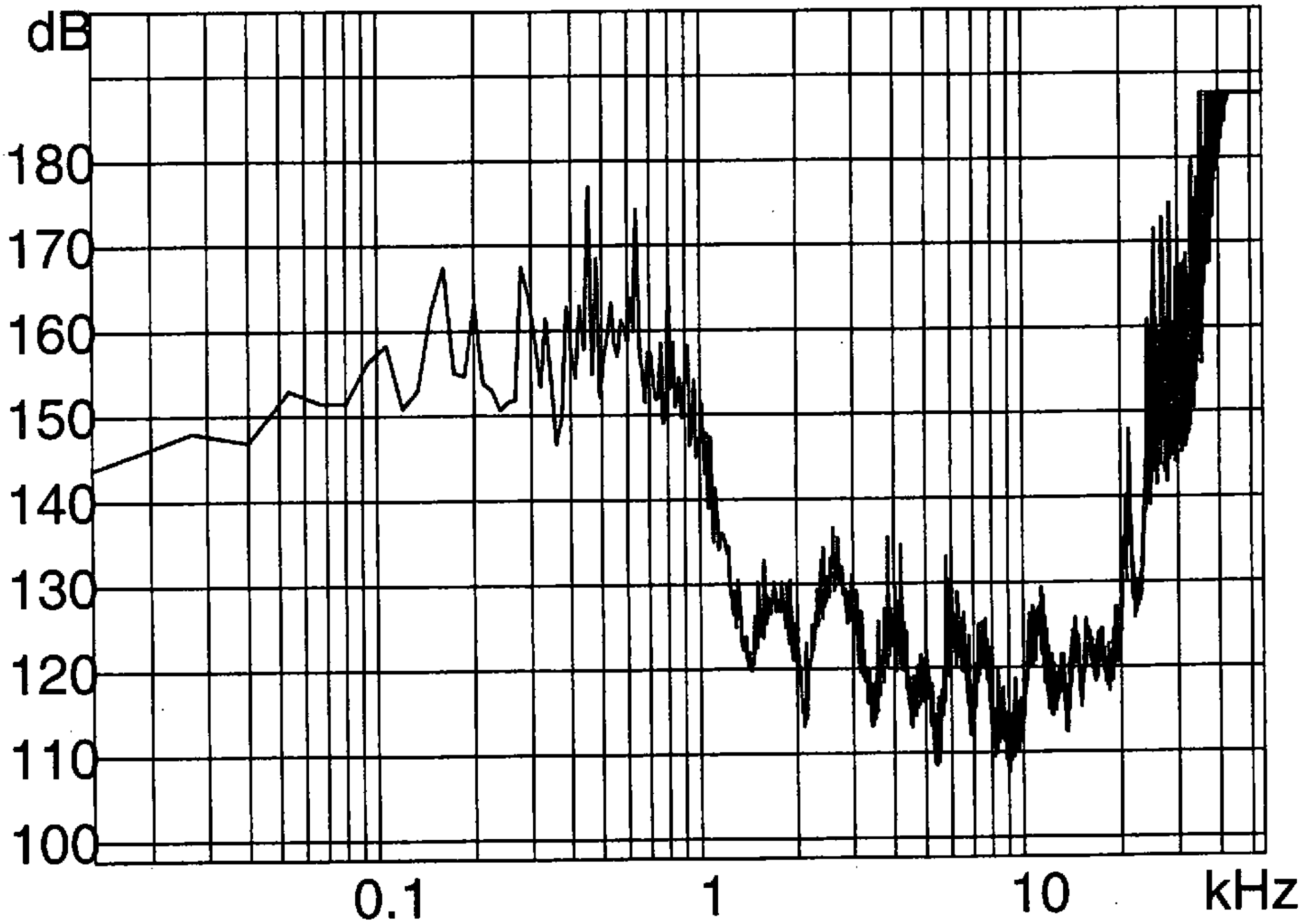
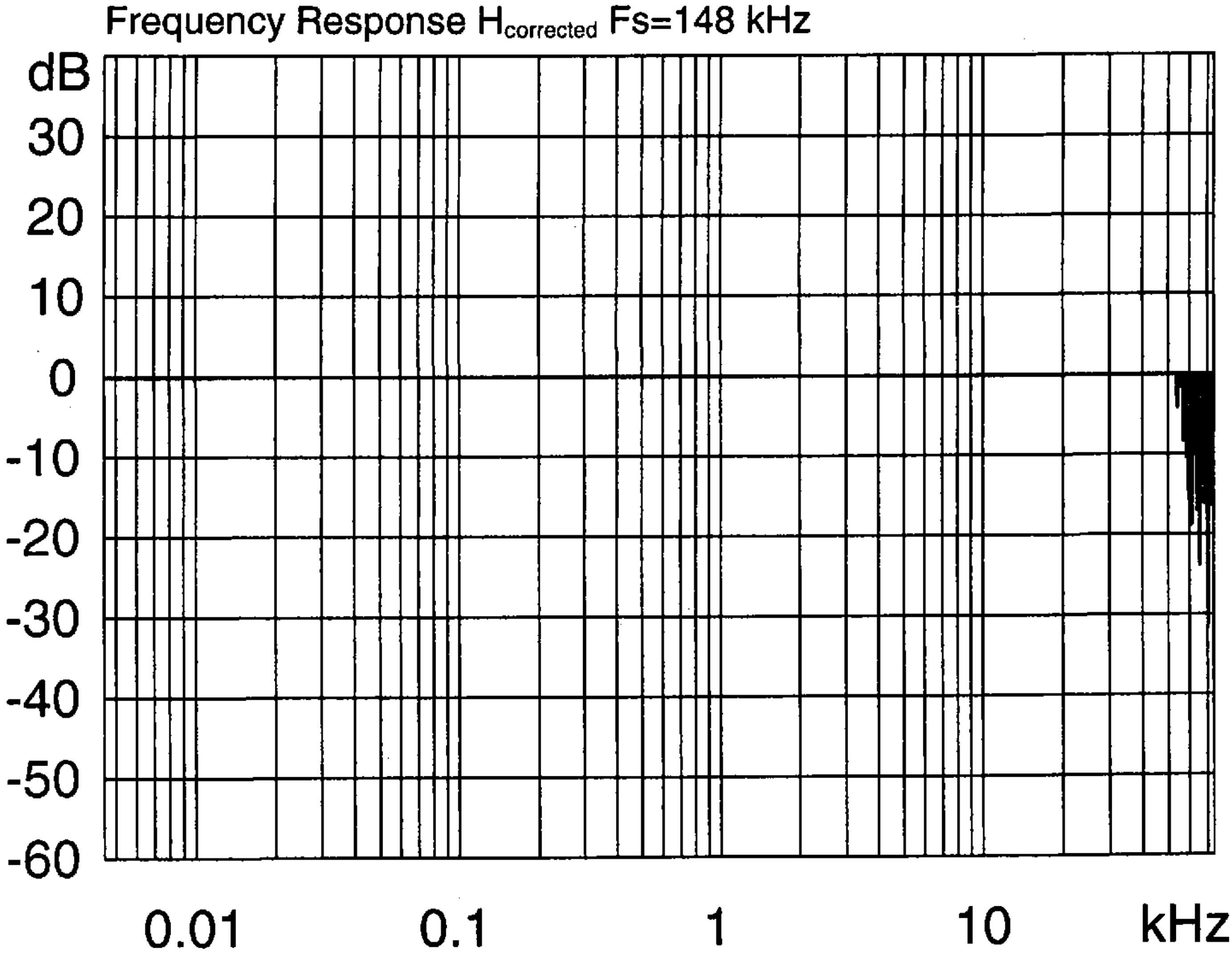


Figure no. 3d


Frequency Response $H_{corrected}$ Fs=148 kHz
dB
30
20
10
0
-10
-20
-30
-40
-50
-60
0.01
0.1
1
10
kHz