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HAMMERSHØI, Dorte; MØLLER, Henrik; SØRENSEN, Michael Friis;
Institute for Electronic Systems, Aalborg University, Aalborg, Denmark
LARSEN, Kim Alan; Perceptive Acoustics A.S., Aalborg, Denmark

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HEAD-RELATED TRANSFER FUNCTIONS: MEASUREMENTS ON 40 HUMAN SUBJECTS

Dorte Hammershøi, Henrik Møller, Michael Friis Sørensen
Institute of Electronic Systems
Aalborg University, Denmark

Kim Alan Larsen
Perceptive Acoustics A/S
Aalborg, Denmark

Summary

Information on Head-related Transfer Functions (HTFs) is used in design and evaluation of artificial heads for binaural recordings. Data are also needed for computer generation of binaural signals. By means of maximum length sequence (MLS) technique, HTFs were measured from 97 directions of sound incidence, covering the whole sphere. Measurements were made on 40 human subjects.

0. Introduction

A *head-related transfer function (HTF)* is a transfer function that - for a certain angle of incidence - describes the sound transmission from a sound source in a free field to a point in the ear canal of a human subject. HTFs are mainly used in binaural technique. Knowledge of HTFs from a large population is essential in design and evaluation of artificial heads. Furthermore, HTFs are used in computer synthesis of binaural signals, for instance in virtual environment applications or for auralization in room modelling systems. For computer synthesis, a HTF should usually be given in the time domain, that is as a *head-related impulse response (HIR)*. HTFs can also be used for determining diffuse field to free field corrections of hearing thresholds and equal loudness contours. The interest in the present investigation, though, is the use in binaural technique.

0.1 Binaural technique

The idea behind the binaural technique is the following. The input to the hearing consists of two signals: sound pressures at each of the eardrums. If these are recorded in the ears of a listener and reproduced exactly as they were (usually through headphones), then the complete auditive experience is assumed to be reproduced, including timbre and spatial aspects. Normally, during recording, the listener is replaced by an artificial head that has the same shape and the same acoustical properties as an average human head.

The reason why the listener is able to perceive the direction to a sound source, is that the sound on the transmission to the ears is exposed to filterings, corresponding to the particular direction. In most cases, amplitude versus frequency response as well as arrival time are different in the two ears. The hearing is able to "recognize" the filtering and thus determine the direction to the source. A thorough description of the directional hearing is given by Blauert [1].

0.2 Previous measurements of HTFs

Measurements of HTFs have already been described in the literature. Some examples are: Shaw [2], Blauert [3], Searle et al. [4], Mehrgardt and Mellert [5], Platte [6], Pösselt et al. [7], Wightman and Kistler [8], Middlebrooks et al. [9], Middlebrooks and Green [10] and Schmitz and Vorländer [11]. The investigations aim at different goals, which explains that the measuring technique, the number of source directions and other parameters vary considerably. For instance, some measurements were based on sine sweeps [2] [4], some on ⅓-octave noise [3], some on various impulse techniques [5] [8] [9] [10], and some on maximum length sequence (MLS) technique [11].

The point in the ear canal, where the recording is made, is called the reference point. Also the choice of this varies. Shaw [2] and Blauert [3] chose a point at the entrance to the ear canal, Mehrgardt and Mellert [5] 2 mm from the entrance, Platte [6] 4 mm from the entrance, Schmitz and Vorländer [11] 5 mm from the entrance, Pösselt et al. [7] also 5 mm from the entrance but blocking the ear canal at this point, Middlebrooks et al. [9] and Middlebrooks and Green [10] at least 5 mm from the entrance, Searle et al. [4] 10 mm from the entrance, and Wightman and Kistler [8] 1-2 mm from the eardrum. It is quite evident, that the sound pressure changes along the ear canal, and consequently the results cannot be compared directly. The change in sound pressure along the ear canal has been shown experimentally, for instance by Mehrgardt and Mellert [5] and Hammershøi and Møller [12]. Also the presence of a microphone in the ear canal may affect the measurement.

The results of previous experiments are most often presented as amplitude responses in the frequency domain. Even when time information is available, a sufficiently precise definition of the time axis may be lacking, if measurements are not carried out simultaneously at the two ears. In some cases the number of sound directions are low, and often no directions below the horizontal plane are included. The number of subjects is often insufficient for an evaluation of inter-individual differences.

Especially the variation in reference point and the differences in microphone technique is considered a major impediment for comparison of results. In the present investigation, a point is introduced, where influence from the transmission within the ear canal is avoided, while the complete directional information is still retained.

0.3 Model of sound transmission

A model of the sound transmission from a source in the free field to the ear canal is described by Møller [13], and verified by Hammershøi and Møller [12]. The sound transmission is divided into two parts, a directional dependent part that creates all directional cues, and a part that is independent of direction.

The directional dependent part of the model consists of the sound transmission from the free field to the Thevenin sound pressure at the entrance to the ear canal. This pressure does not normally exist physically, but if the ear canal is blocked, for example with an earplug placed with its outer end flush with the entrance to the ear canal, the Thevenin pressure is found outside the earplug.

The Thevenin pressure is called p_2. The free field sound pressure p_1 at the head center position - but with the listener absent - is used for reference. $[p_1 \rightarrow p_2]$ is used to denote the impulse response of the sound transmission from p_1 to p_2. It is easily verified that $[p_1 \rightarrow p_2]$ is non-causal for directions, where the ear is closer to the sound source than the middle of the head.

In the present paper, small letters are used to denote signals in the time domain, while capital letters denote the corresponding signals in the frequency domain. With the reference point chosen, the head-related transfer function (HTF) is denoted $[P_2/P_1]$, and the Head-related Impulse Response (HIR) is $[p_1 \rightarrow p_2]$.

The non directional dependent part of the model of the sound transmission is given in Figure 1. In the model, the sound source is p_2, and the source impedance is the radiation impedance seen from the ear canal, $Z_{radiation}$. The sound pressure at the entrance to the ear canal is denoted p_3. The ear canal acts as an acoustical transmission line terminated by the eardrum impedance $Z_{eardrum}$, and the pressure at the eardrum is denoted p_4.

As the transmission from p_2 through p_3 to p_4 is independent of direction, all three sound pressures contain the spatial cues, and they can all be used for binaural recordings. p_2 is the most convenient sound pressure to use, as it does not require insertion of microphones deeply in the ear canals, and the microphones can be placed in an earplug without interfering the measurement. Furthermore, recordings at other points may be unnecessary influenced by individual differences in the subject's ear canal. p_2 can be characterized as the sound pressure that includes the directional information and in addition to that as little individual information as possible. The choice of p_2 for recording also means that the ear canal can be avoided in the design of artificial heads. Based on these considerations a reference point outside the blocked ear canal has been chosen in the present investigation.

Møller showed elsewhere [13], how the correct total sound transmission in a binaural system can be achieved by the introduction of an electronic equalizer. The expression for the equalization needed becomes especially simple, if the reproduction involves a pressure division similar to the one seen between $Z_{ear\ canal}$ and $Z_{radiation}$ during

recording. In order to deliver data to an evaluation of this assumption, the present investigation includes measurements of this pressure division, that is $[P_3/P_2]$.

0.4 Goal of present investigation

It is the purpose of the present investigation to provide knowledge about the head-related impulse responses, $[p_1 \rightarrow p_2]$, (HIRs) and head-related transfer functions, $[P_2/P_1]$, (HTFs) for a large number of people and for a large number of angles, covering the whole sphere.

In addition to this, knowledge should be obtained about the pressure division $[P_3/P_2]$. This is assumed independent of direction, and it only need to be measured for a few directions to the sound source.

1. Method

Measurements were carried out on subjects standing in an anechoic chamber. 8 loudspeakers were used, and the subjects were rotated to yield measurements from 97 different directions, covering the whole sphere. Impulse responses were measured for the transmission from a voltage at the input of the power amplifier to output of the measuring microphone, placed to measure p_1, p_2 or p_3. The impulse responses were measured with maximum length sequence (MLS) technique, and a master-slave configuration of two measurement systems enabled sample synchronous measurements at both ears. HTFs $[P_2/P_1]$ and pressure divisions $[P_3/P_2]$ were obtained through Fourier transformation of the measured impulse responses, followed by appropriate divisions. Subsequent inverse Fourier transformations produced HIRs $[p_1 \rightarrow p_2]$.

For each subject 107 measurements were made. p_2 was measured for 97 directions with electret microphones. Both p_2 and p_3 were measured for 5 directions (with incidence of the sound wave being front, back, left, right and above) with probe microphones. p_1 was measured with each electret microphone for each loudspeaker every working day.

1.1 Subjects

40 subjects participated, 22 male and 18 females. 31 were randomly chosen students and 9 were staff members. All subjects had controlled normal hearing, and furthermore none had reported ear abnormalities that might affect the middle ear function. Full size, torso, and pinna photos of the subjects were made for the purpose of relating the subject's anatomy to the HTFs. The same subjects participated in a parallel investigation of headphone transfer characteristics by Møller et al. [14].

1.2 Microphones

The choice of microphones for measurements in the human external ear is a compromise between having a satisfying sensitivity and a microphone of acceptable dimensions. For measurement of the Thevenin pressure p_2, a Sennheiser KE 4-211-2 miniature microphone was used. This is an electret microphone, cylindrical in shape Ø 4,75 mm × 4,20 mm, see photo in Figure 2.

The KE 4-211-2 is a pressure microphone of the back electret type, and it has a built-in FET amplifier. The microphone itself has a sensitivity of approximately 10 mV/Pa. Coupled with a gain as suggested in the data sheet, the sensitivity increases to approximately 35 mV/Pa. A small battery box was designed, and in order to increase the output signal and to reduce the output impedance, a 20 dB amplifier was built into the same box. Two selected microphones were used throughout the experiment, one for each ear. Frequency responses of these microphones including the front-end amplifiers are given in Figure 3.

During measurement of p_2 the microphone was mounted in an EAR earplug placed in the ear canal. The microphone was inserted in a hole in the earplug, and then the soft material of the earplug was compressed during insertion in the ear canal. As the earplug relaxed, the outer end of the ear canal was completely filled out. The end of the earplug and the microphone were mounted flush with the ear canal entrance. The placement of the microphone is sketched in Figure 4, and a photo is given in Figure 5.

The reference sound pressure p_1 from each loudspeaker was measured with each of the miniature microphones. The microphone was placed at the position, where the middle of the subject's head would be during measurement. In order to disturb the field as little as possible, the microphones were fixed by a thin wire and with an orientation giving 90° incidence of the sound wave for all loudspeakers. In this way, the p_1 measurements were minimally influenced by the presence of the microphone in the sound field, and equally influenced for all p_1 measurements.

During measurement of p_3 at the entrance to the open ear canal, the microphone could not be concealed in an earplug. It was estimated that the miniature microphone was so big that it would disturb the sound field. For measurement of p_3 a probe microphone was therefore used. This type of microphone would not disturb the measurement, since the interference of the probe tip is minimal, but it has the disadvantage of a low sensitivity. The probe microphone chosen was a B&K 4182 with a 45 mm metal tube extended by a short piece of flexible tube. The sensitivity of the probe microphone was approximately 3 mV/Pa, and frequency responses with a 50 mm metal tube are seen in Figure 6.

The probe microphone was attached to the external ear by a flexible metal strap. The strap and the flexible tube were individually adjusted to fit the shape of the external ear. To avoid displacement of the probe microphone during the experiment it was fixed along the subject's neck with tape, and the position of the tip was controlled before and after each single measurement.

In order to reduce the post processing of data needed for determination of $[P_3/P_2]$, and thereby the possible sources of errors, p_2 measurements were also carried out with the probe microphone for the five angles where p_3 measurements were made.

During measurement of p_2 the flexible tube bent slightly against the earplug. A paper label was attached to the soft and rough surface of the earplug to avoid occlusion of the tip. The placement of the probe microphone with the ear canal blocked and unblocked is sketched in Figure 7. The photo in Figure 8 shows, how the probe microphone is attached to the subject's ear and neck.

In order to minimize the error due to displacement of the probe tip between measurement of p_2 and p_3, the probe measurements were carried out in the following way: p_2 was measured first. Then the earplug was carefully removed with as little disturbance of the probe microphone as possible. Immediately afterwards p_3 was measured.

Even though efforts were undertaken to keep the body of the probe microphone close to the subject's body, a minor disturbance of the sound field could be expected from the microphone housing. Therefore, the measurements with the probe microphones may be considered less suitable for determination of HTF and HIR than the measurement with miniature microphones.

1.3 Free field setup

The measurements were carried out in an anechoic chamber with a free space between the wedges of 6,2 m (length) by 5,0 m (width) by 5,8 m (height). Measurements were made with sound arriving from a total of 97 different directions, covering the whole sphere. The convention for indication of distance and direction is seen in Figure 9. ϕ defines azimuth, θ elevation and r the distance from the sound source to the center of the subject's head. $(\phi, \theta) = (0°, 0°)$ is defined as the direction in front of the subject. Positive values of ϕ indicate directions to the left of the subject, while positive values of θ indicate directions above the horizontal plane. Thus the whole sphere is covered for $-180° < \phi \leq 180°$ and $-90° < \theta \leq 90°$.

Both ϕ and θ were chosen in steps of 22,5°. An exception was at $\theta = \pm 67{,}5°$, where the step size for ϕ was chosen to be 45°, since 22,5° would cause the directions to lie rather close on the sphere. Figure 10 illustrates the positions of the sound sources for a quarter of the sphere. r was chosen to be 2 m, except for $\theta = -67{,}5°$, where the existence of a net floor made a reduction to 1,95 m necessary. No measurements were made at $\theta = -90°$, that is from a position directly under the subject.

The number of loudspeakers was 8. The speakers were fixed in an arc of a circle with θ ranging from 90° to -67,5° in steps of 22,5°. The setup in the anechoic chamber is seen in Figure 11. The subject stood on a rotatable platform which could be fixed at the desired values of ϕ. The subject was standing on the platform in a natural upright position, and a small backrest mounted on the platform helped the subject to stand still. The position of the platform and an individual adjustment of its height caused an imaginary point right between the subject's ears to be in the center of the sphere.

For help in the control of horizontal position and orientation of the subject's head, the subject had a paper marker on top of the head. This marker was observed through a video camera placed in the direction $\theta = 90°$ and shown on a moveable monitor in front of the subject. Using this, the subject could correct position and azimuth, see Figure 12.

The operators had a similar monitoration for observation of the subject's exact position and for controlling that the subject did not move during each single measurement. If movements were observed, the measurement was discarded and redone.

The loudspeakers used were 7 cm membrane diameter midrange units (Vifa M10MD-39) mounted in 15,5 cm diameter hard plastic balls. Free field response of a loudspeaker is shown in frequency and time domain in Figure 13.

1.4 Measuring setup

The general purpose measuring system known as MLSSA (Maximum Length Sequence System Analyzer) was used. Maximum length sequences are binary two level pseudo-random sequences. The basic idea of MLS technique is to apply an analogue version of the sequence to the linear system under test, sample the resulting response, and then determine the system impulse response by cross-correlation of the sampled response with the original sequence.

The method offers a number of advantages compared to traditional frequency and time domain techniques. The method is basically noise immune, and combined with averaging, the achieved signal to noise ratio is high. A thorough review of the MLS method is given by Rife and Vanderkooy [15].

For the purpose of measuring at both ears simultaneously, two MLSSA systems were used, coupled in a master-slave configuration by a purpose made synchronization unit allowing sample synchronous measurements. A sketch of the complete measuring setup is given in Figure 14. The MLSSA systems were set up with autorange enabled, allowing the best possible utilization of the dynamic range. If one of the systems adjusted input gain during a measurement, the two recordings were no longer synchronous, and an error state was indicated by the synchronization unit. The actual measurement was then discarded and redone.

The 4 V peak-to-peak stimulus signal from the master MLSSA board was sent to the power amplifier (Pioneer A-616) that was modified to have a calibrated gain of 0,0 dB. From the output it was directed through a switch-box to the loudspeaker in the measurement direction. The free field sound had a level of 75 dB(A) at the subject's position, a level where the stapedius was assumed to be relaxed.

From the microphone (miniature as well as probe) the signal was sent through a measuring amplifier, B&K 2607. The output signal from the measuring amplifier was attenuated by a factor of 2 in order to adjust the dynamic range of the signal to the input range of the MLSSA board. The microphone sensitivities at 1 kHz were measured every work day during the two month measurement period, and deviations from the nominal value were below ±0,3 dB.

The sampling frequency of 48 kHz was provided by an external clock. To avoid frequency aliasing, the 20 kHz Chebyshev low pass filter of the MLSSA board and the 22,5 kHz low pass filter of the measuring amplifier were used. Also the 22,5 Hz high pass filter on the measuring amplifier was active.

Preliminary measurements on the free field setup using the maximum MLS length offered by MLSSA, 65535 points, showed that a length of 4095 points was sufficient to avoid time aliasing. In order to achieve a high signal to noise ratio, the recording was averaged 16 times, called pre-averaging in the MLSSA system. Even with this averaging the total time for a measurement was as short as 1,45 seconds. During this period the subjects were normally able to stand still. All measured impulse responses were very short, and only the first 768 samples of each impulse response - corresponding to 16 milliseconds - were computed and saved.

1.5 Data processing

Results of the measurements were impulse responses for the transmission from input to the power amplifier to output of the measuring amplifier. In order to obtain the wanted information, some post processing was needed. This was carried out in MATLAB.

The measured impulse responses all included an initial delay, corresponding to the propagation time from the loudspeaker to the measuring point (approximately 6 milliseconds). All responses were very short, duration only a few milliseconds. Therefore, only samples from 256 through 511 were processed (time from 5,33 ms to 10,65 ms). The restriction to this time window eliminated reflections from the monitor in the anechoic chamber.

For determination of the HTF $[P_2/P_1]$, the selected portion of the p_1 and p_2 impulse responses were Fourier transformed, and a complex division was carried out in the frequency domain. As the same equipment was involved during measurement of p_1 and p_2, the influence of equipment cancels out in the division.

A similar procedure was used to determine the pressure division $[P_3/P_2]$. In this process, it is essential that the probe microphone was used for measurement of both p_2 and p_3.

The HIR $[p_1 \rightarrow p_2]$ was determined through an inverse Fourier transform of $[P_2/P_1]$. Before the transformation, $[P_2/P_1]$ was filtered by a 4th order Butterworth filter (bilinearly transformed) in order to prevent from frequency aliasing.

1.6 Quality of measurements

As the subjects during the measurements had a natural upright position without fixation of the head, errors due to wrong position and orientation could not be avoided, even when the position marker was carefully placed. From observations of the subjects during the experiments, the following was concluded: Displacements in the vertical and horizontal plane were in general below 1 cm, leading to total errors on the absolute time scale rarely exceeding ±2 sample (0,7 cm corresponds to 1 sample at 48 kHz).

The error due to wrong azimuth as read from the monitor was estimated to be below ±3°, leading to a displacement of each ear below ±0,5 cm, corresponding to a time displacement below ±0,7 sample. A second azimuth error of the same order of magnitude is due to the fixation of the position marker. A further displacement of the ears up to ±1 cm in horizontal and vertical direction was seen, due to asymmetry of the human head.

Error in elevation due to tilting of the head might exceed the above mentioned error for azimuth, since it was given higher priority to leave the subject's head free of any fixation devices, than to control the exact elevation angle. It should be kept in mind that the measurements were made simultaneously in the two ears, and all responses do represent a real direction. This would not be the case, if the head position and orientation could be different for measurements from the two ears.

In the free field setup, the only objects that could give rise to reflections within the time window, were other loudspeakers and the arc holding the loudspeakers, the backrest and the platform on which the subject was standing. Reflections from the loudspeakers and the arc were eliminated by means of sound absorbing material wrapped around the speakers and the arc. The effect was verified through free field measurements at the subject's position with parts of the loudspeaker arc removed and repositioned.

Influence of reflections from the backrest was totally eliminated by covering it with sound absorbing material. This was verified through measurements on a head and torso simulator B&K 4128, measured in expected worst case directions, with and without backrest. Possible reflections from the platform would only occur in the time window used for elevations between -67,5° and -22,5°. The platform was also wrapped with sound absorbing material, and the foot plate was shielded in the direction of the loudspeakers with a sound absorbing fibre mat. For practical reasons the effect of the coverage of the platform was not verified through measurements.

At high frequencies the exact position of possible narrow peaks and dips will depend on the precise placement of the miniature microphone or the probe tip relative to the ear. Inaccurate placement will be especially noticeable in data for the pressure division $[P_3/P_2]$, since the p_3 data are obtained after replacing of the probe tip. The worst case error made when replacing the probe tip was estimated to be 2 mm. If, for instance, a dip at 8,5 kHz is caused by a first order cancellation due to a reflective surface 10 mm from the microphone, a 2 mm displacement of the microphone towards the surface will move the dip to 10,6 kHz. The error in the resulting pressure division

depends on the height and width of the peaks and dips in the measured frequency responses to p_2 and p_3. The general shape of the pressure division curve is not influenced, but narrow fluctuations should be expected at high frequencies due to imperfect replacing of the probe tip.

When determining HTF the reliability depends on the ability to repeat the measurements. The well defined reference point in combination with the miniature microphone inserted in the earplug allows repetition of the reference point within ±1 mm. At 20 kHz the 2 mm worst-case difference corresponds to less than 1/8 of a wavelength.

The signal to noise ratio cannot be expressed as a single figure, since it depends on the response being measured. For illustration, the total noise level was found by repeating a measurement, but with the loudspeaker electrically replaced by a resistor. All gain settings were as in the original measurement. Examples with the two types of microphones are, given in Figure 15 and Figure 16. The signal to noise ratio is typically around 70 dB and never below 50 dB for the miniature microphones. For the probe microphones corresponding figures are approximately 10 dB lower.

2. Results

At the time of submission of the manuscript, only processed data from 12 subjects were available. A thorough analysis will await the remaining data. Only examples of results are given in this section. Data are chosen by random and are not necessarily representative. The comments to the results should not be regarded as conclusive.

2.1 HTFs and HIRs

Figure 17 is an example of HTF and HIR shown for one subject in the direction $(\phi, \theta) = (90°, 0°)$ (left of subject, horizontal plane). As expected, the signal at the right ear is attenuated compared to the left ear, especially at high frequencies. It is also seen that the impulse response is non-causal for the left ear, because this ear is closer to the sound source than the middle of the head. The interaural time difference is approximately 600 μs.

The Figures 18, 19 and 20 show HTFs for one subject's left ear in the median, horizontal and the frontal plane. The variation of the HTF for different directions is easily seen. The dominant variation is found in the frequency range 1 kHz to 20 kHz, where some characteristic notches are seen to be dependent on the direction. In the median plane the impulse response starts approximately at a time of zero. For directions in the other planes, where the ear is closer to the sound source than the middle of the head, the impulse response is non-causal.

2.2 Pressure divisions $[P_3/P_2]$

Figure 21 shows the pressure division $[P_3/P_2]$ for one subject, both ears, in 5 different directions. The major information in this figure is that the pressure division is independent of direction as assumed. Local fluctuations probably due to imperfect replacing of the probe tip are though seen. In addition it can be seen that the pressure division for both ears are very similar, thus indicating symmetry.

2.3 Inter-individual variation

Figure 22 gives an indication of the individual differences. The upper set of curves show HTF for the left ear of 12 subjects for one direction, $\theta = 90°$. It is seen that some variation is present, although the general shape is quite clear up to approximately 13 kHz.

The inter-individual variation in the pressure division is illustrated in the second set of curves, where $[P_3/P_2]$ is shown for the same 12 subjects. Here the variations are larger, and they appear already from approximately 8 kHz. One subject deviates significantly as far down in frequency as 2,5 kHz.

p_3 will reflect the individual variation in HTF $[P_2/P_1]$ as well as the individual variation in the pressure division. Therefore, the variation in $[P_3/P_1]$ is expected to be larger than the variation in $[P_2/P_1]$. This is exactly what is seen from the lower set of curves, where $[P_3/P_1]$ is shown.

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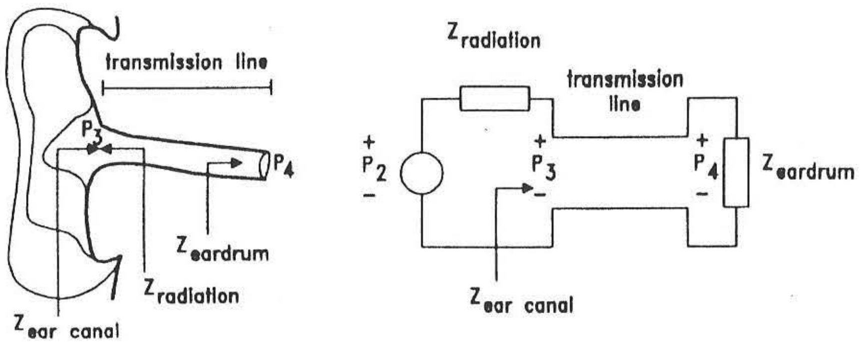


Figure 1. Sound transmission through the external ear; sketch of the anatomy and an analouge model, as described by Møller [13].

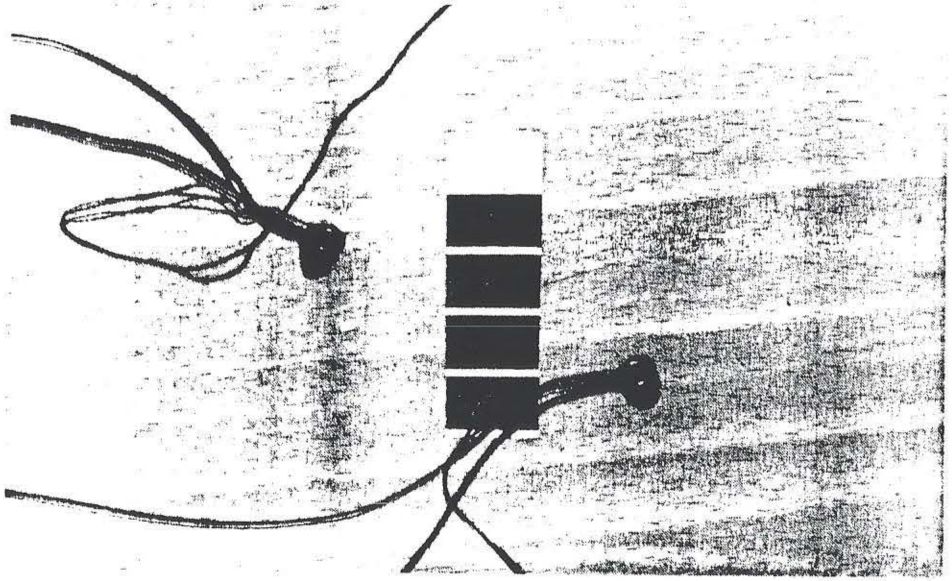


Figure 2. Photo of two miniature microphones, Sennheiser KE 4-211-2. The white lines indicate 1 cm intervals. Two soft electrical wires are connected by soldering at the back side. Additionally, a thread is mounted as an aid during removal of the microphone from the ear canal. The microphone and the connections are protected by flexible heat-shrinkable tubing.

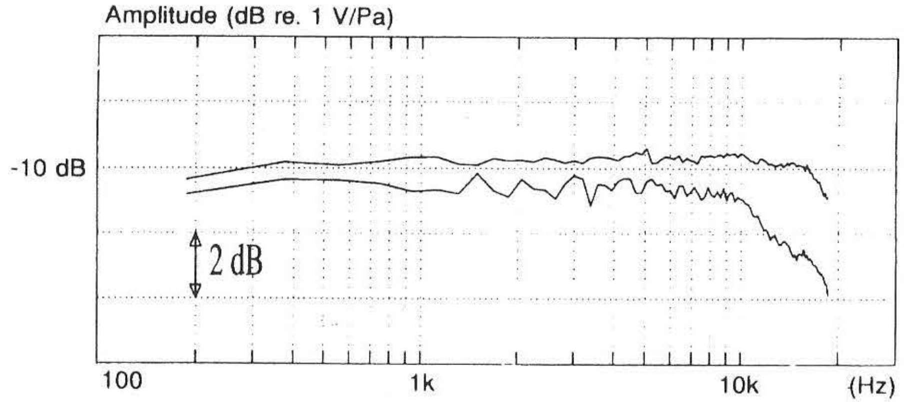


Figure 3. Frequency responses of the two miniature microphones, Sennheiser KE 4-211-2, including the front-end amplifiers.

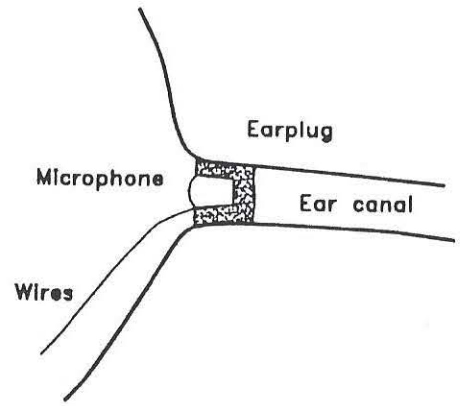


Figure 4. Sketch of the miniature microphone inserted in an earplug in the ear canal.

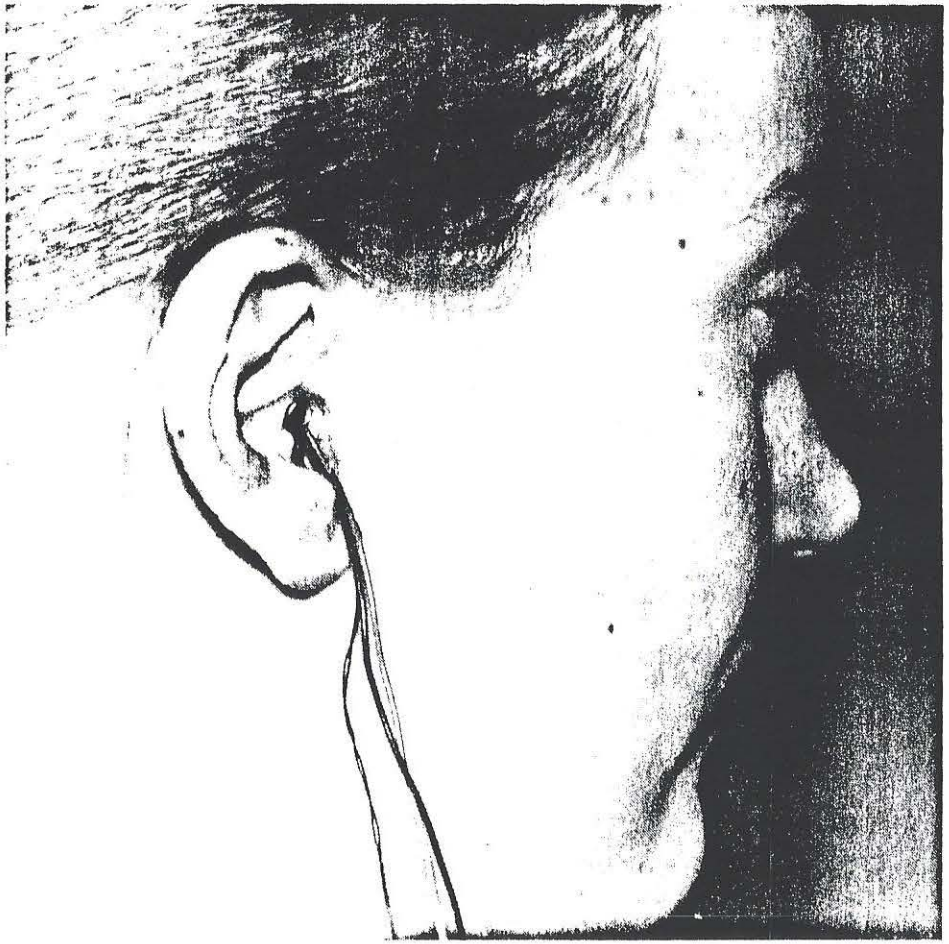


Figure 5. A photo of the miniature microphone, mounted in the ear of a subject.

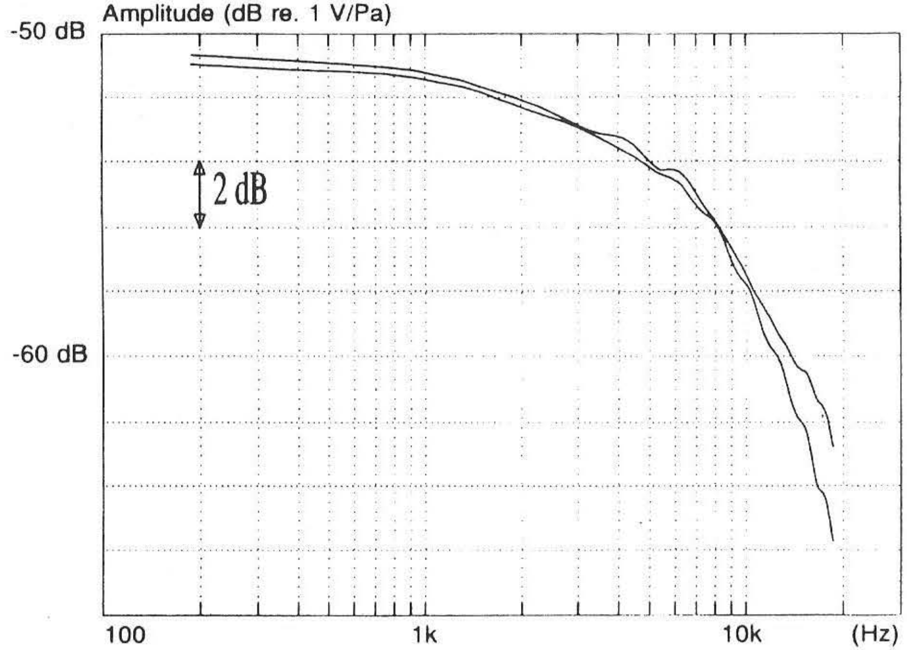


Figure 6. Frequency responses of the two probe microphones, B&K 4182, including a 50 mm metal tube.

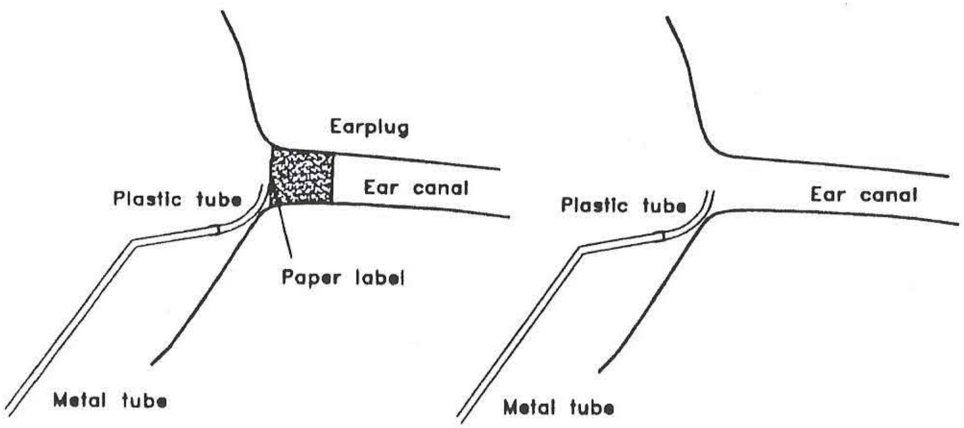


Figure 7. Sketch of placement of the flexible tube extending the metal tube of the probe microphone. Left side with the ear canal blocked using an earplug, right side with the ear canal unblocked.

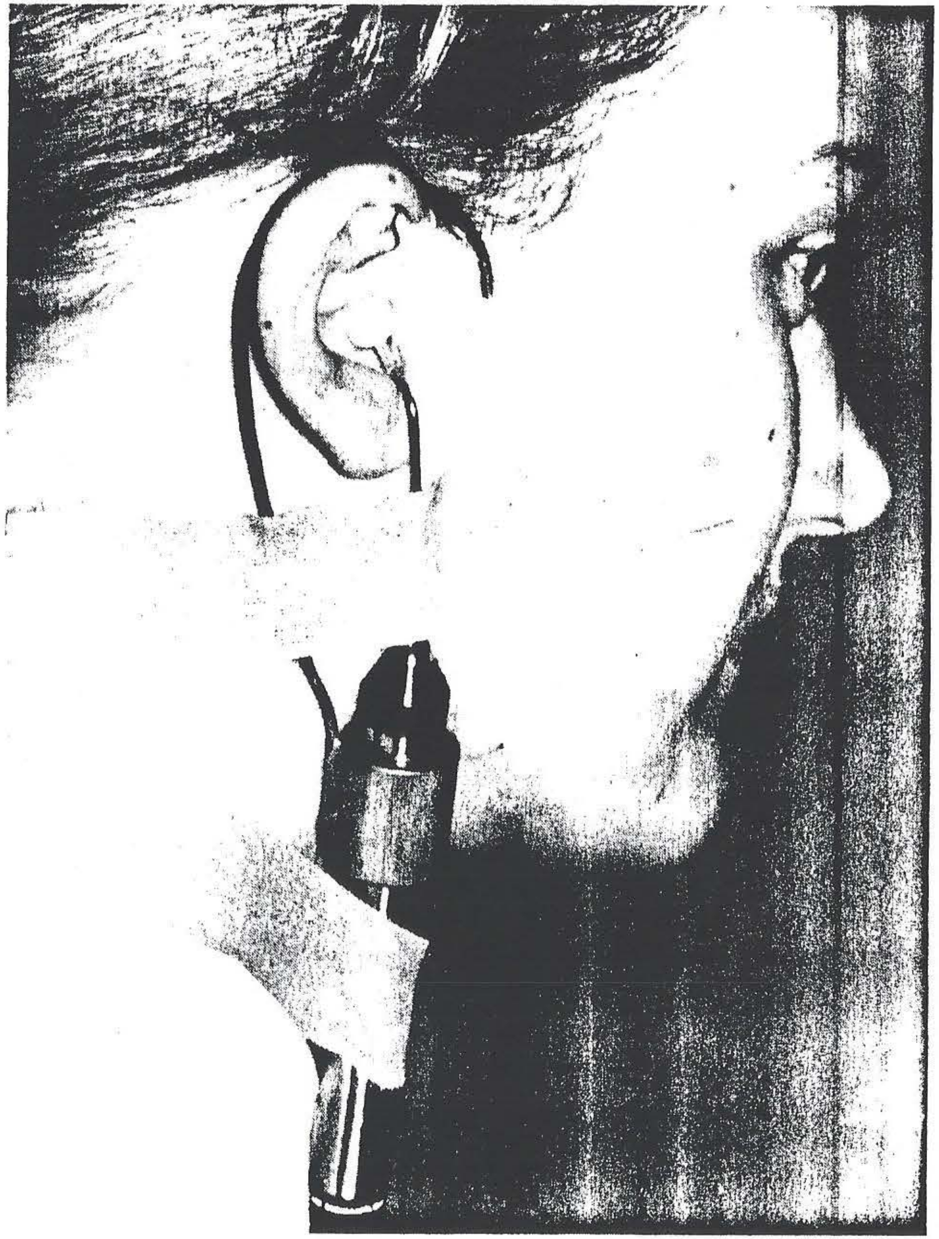


Figure 8. Photo of the probe microphone, mounted on a human subject.

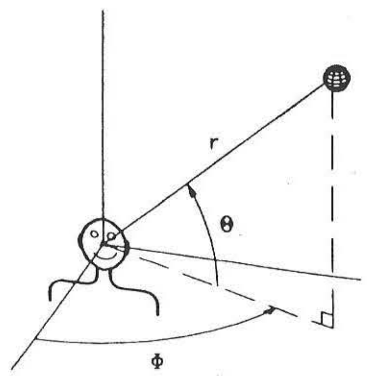


Figure 9. Sound source and listener in a free field. Conventions for the variables indicating distance and direction are shown. Origo is at the midpoint of the imaginary line connecting the two ears.

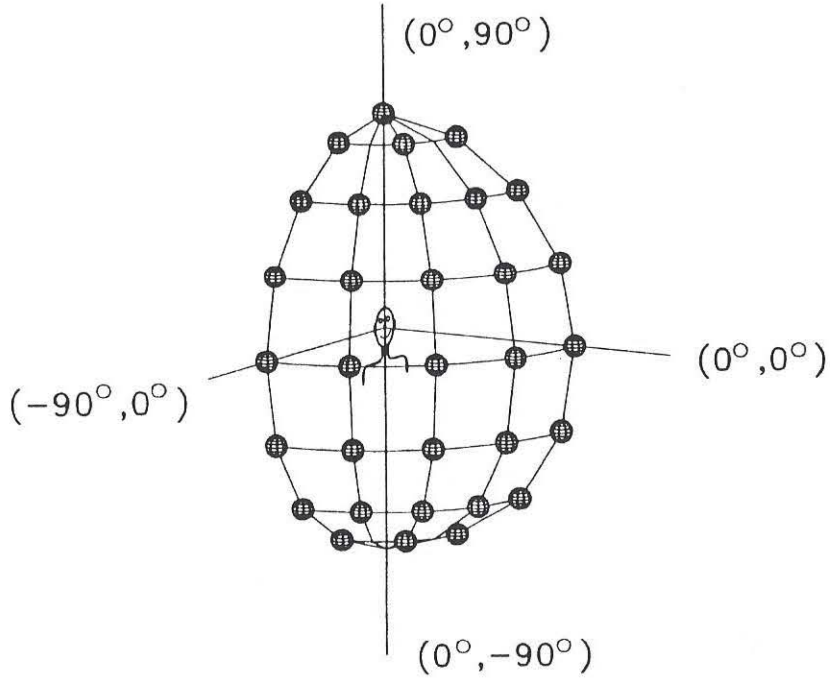


Figure 10. Positions of the sound sources shown for a quarter of a sphere.

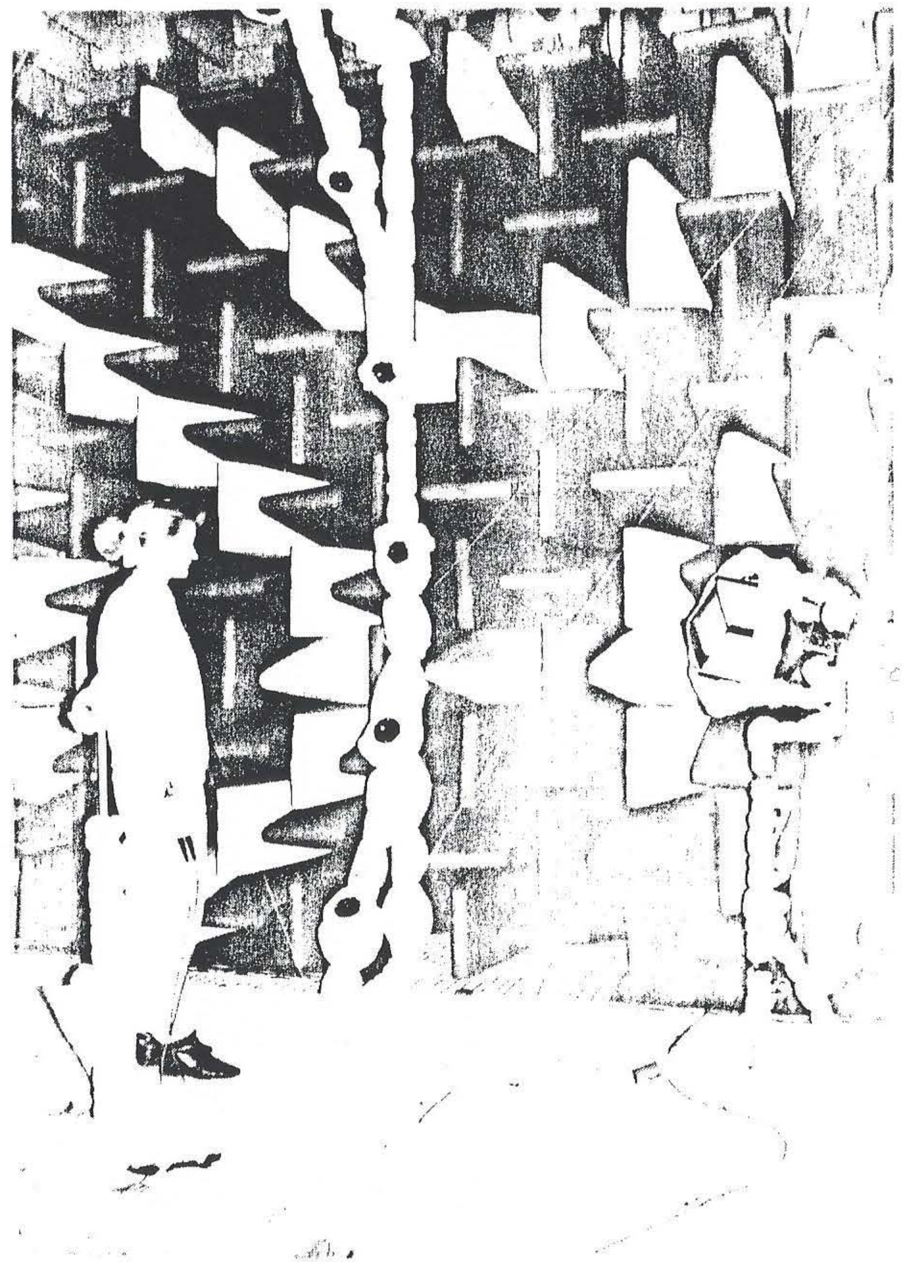


Figure 11. Photo showing the free field setup in the anechoic chamber. Note that for phototechnic reasons the monitor is slightly closer to the subject than during measurements.

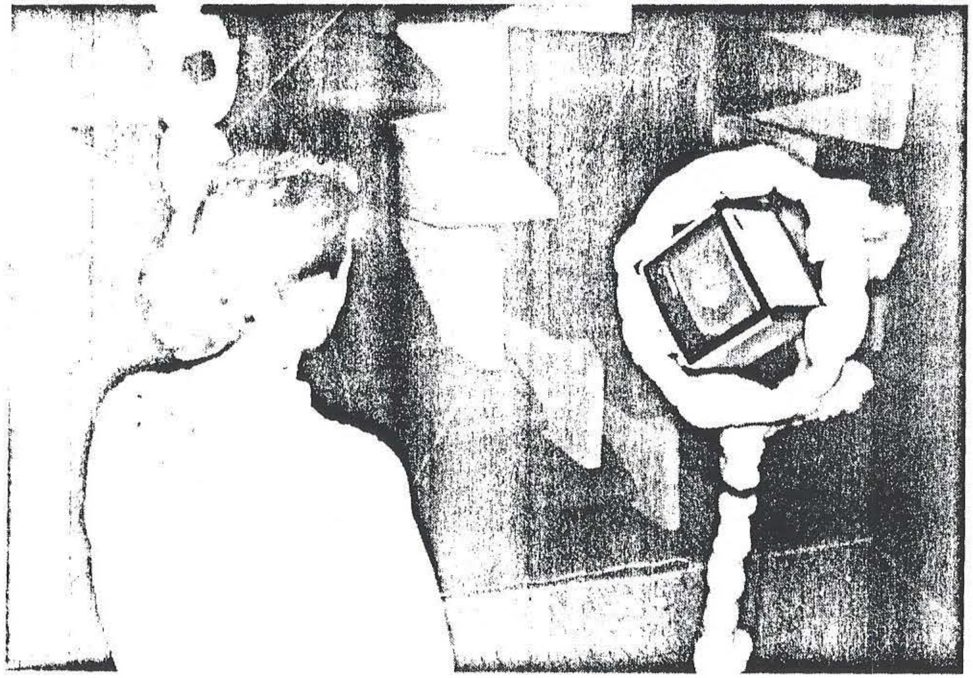


Figure 12. Photo showing the direction marker and the moveable monitor. Note that for phototechnic reasons the monitor is closer to the subject than during measurements.

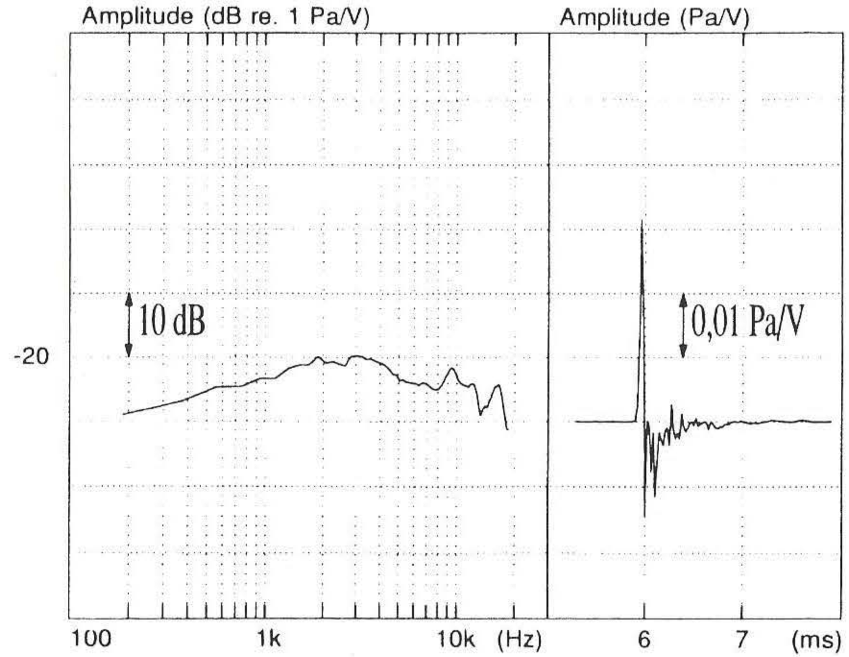


Figure 13. Typical frequency response and impulse response for the loudspeakers used (Vifa M10MD-39 mounted in 15,5 cm hard plastic ball), measured at a distance of 2 m with a B&K 4136 microphone, 90° incidence.

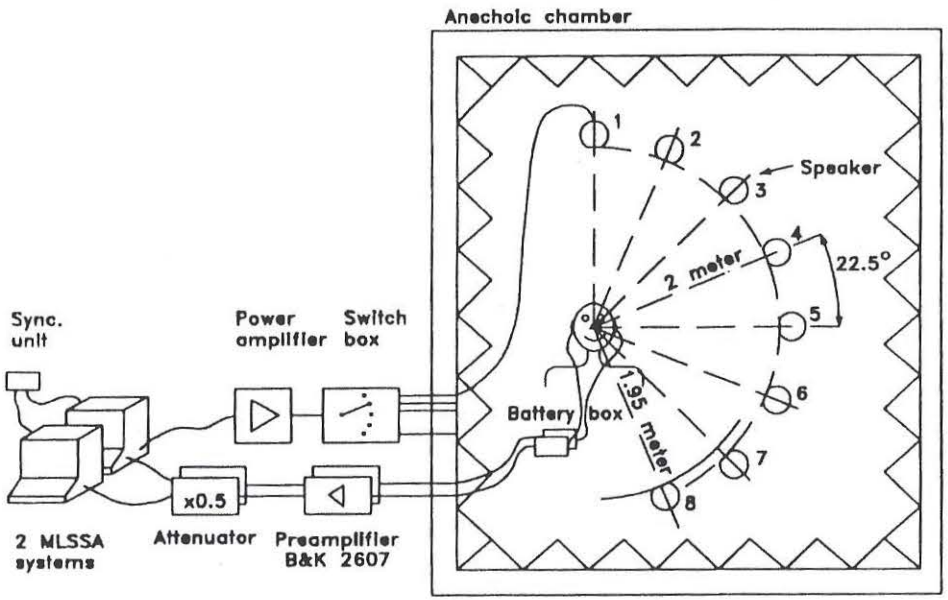


Figure 14. Sketch of the measuring setup.

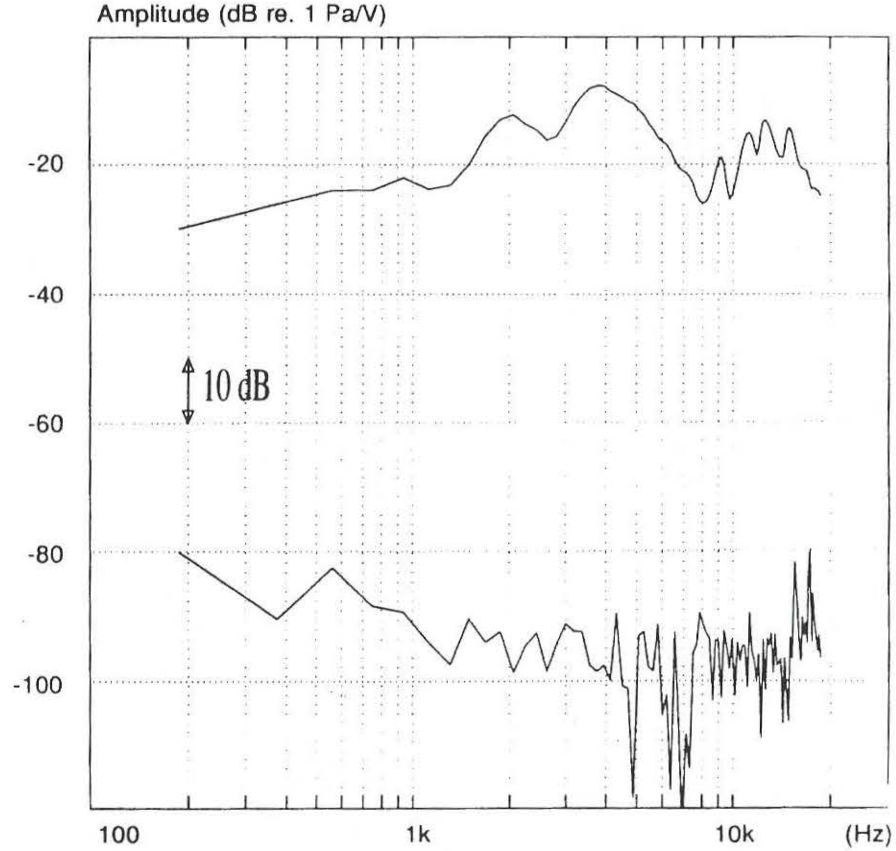
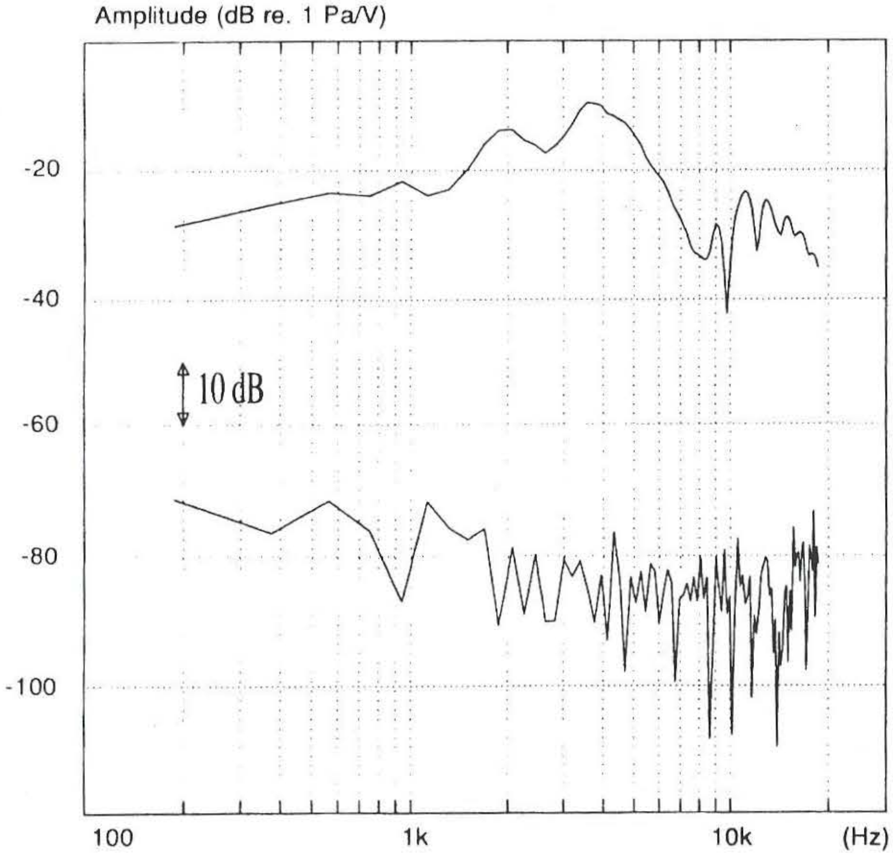


Figure 15. Illustration of signal to noise ratio. Upper curve shows a typical miniature microphone measurement, lower curve shows a measurement with all settings unchanged, but with the loudspeaker replaced by an 8 Ω resistor. The signal to noise ratio (distance between the two curves) varies from 50 dB to more than 80 dB. It is emphasized that these data are original measurements and not computed HTFs.



igure 16. Illustration of signal to noise ratio. Upper curve shows a typical probe icrophone measurement, lower curve shows measurement with all settings nchanged, but with the loudspeaker replaced by an 8 Ω resistor. The signal to noise itio (distance between the two curves) varies from dB 40 to more than 70 dB. It is mphasized that these data are original measurements and not computed HTFs.

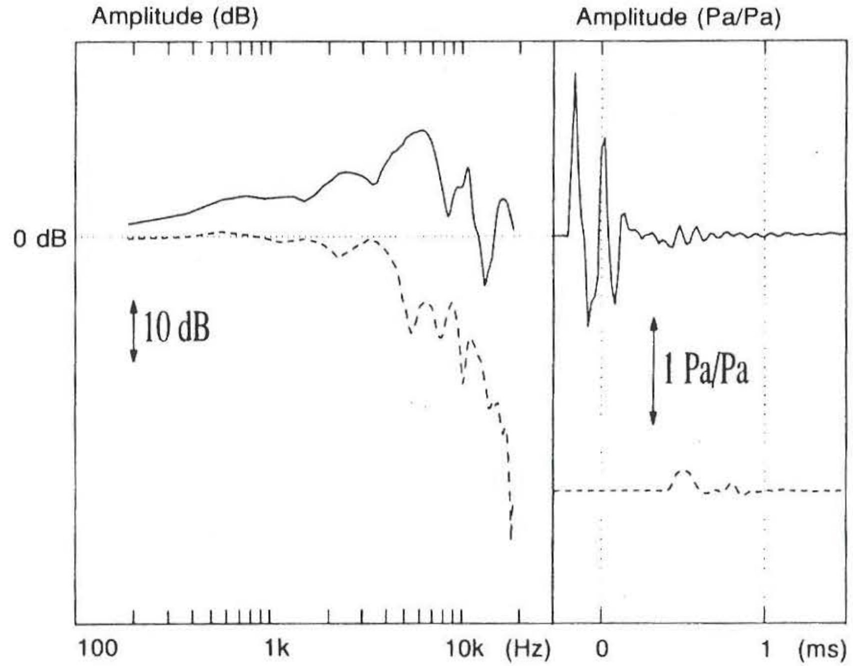


Figure 17. Example of HTF and HIR shown for one subject for the direction (ϕ, θ) = (90°, 0°) measured with the miniature microphone. Both ears are shown (dashed line is the right ear).

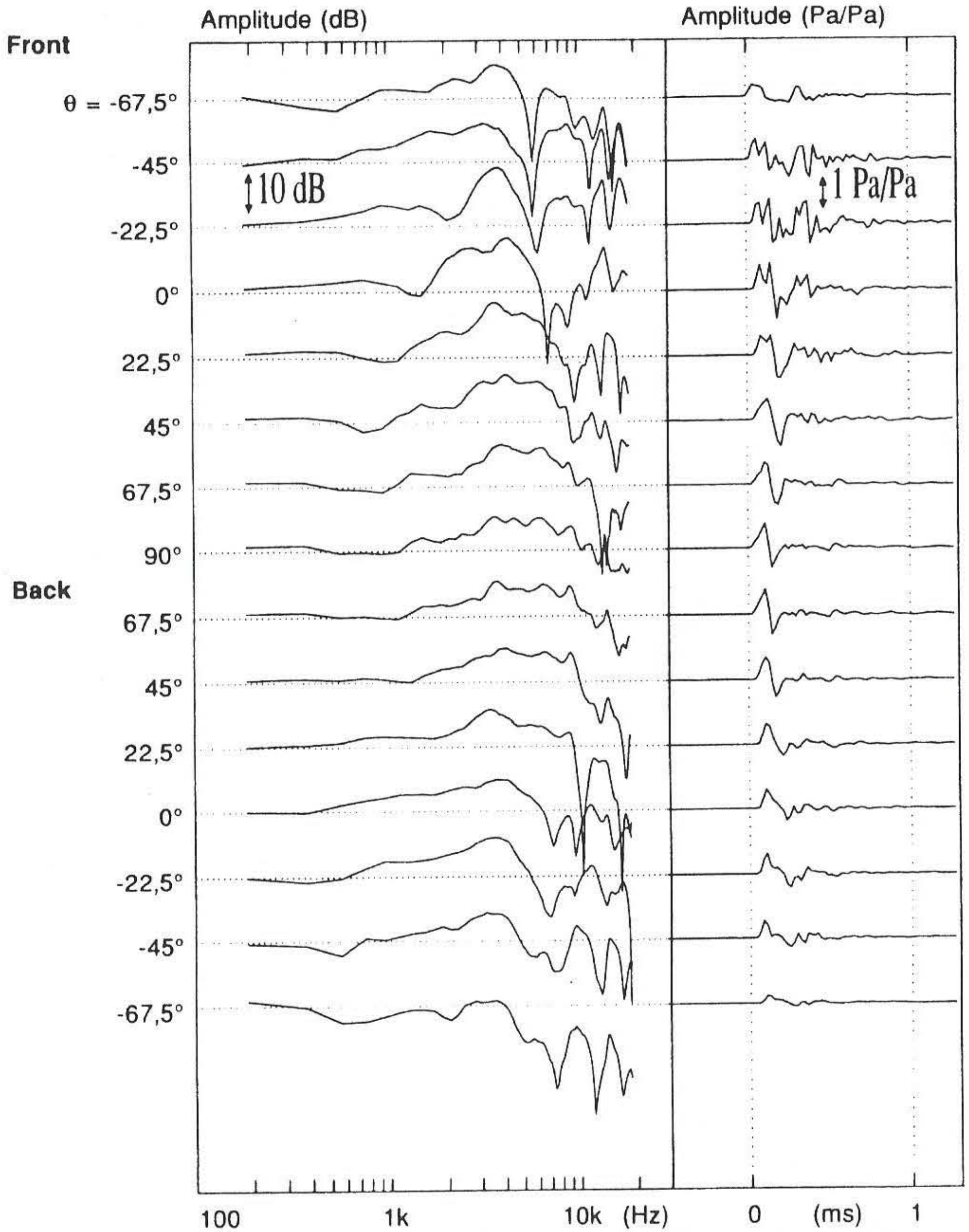


Figure 18. HTF and HIR shown for the left ear of one subject, covering the median plane. The horizontal dotted lines indicate 0 dB for each HTF.

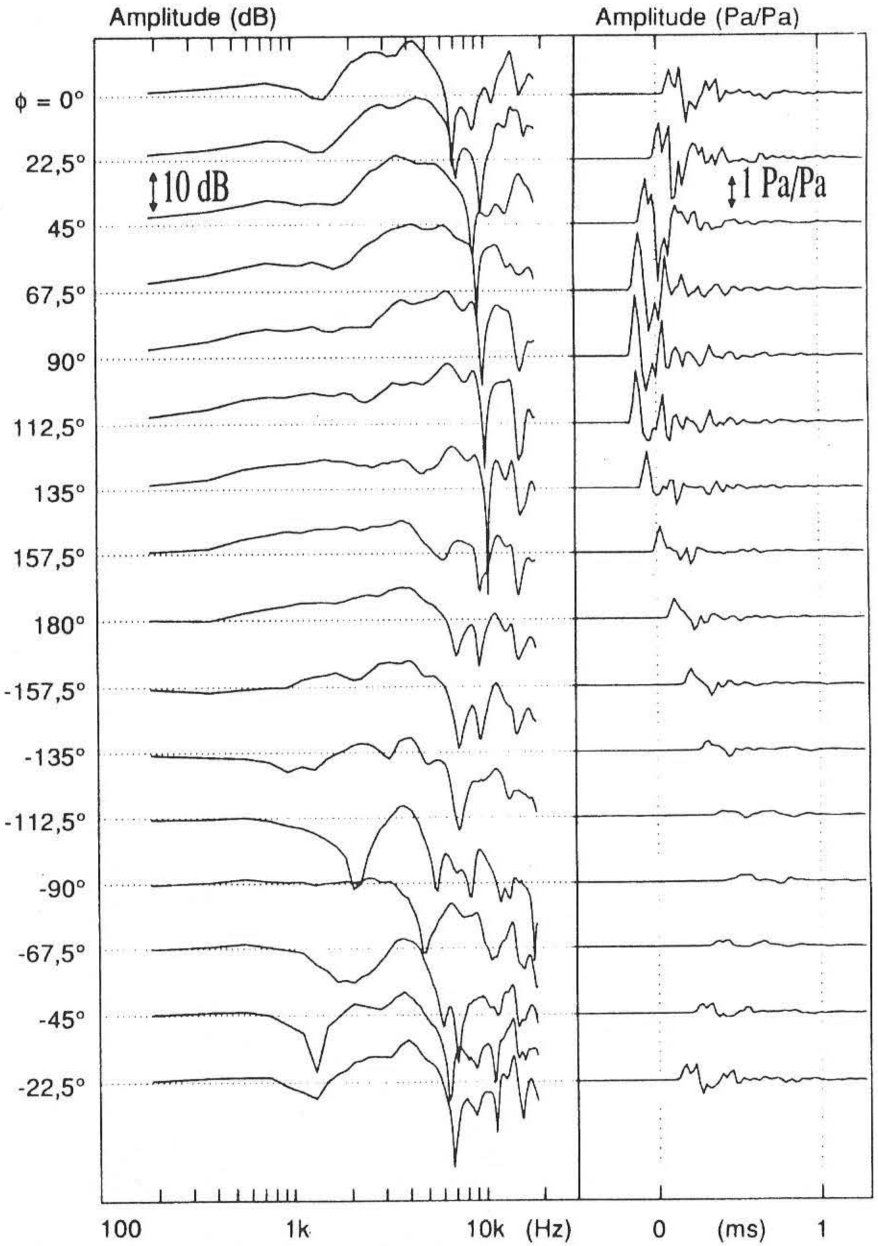
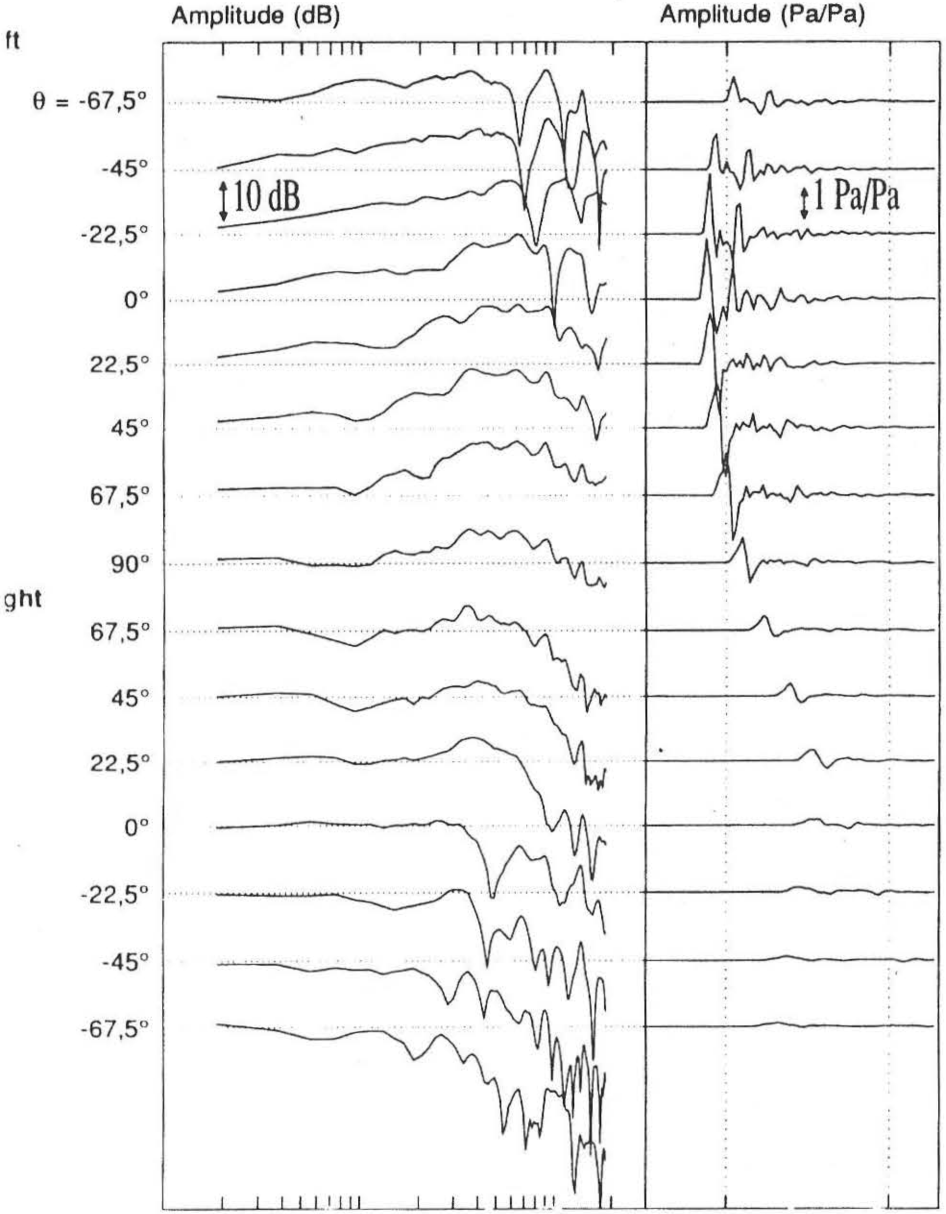


Figure 19. HTF and HIR shown for the left ear of one subject, covering the horizontal plane. The horizontal dotted lines indicate 0 dB for each HTF.



ıure 20. HTF and HIR shown for the left ear of one subject, covering the frontal .ne. The horizontal dotted lines indicate 0 dB for each HTF.

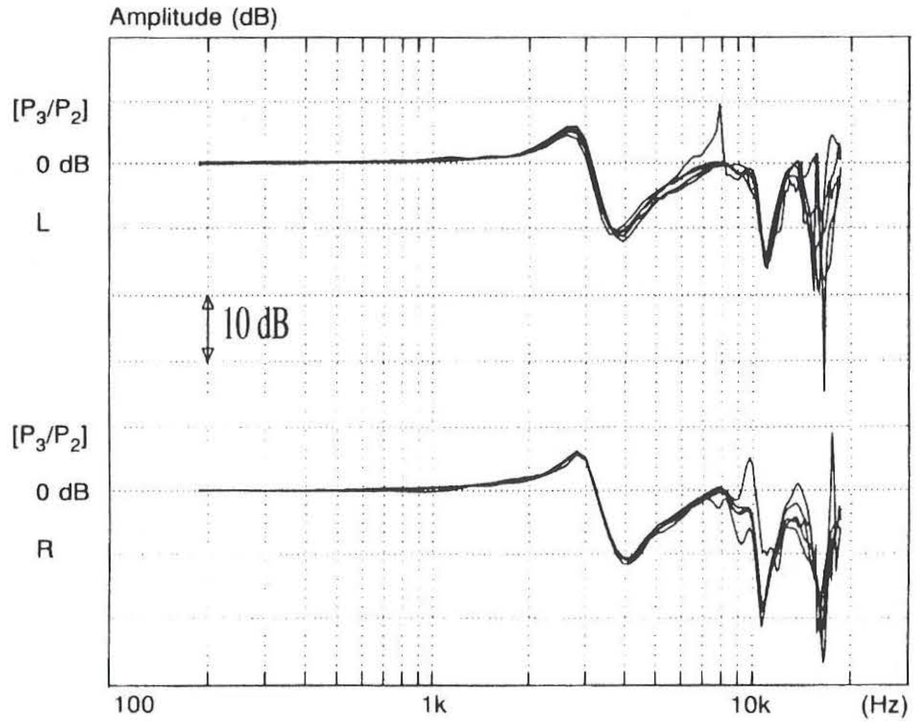


Figure 21. Example of the pressure division $[P_3/P_2]$ for one subject in the 5 directions measured: ϕ = 0°, 90°, 180°, and -90° in the horizontal plane, and for θ = 90°. Upper set of curves is left ear, lower set right ear.

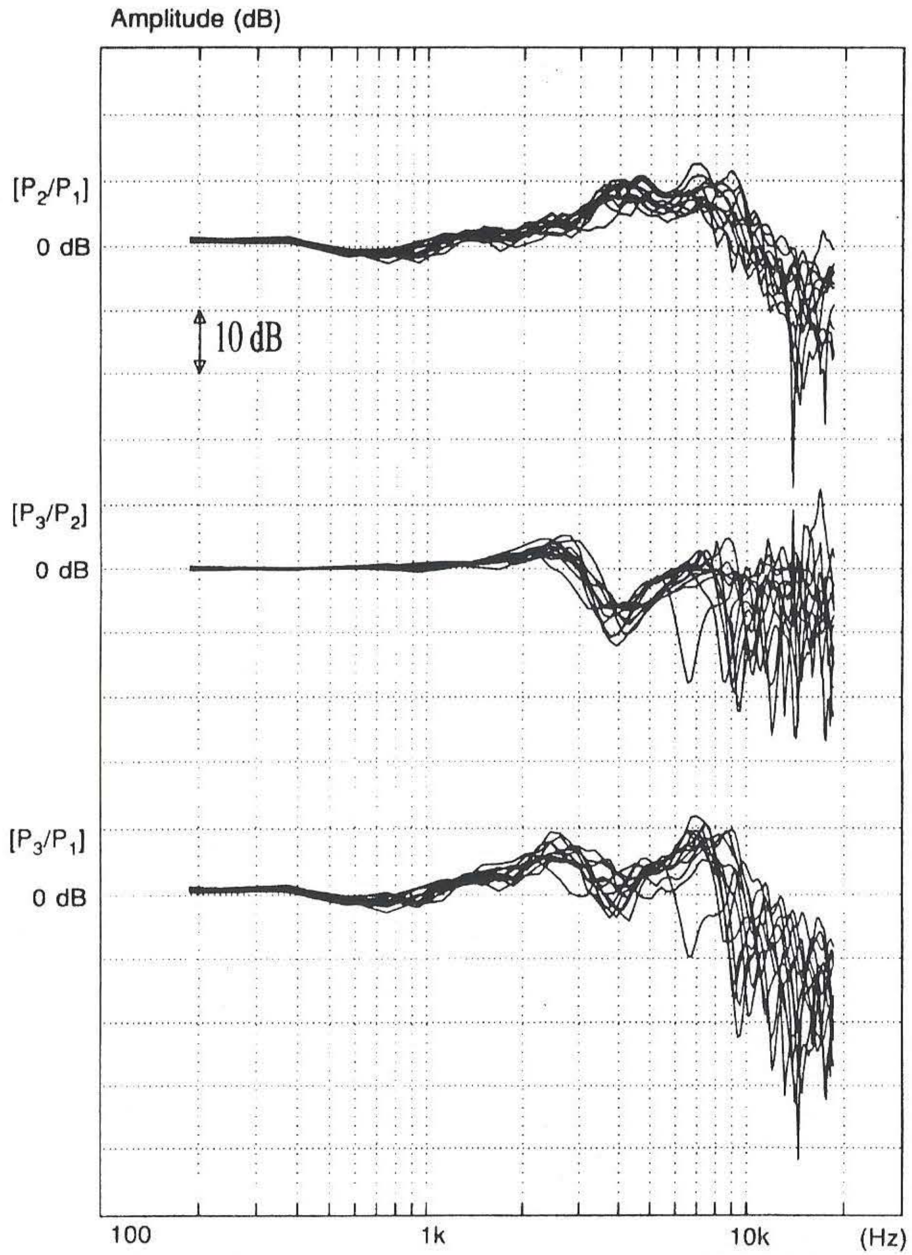


Figure 22. $[P_2/P_1]$, $[P_3/P_2]$, and $[P_3/P_1]$ for the left ear of 12 subjects in the direction $\theta = 90°$. All responses are based on measurements with the probe microphone.