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# Free-field sound transmission to the external ear; a model and some measurements

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## Introduction

In all applications of binaural technology the basic principle is to reproduce the true eardrum signal in a playback situation. Somewhere along the transmission path from a sound source to the sound pressure at the eardrum, the chain must be broken to record a signal. This signal must include all information needed to reproduce the original eardrum signal. The purpose of this work is, by modelling and measuring the acoustic system, to determine a point at which 1) sound can be recorded that includes all directional information 2) without being unnecessary individually influenced.

## Acoustic model

The endpoint of the acoustic system we are measuring is illustrated for one side in Figure 1. a) shows an anatomical sketch and b) is the analogue model used. A complete description of the model and terminology used here can be found in Henrik Møller [1]. With listener and sound source placed in a free field, $p_4$ is the resulting sound pressure at the eardrum. $p_3$ is the resulting sound pressure at any point inside or outside the ear canal, from which the transmission to $p_4$ is one-dimensional. One-dimensional transmission through an acoustic canal can be expected for wavelengths much larger than the diameter of the canal, and for typical ear canal dimensions, this means for frequencies up to 10 kHz. $p_2$ describes the "open circuit" pressure and forms together with $Z_{radiation}$ the Thevenin equivalent.

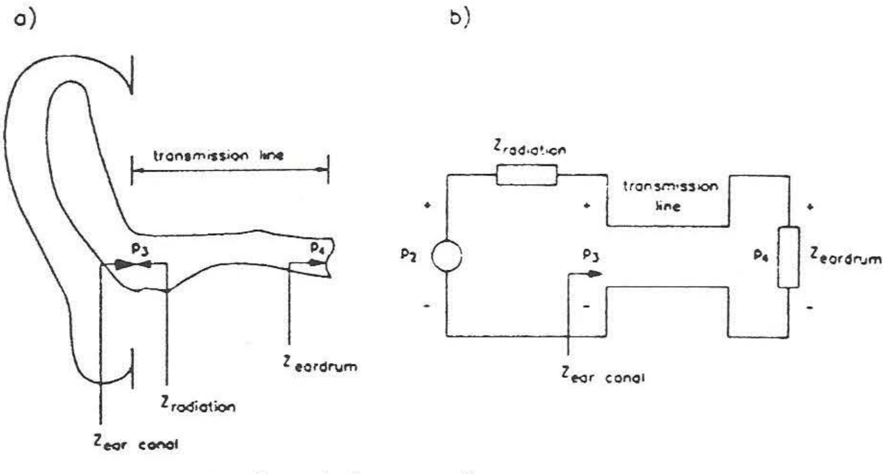


*Figure 1. Sound transmission through the external ear; a) sketch of the anatomy and b) an analogue model.*

$p_4$ and $p_3$ exist physically at listening situation, while $p_2$ does not. $p_2$ can be measured just outside the blocked ear canal, where $p_3$ and $p_4$ are absent. $p_1$ denotes the sound pressure at listening position, when the listener is absent. With these definitions, the transfer function, $[P_2/P_1]$, describes the transformation from 3- to 1-dimensional transmission, and is included both in $[P_4/P_1]$ and $[P_3/P_1]$.

**Measuring-setup and -procedure**

For all measurements the general measuring system known as MLSSA (Maximum Length Sequence System Analyzer) was used. A 4095 point Maximum Length Sequence at 50 kHz (81,9 ms) was used as stimulus. By recording the output of the acoustical system, the impulse response is found by a decorrelation of the MLS-sequence from the recorded signal.

Stimulus was fed into a power amplifier, where the output signal could be switched to one of three ball loudspeakers (membrane diameter 7,5 cm) in an-echoic chamber. The loudspeaker voltage was 1,87 $V_{RMS}$, which gave a free-field sound pressure of 75 dB(A) at the listening position. The signal was recorded by a B&K 4182 probe-microphone with a 7 cm soft, flexible tube (diameter 0,76 mm) mounted at the microphone tip. The microphone signal was amplified by a B&K 2607 measuring amplifier, and converted by the 12 bit A/D converter on the MLSSA-board. Each impulse response was calculated from the averaged result of 16 recordings, and the overall SNR of the measuring system was between 20 and 30 dB in the frequency range of 200 - 20 kHz. The major noise signal was leakage through the walls of the tube. For further analysis, all impulse responses were represented in frequency domain as 256-points Fourier Transforms (rectangular window) of the segment between 5,12 ms and 10,22 ms (samples 256 - 511), which included the entire response, and no reflections from floor etc. All free-field transfer-functions were found as one measured transfer-function referenced to another measured transfer-function, not measured at the same time, and not necessarily existing at the same time.

Subjects were placed in a hairdresser's chair with a neckrest to prevent head movements and with damping material on their knees. The probe microphone was fixed by a small racket on the chair, and was adjusted to lie close to the neck of the person. The length of the ear canal had been measured initially by a medical doctor, and marks where made on the tube to assure correct placement. The flexible tube was placed at the desired measuring point in the ear, and only the tube was displaced from measurement to measurement. All measurements (lasting 10 - 20 min. for a subject) were repeated with a coffee break (and replacement of the subject and probe) in between.

**SERIES 1: Where does the one-dimensional transmission start?**

To decide at which point the transmission through the external ear becomes one-dimensional, different measuring points for $p_3$ were tried, and the transfer function $[P_4/P_3]$ was examined for these different measuring points. For 4 subjects, distance = 2 m, elevation = $0^o$ and azimuth = $0^o$, $90^o$ and $180^o$ the following sound pressures were recorded:

| | |
|---|---|
| $p_4$ | 1-2 mm from the eardrum |
| $p_{3,1}$ | 12 mm within the ear canal |
| $p_{3,2}$ | 6 mm within the ear canal |
| $p_{3,3}$ | at the entrance of the ear canal |
| $p_{3,4}$ | 6 mm outside, following the ear canal in a horizontal line |
| $p_{3,5}$ | "caudal cavum conchae" |
| $p_{3,6}$ | "posterior cavum conchae". |
| $p_{3,7}$ | 3 - 4 cm outside the ear canal |
| $p_2$ | at the entrance of the blocked ear canal |

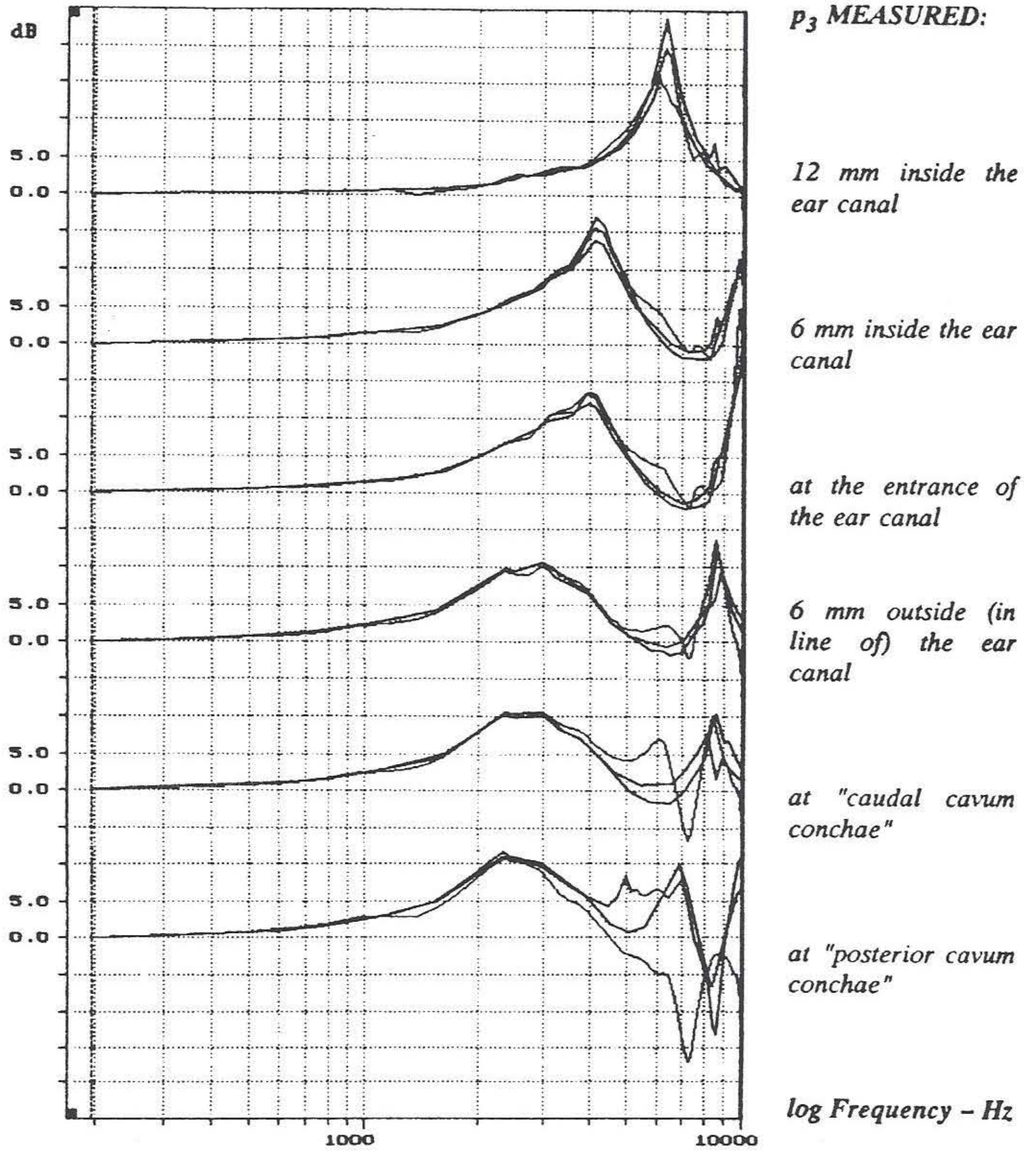


*Figure 2.* *$[P_4/P_3]$ sound transmission in the external ear. One subject, distance = 2 m, elevation = $0^o$, overlay of azimuth = $0^o$, $90^o$ and $180^o$ for each measuring point of $p_3$.*

Figure 2 shows the sound transmission $[P_4/P_3]$ for 6 different measuring points of $p_3$ for one person. Small differences for the three different directions can be observed, but the transmission through the ear canal, even from a point 6 mm outside, in line of the ear canal, seems to be one-dimensional. Both at "caudal cavum conchae" and "posterior cavum conchae" (which are both convenient measuring points), the transmission is directional-dependent.

**SERIES 2: How do the transmission elements look for several persons?**

To obtain additional information on how individual characteristics influence the different transmission elements in the model, the experiment was repeated for 8 more subjects, however only using two points for $p_3$, namely $p_{3,2}$ and $p_{3,3}$.

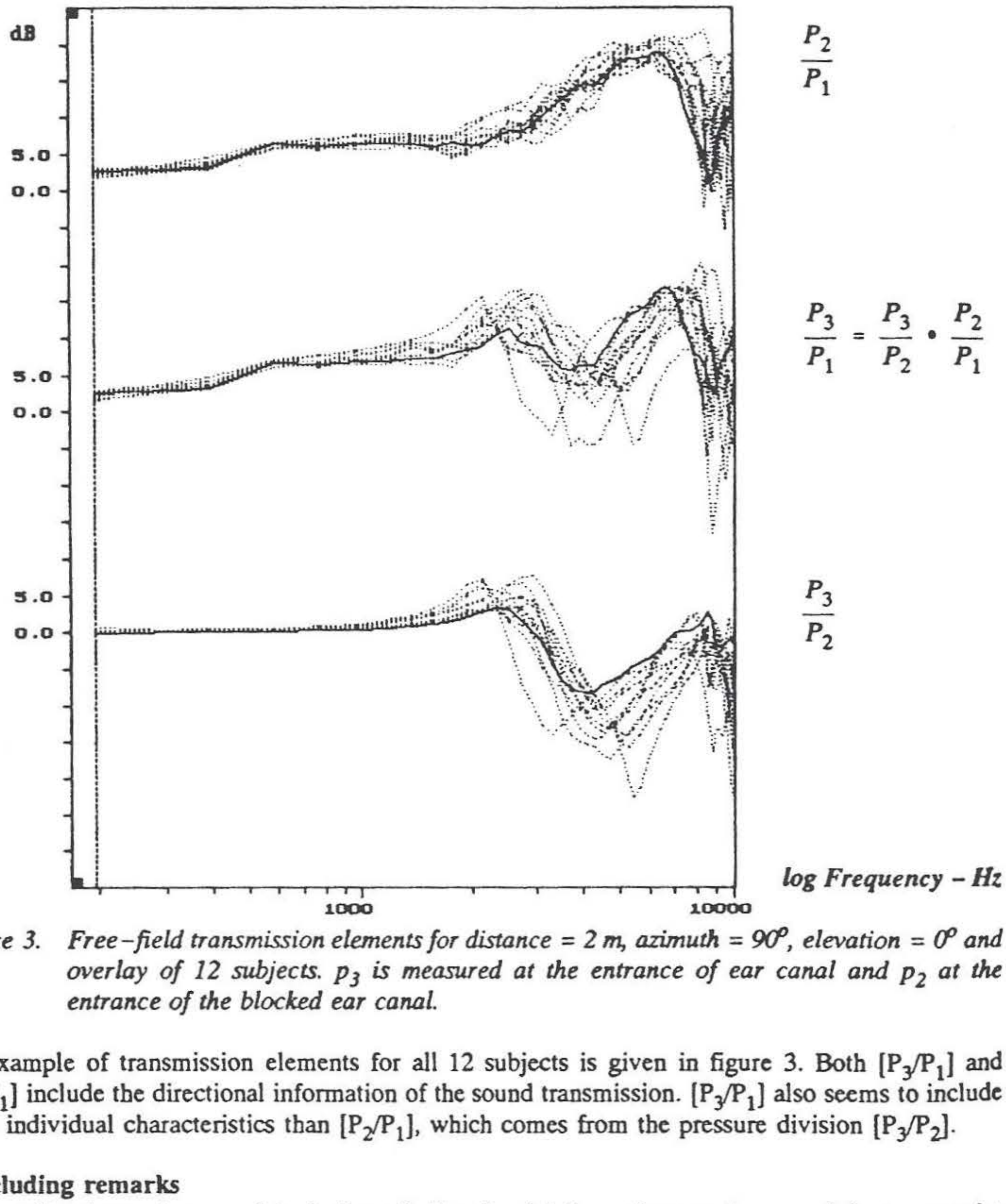


*Figure 3. Free-field transmission elements for distance = 2 m, azimuth = 90°, elevation = 0° and overlay of 12 subjects. $p_3$ is measured at the entrance of ear canal and $p_2$ at the entrance of the blocked ear canal.*

An example of transmission elements for all 12 subjects is given in figure 3. Both $[P_3/P_1]$ and $[P_2/P_1]$ include the directional information of the sound transmission. $[P_3/P_1]$ also seems to include more individual characteristics than $[P_2/P_1]$, which comes from the pressure division $[P_3/P_2]$.

**Concluding remarks**

The results show, that sound including all directional information can be recorded at any point within the ear canal (even at a point 6 mm outside the ear canal) blocked or un-blocked. It is also indicated, that $[P_3/P_1]$ and $[P_4/P_1]$ contains additional unnecessary individual information. Depending on the binaural application, errors are introduced due to different heads used for recording and playback. In some cases a third head is used for calibration, and in simulation situations a "fictive head" based on data can simulate the sound transmission paths. In each case it must be evaluated whether transmission elements with large individual differences are included, and if it can be compensated for - either by choosing another reference point or by additional equalization.

[1] Henrik Møller: "Binaural recording technique", Acoustics Laboratory, Aalborg University, 1991.