

Reproduction of artificial-head recordings through loudspeakers

Møller, Henrik

Published in:
Journal of the Audio Engineering Society

Publication date:
1989

[Link to publication from Aalborg University](#)

Citation for published version (APA):

Møller, H. (1989). Reproduction of artificial-head recordings through loudspeakers. *Journal of the Audio Engineering Society*, 37(1/2), 30-33. <http://www.aes.org/e-lib/browse.cfm?elib=6106>

General rights

Copyright and moral rights for the publications made accessible in the public portal are retained by the authors and/or other copyright owners and it is a condition of accessing publications that users recognise and abide by the legal requirements associated with these rights.

- Users may download and print one copy of any publication from the public portal for the purpose of private study or research.
- You may not further distribute the material or use it for any profit-making activity or commercial gain
- You may freely distribute the URL identifying the publication in the public portal -

Take down policy

If you believe that this document breaches copyright please contact us at vbn@aub.aau.dk providing details, and we will remove access to the work immediately and investigate your claim.

Reproduction of Artificial-Head Recordings through Loudspeakers*

HENRIK MØLLER

Institute of Electronic Systems, Aalborg University, DK-9220 Aalborg Ø, Denmark

Good results are obtained with artificial-head recording techniques when the directional aspects are considered. Unfortunately the system is traditionally restricted to reproduction through headphones, since reproduction through loudspeakers introduces detrimental crosstalk between channels. Different analog approaches have been made to the cancellation of crosstalk. However, digital signal processing opens up new opportunities. A system has been constructed in which a digitally filtered combination of the two channels is fed to each loudspeaker in a traditional stereo setup. The crosstalk is effectively canceled while the good imaging properties of headphone reproduction are preserved. The effect is unexpectedly independent of head position as long as the distances to the loudspeakers are equal. The system is shown to work in an anechoic room, but it is not formally limited to this. For use in a normal living room, more computing power is needed.

0 INTRODUCTION

The basic concept of the artificial-head recording technique is well known. In any listening situation the input to the hearing mechanism consists of two one-dimensional signals, the sound pressures at the two eardrums. If a recording/playback equipment is able to create the same sound pressures at the eardrums of a listener as would have been created in the concert hall, then the acoustic experience is reproduced correctly, including directional aspects, reflections, reverberation, and so on.

In practice, recordings are made with microphones in the ear canals of a carefully designed model of a human head, including pinnae. Reproduction is carried out through headphones, which ensures that each channel is reproduced in one ear only.

1 REPRODUCTION THROUGH LOUDSPEAKERS

The good directional characteristics of an artificial-head recording are destroyed if it is reproduced through loudspeakers. This is due to the crosstalk, which is introduced in any free-field listening. Crosstalk means that the right loudspeaker is heard not only with the

right ear but also with the left ear, and vice versa.

However, it can be shown that it is possible to add artificial crosstalk, which cancels out the natural crosstalk. This principle is shown in the following. X_{left} and X_{right} denote the two channels that are to be reproduced as sound pressure at the eardrums; Y_{left} and Y_{right} are the signals presented to the loudspeaker terminals; Z_{left} and Z_{right} denote the sound pressures at the two eardrums. X and Y have the unit of volts, while the unit of Z is pascals. The transfer functions from Y to Z are denoted by H , as indicated in Fig. 1. H has the unit of pascals per volt.

Now we have

$$Z_{\text{left}} = H_{\text{left-left}} \cdot Y_{\text{left}} + H_{\text{right-left}} \cdot Y_{\text{right}} \quad (1)$$

$$Z_{\text{right}} = H_{\text{left-right}} \cdot Y_{\text{left}} + H_{\text{right-right}} \cdot Y_{\text{right}} \quad (2)$$

We want that

$$Z_{\text{left}} = k \cdot X_{\text{left}} \quad (3)$$

$$Z_{\text{right}} = k \cdot X_{\text{right}} \quad (4)$$

k being a constant with units of pascals per volt. If we combine Eq. (1) with Eq. (3) and Eq. (2) with Eq. (4), and we solve with respect to Y , we get

* Presented at the 84th Convention of the Audio Engineering Society, Paris, France, 1988 March 1-4.

$$Y_{\text{left}} = k(H_{\text{right-right}} \cdot X_{\text{left}} - H_{\text{right-left}} \cdot X_{\text{right}})/D \quad (5)$$

$$Y_{\text{right}} = k(H_{\text{left-left}} \cdot X_{\text{right}} - H_{\text{left-right}} \cdot X_{\text{left}})/D \quad (6)$$

$$D = H_{\text{left-left}} \cdot H_{\text{right-right}} - H_{\text{left-right}} \cdot H_{\text{right-left}} \quad (7)$$

Some further manipulation is appropriate.

The loudspeaker frequency response is isolated through the introduction of P , the sound pressure at the center of the head, but with no head present. We have

$$\begin{aligned} H_{\text{left-left}} &= \left. \frac{Z_{\text{left}}}{Y_{\text{left}}} \right|_{Y_{\text{right}}=0} \\ &= \left. \frac{Z_{\text{left}}}{P} \right|_{Y_{\text{right}}=0} \cdot \left. \frac{P}{Y_{\text{left}}} \right|_{Y_{\text{right}}=0} \end{aligned} \quad (8)$$

Then

$$\left. \frac{Z_{\text{left}}}{P} \right|_{Y_{\text{right}}=0} = \text{FFC}_{\text{left-left}} \quad (9)$$

could be denoted the free-field correction of the head for a sound originating from the left loudspeaker and reaching the left ear. The unit is pascals per pascal.

$$\left. \frac{P}{Y_{\text{left}}} \right|_{Y_{\text{right}}=0} = \text{LFFR}_{\text{left}} \quad (10)$$

is the loudspeaker free-field response for the left loudspeaker. The unit is pascals per volt. Now Eq. (8) can

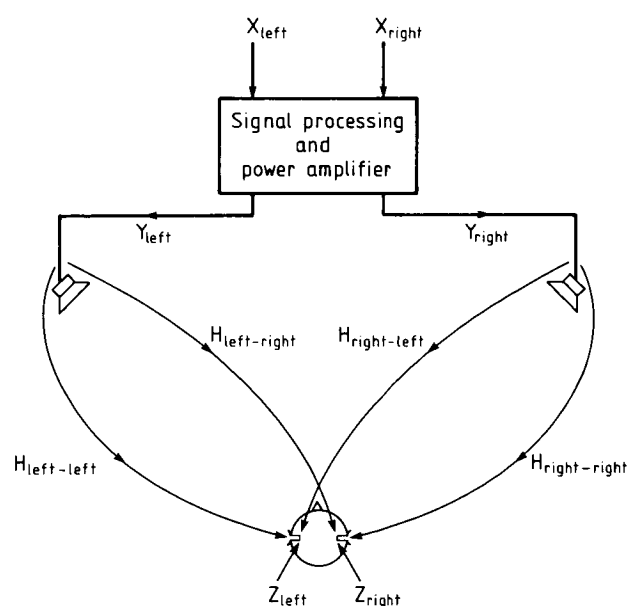


Fig. 1. Principal diagram showing the transmission from recording to sound pressures at the eardrums.

be written as Eq. (11), and similar expressions can be found for the other H values, as given in Eqs. (12)–(14),

$$H_{\text{left-left}} = \text{FFC}_{\text{left-left}} \cdot \text{LFFR}_{\text{left}} \quad (11)$$

$$H_{\text{left-right}} = \text{FFC}_{\text{left-right}} \cdot \text{LFFR}_{\text{left}} \quad (12)$$

$$H_{\text{right-left}} = \text{FFC}_{\text{right-left}} \cdot \text{LFFR}_{\text{right}} \quad (13)$$

$$H_{\text{right-right}} = \text{FFC}_{\text{right-right}} \cdot \text{LFFR}_{\text{right}} \quad (14)$$

If symmetry is assumed, then

$$\text{LFFR}_{\text{left}} = \text{LFFR}_{\text{right}} = C \quad (15)$$

$$\text{FFC}_{\text{left-left}} = \text{FFC}_{\text{right-right}} = A \quad (16)$$

$$\text{FFC}_{\text{left-right}} = \text{FFC}_{\text{right-left}} = B \quad (17)$$

Eqs. (5) and (6) can now be rewritten as

$$Y_{\text{left}} = \frac{A}{A^2 - B^2} \left(X_{\text{left}} - X_{\text{right}} \frac{B}{A} \right) \frac{k}{C} \quad (18)$$

$$Y_{\text{right}} = \frac{A}{A^2 - B^2} \left(X_{\text{right}} - X_{\text{left}} \frac{B}{A} \right) \frac{k}{C} \quad (19)$$

This signal processing is shown in block-diagram form in Fig. 2. The blocks to the right perform a gain control and an equalization of the loudspeakers. The left blocks are similarly introduced in the direct signal path of both channels and thus also perform an equalization. Among other things, this compensates for the fact that the ear canal appears two times in the transmission path, once at the recording and once at playback.

The real suppression of the crosstalk is carried out by the cross coupling of the two center blocks.

2 REALIZATION

The functions given in the preceding section were measured on a Neumann type KU-80i artificial head (Fig. 3). Until now the effort had been concentrated on realizing the cross coupling B/A as shown by curve c in Fig. 3. A possible way will be to approximate B/A with an analog filter or a recursive digital filter.

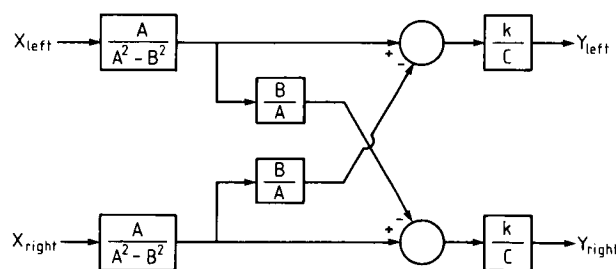


Fig. 2. Signal processing of Eqs. (18) and (19) in block-diagram form.

However, as accuracy is expected to be essential, it has been realized as a finite impulse response filter using Motorola XSP 56200 processors. The impulse response of the filter, calculated as the inverse Fourier transform of B/A , is shown in Fig. 4. At the time of printing, the blocks k/C and $A/(A^2 - B^2)$ have not been realized.

3 ASSESSMENT

The suppression of crosstalk is very effectively demonstrated in an anechoic room. With pink noise applied to both channels, the listener perceives the sound as being located in the head, as when listening with headphones. When the noise is applied to only one channel, the listener gets the impression of listening to a sound source located immediately outside the appropriate ear. If this ear is closed with a finger or an earplug, the listener is able to clearly indicate the correct position of the head by searching for minimum sound level at the opposite ear.

At present the reproduction of processed artificial-head recordings has only been evaluated subjectively. In general, listeners agree that the directional reproduction is at least as good as with headphones. Many

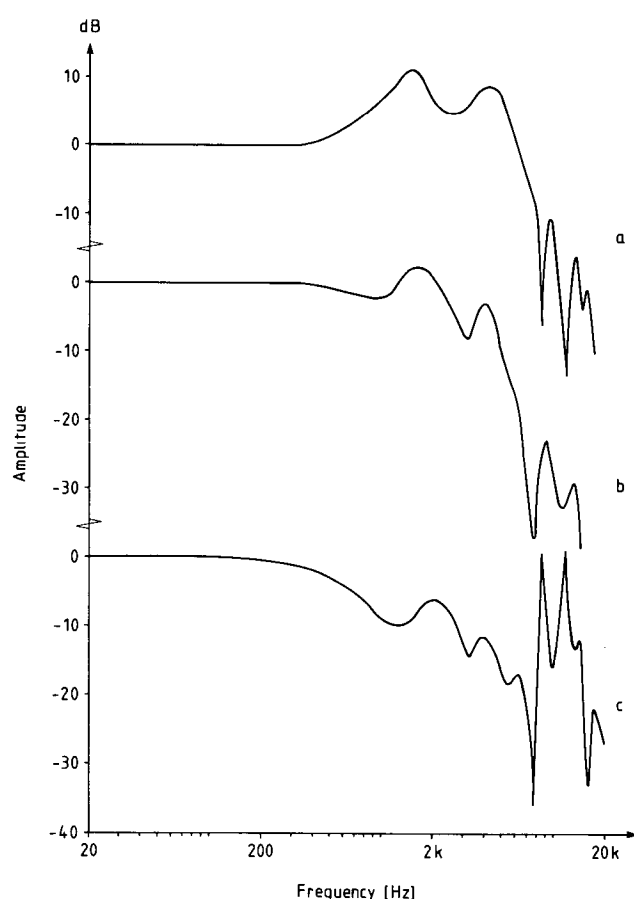


Fig. 3. Magnitude of transfer functions. Curve a— A (free-field correction for the ear for sound coming from the same side as the ear); curve b— B (free-field correction for the ear for sound coming from the opposite side); curve c— B/A . All curves are given for sound in the horizontal plane, 45° off frontal incidence.

listeners even indicate a better spatial discrimination, especially in the front region. The effect of the system is unexpectedly independent of the head position as long as the distances from the two loudspeakers are kept equal.

4 FUTURE WORK

Further investigations have been planned on the following matters:

1) Quantitative and objective evaluation of the observations given in Sec. 3.

2) Evaluation of the significance of blocks $A(A^2 - B^2)$ and k/C .

3) Significance of the impulse duration in the realization of filters.

4) The possibility of realizing the blocks of Fig. 2 using recursive infinite impulse response filters.

5) The possibility of using a normal listening room. The system is not formally limited to anechoic rooms, but for normal living rooms much longer impulse responses occur, and more computing power is needed.

6) Construction of the artificial head. Fig. 5 shows a principal diagram of the signal path from the sound field without a listener present to the sound pressure at the eardrum. It can be argued that full directional information is present in the open-circuit Thévenin sound pressure at the entrance to the ear canal. If this pressure is recorded rather than the pressure at the eardrum, all transfer functions are more regular, and the

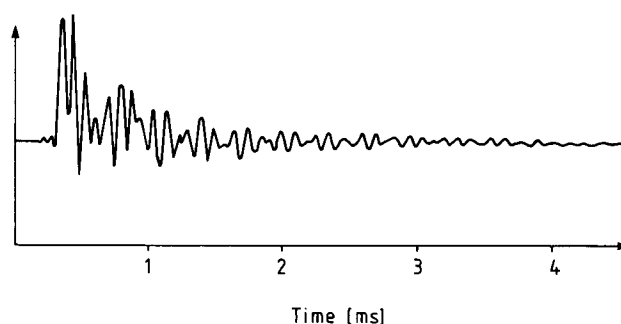


Fig. 4. Impulse response of B/A .

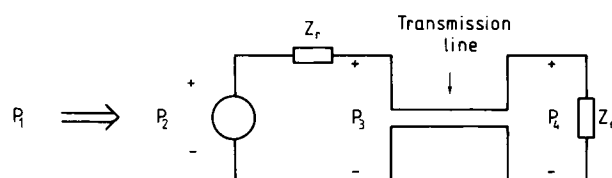


Fig. 5. Principal diagram of signal path from sound field without a listener present to sound pressure at one eardrum. P_1 —pressure without a listener; P_2 —open-circuit Thévenin pressure at entrance to ear canal; P_3 —actual pressure at entrance to ear canal; P_4 —pressure at eardrum, Z_r —radiation impedance seen from ear canal; Z_e —impedance of eardrum. The ear canal is represented by a transmission line. Only the transmission from P_1 to P_2 is dependent on the angle of incidence and the distance of the sound source. Thus it can be argued that P_2 contains full spatial information.

blocks of Fig. 2 become easier to realize. Furthermore, the final reproduction is less influenced by differences between the artificial ear and the ear of the listener, since a smaller part of the artificial ear is used. Recording with the Neumann type KU-81i artificial head is expected to approach this condition.

7) Problems at frequencies where A and B are approximately equal, and thus D is close to zero. This

happens especially at low frequencies.

5 ACKNOWLEDGMENT

The hard work was carried out by several students, whose contributions are gratefully acknowledged. I especially want to thank Knud Bank, Flemming Jensen, and Gert Ravn Jensen, all B.Sc. at present.

THE AUTHOR



Henrik Møller was born in 1951 in Aarhus, Denmark. He studied electrical engineering at the Danish Engineering Academy and received a B.Sc. degree in 1974. He later studied acoustics and earned a Ph.D. from Aalborg University in 1984. He has worked as a development engineer, a research engineer, and a professor of acoustics at Aalborg University. His main area of research involved the effects of infrasound and low-frequency noise on humans. His recent work involves the planning, construction, and testing of new acoustics

laboratories at Aalborg University. His most recent area of study is sound reproduction systems.

Dr. Møller is the author of several articles including the "Effects of Infrasound on Man," published in 1984. He received the Rockwool Sound Prize in Copenhagen in 1979 and the A.R. Angelos Grantee, Copenhagen, in 1986. He holds membership in several professional societies including the Danish Engineering Society, the Danish Acoustical Society, the Danish Biomedical Society, the IEEE, and others.