INTRODUCTION

Dummy-head systems are well suited for long-time acoustical measurements in the field of directional and human spatial hearing. In human localization the outer ears play a significant role. The sound waves reaching the eardrums are affected by directional filtering of the pinnae, head, and torso. This binaural filtering as well as the interaural intensity and time differences determine the perception of the direction of the sound sources [1] – [13]. The complex transfer functions of the outer ears are the head-related transfer functions (HRTFs). HRTF measurements and binaural recordings can be made on real human subjects or by using dummy heads [1] – [7], [14] – [17]. Head and torso simulators (HATS) are created to model the median human adult body geometrically. Binaural recordings may result in worse localization performance when dummy-head HRTFs are used rather than individuals [18] – [22]. If it is our goal to get precise data from long-time measurements, it is preferable not to deal with human subjects.

Eq. (1) defines the free-field complex HRTF,

\[
\text{HRTF}(\omega) = \frac{P_1(\omega)}{P_2(\omega)},
\]

where \( P_1 \) is the sound pressure at the eardrum and \( P_2 \) is the sound pressure at the origin of the head-related coordinate system for the same signal and sound source, but recorded with a unidirectional microphone [1]. The sound transmission in the ear canal is independent of direction [4] – [7], [23] – [25].

Our present investigation is concerned with determining variations in the HRTFs caused by small changes of the acoustical environment near the head [26], [27]. The first step was to install a system with increased signal-to-noise ratio (SNR) and good spatial resolution. Our updated system and methods are based on earlier measurements [28] – [30].

1 GENERAL SETUP

The HRTFs were measured using a Brüel & Kjaer head and torso simulator type 4128 placed on a turntable in a 125-m^3 anechoic room. The elevation of the sound source is adjustable in 5° steps by using strings between \(-45^\circ\) and \(+90^\circ\). For the proper setting of the source elevation a laser pointer is used based on geometrical calculations. A similar method was used by Gardner, who measured KEMAR HRTFs using a motorized turntable and ray projection from the center of the KEMAR face to set elevational positions in 5° steps [31].

Usually HRTFs are measured using impulse excitations or noise stimuli [32], [33]. We used a pseudorandom broad-band noise signal, and two channel responses were collected and averaged in a reference measurement. The effects of the undesired transfer characteristics in the measurement chain were eliminated by the reference signal and by calculating the HRTFs as usual,

\[
\text{HRTF}(\omega) = \frac{H_{\text{outerears}}(\omega)}{H_{\text{reference}}(\omega)}.
\]

The measuring software controls the turntable, delivers the stimuli from the DSP card, and stores the responses of both ears simultaneously with 50-kHz sampling frequency, 16-bit resolution, and 4096-point fast Fourier transform (FFT). A Brüel & Kjaer 2706 power amplifier and type 2636 measuring amplifiers were used. The reference signal was measured with the Brüel & Kjaer 4166 microphone. The validity of the HRTFs is above 200 Hz.

To increase the SNR a robust averaging procedure was
built in. If a deterministic stimulus is repeated periodically and the number of measurement frames are doubled, a +3-dB improvement in the SNR can be achieved by averaging. This is only valid if there are no time variance effects within a single measurement or between measurement frames [34]. Pseudorandom noise is a deterministic signal with the many advantages of a noise excitation. In contrast to a measurement system using impulse excitation, a greater SNR can be obtained when broad-band signals are used [35].

Averaging helps against random uncorrelated measurement noise. Our earlier measurements with this system had an average SNR of only about 60 dB [28]. In the literature of spatial hearing and acoustical measurements SNRs of 20–30 dB up to 60–70 dB have been reported [4]–[8], [31], [36]. With our measurement setup the average SNR was about 89 dB.

### 2 PSEUDORANDOM NOISE EXCITATION

For the measurement a pseudorandom noise stimulus is used. A broad-band noise excitation is advantageous, because more signal power can be produced than with impulse excitation. This means a greater dynamic range and higher SNR. The pseudorandom noise signals have the advantageous properties of white noise signals (they are generated by a random algorithm). Furthermore as deterministic signals (stored as numbers on the hard disk) they can be repeated exactly [37], [38]. This permits the use of repeated measurements and an increase in the SNR by averaging.

A frequency-independent SNR can be achieved if the spectrum of the stimulus “looks like” the spectrum of the system noise, that is, the higher the noise in a given frequency domain, the greater should be the energy contained in the stimulus.

To generate a noise signal, the maximum-length-sequence (MLS) technique is a widely used procedure through the fast Hadamard transformation [31], [34], [39] or using Golay codes [40]–[42]. An alternative way to generate a pseudorandom noise signal is presented here, and a similar signal is described in [43]. The generation of a pseudorandom phase spectrum with uniform distribution between $-\pi$ and $+\pi$ and a flat-magnitude response, 4096-point inverse fast Fourier transform (IFFT), and 44 100-Hz sampling frequency are used in the averaging of 100 frames versus uncorrelated noise elements and to increase the SNR. A different algorithm for noise signal generation with Gaussian magnitude distribution is described in [44].

Periodic signals of length $2^N$ are better for the FFT than binary MLS sequences, which usually are generated with an $N$-staged shift register and an XOR gate connected to each other, so they only have $2^N - 1$ states running through [33]. Nowadays sweep measurements are also preferred to MLS methods because they have a rather large SNR (about 90 dB) without the effects of loudspeaker nonlinearity and harmonic distortion.

The algorithm presented here was created in order to approximate the average power spectrum of the entire measurement system. To get this input information, we made repeated measurements with the system using zero excitation, and the averaging was done based on the signal power.

$$
\text{Re}_{\text{avg}}[i] = \frac{1}{N} \sum_{j=1}^{N} (\text{Re}_j[i] + \text{Im}_j[i])
$$

$$
\text{Im}_{\text{avg}}[i] = 0
$$

where $i = 0, \ldots , 2047$ and $N = 18$.

The measurement was made with a unidirectional microphone in the horizontal plane turning in 20° steps ($j = 1, \ldots , 18$). The algorithm will disregard the measured phase information.

The stimulus has to meet the following three requirements:

1) It has to be periodic, and the length of the period should exceed $T$, where $T$ is the effective length of the impulse response computed from the transfer function of the actual system.

2) The spectrum of the stimuli has to be a good approximation of the average noise spectrum of the system calculated above in General Setup. This results in a frequency-independent SNR.

3) The crest factor has to be small, near unity, because in this way the power of the stimuli can be at a maximum without any distortion or overload. Furthermore the quantization noise will be the smallest. The crest factor is defined as the ratio of the peak to rms voltage,

$$
\text{crest factor} = P_n = \frac{n_{\text{peak}}(t)}{n_{\text{rms}}(t)}.
$$

Here

$$
n_{\text{rms}}(t) = \lim_{T \to \infty} \left( \frac{1}{T} \int_{0}^{T} x^2(t) \, dt \right)^{1/2}
$$

where $T$ is the linear averaging time.

The crest factor indicates how much energy is lost using a signal compared to the ideal case of a stimulus whose rms value is equal to its peak value. Through normalization the maximum energy in a measurement can be extracted [33].

The exact mathematical solution of this problem is not known, but based on numerical analysis the following algorithm is suitable for generating a sufficient signal within a reasonable running time.

### 2.1 Algorithm

The time difference between the samples of the stimuli is

$$
t = \frac{1}{f}
$$

where $f$ is the sampling frequency, that is, 50 kHz. For the block length we have

$$
N > \frac{T}{t}
$$

It is comfortable to choose $N$ as the nearest power of 2 for a rapid FFT. In our case $N = 4096$, so the length of the
period of the stimuli is 81.92 ms.

1) To start take the average power spectrum \( \text{Re}_{\text{avg}} \) calculated in Eq. (3). Let the phase \( \text{Im}_{\text{avg}} \) be a random variable with uniform distribution over the interval 0, ..., \( 2\pi \). The IFFT of this spectrum satisfies the first two requirements but does not satisfy the third,

\[
\mathcal{N}(t) = \text{IFFT}\left\{ \text{Re}_{\text{avg}} + j\text{Im}_{\text{avg}} \right\}. \tag{9}
\]

2) Compute the crest factor and its square root in the time domain,

\[
P_a = \frac{n_{\text{peak}}(t)}{n_{\text{rms}}(t)} \tag{10}
\]

\[
Q_n = \sqrt{P_n}. \tag{11}
\]

3) If there are sample whose absolute values exceed

\[
Q_n^* = n_{\text{rms}}(t) \cdot Q_n
\]

reduce these samples to the \( Q_n \) value of Eq. (11) without affecting their signs.

4) Compute the FFT. The spectrum usually does not satisfy the second requirement. Normalize the spectrum so that its total power will be equal to the power of the spectrum at the start.

5) Let

\[
q_n = 20 \log Q_n. \tag{13}
\]

6) Compute the IFFT again, and repeat steps 2 to 5 until the crest factor seems to be small enough and the target spectrum is "close enough" to the starting spectrum.

It is not proven that using this procedure will result in the values converging to an optimal value, but the numerical tests suggest this. Setting up a mathematical thesis and demonstration seems difficult. We observed that during the iteration the \( P_n \) value is generally decreasing, but a temporary increase is possible. For a short running time a crest factor of 1.1 and a 0.2-dB deviation are obtainable. This signal is used for the measurements for recording the reference signal and the transfer characteristic of a loudspeaker.

3 TURNTABLE CONTROL

The best localization performance of a human is at frontal incidence in the median plane [1], [8], [45]–[47]. Subsequent to earlier psychoacoustic investigations we decided to use a spatial resolution of 1° in the horizontal plane while collecting the HRTFs. Our goal was to control the turntable in 1° steps with reproducible precision. The original motor in a Brüel & Kjaer type 3921 turntable was replaced with a small stepping motor, which can be controlled more precisely. Due to the gear ratio the motor makes 32 000 steps while the turntable turns 360°. This is equal to 88.88 steps per degree on average. The signal of a trigger switch of the turntable is used to synchronize the \( \phi = 0° \) azimuth. The measurements were always made in the same direction of turning.

The precision of the movement depends on the azimuthal resolution. Corresponding to the number of the steps of the motor, any arbitrary position can be set with a precision of 1/32 000, after synchronization. But if we are measuring an entire circle in the horizontal plane by an arbitrary resolution, the precision will decrease. One-degree steps can be made only with an average precision of 1/88.88. This corresponds to a satisfactory relative value of 1.14%.

4 CONCLUSIONS

The HRTF database was recorded in the anechoic room using a dummy head placed on a turntable with the following properties. The data for the measuring system are given in Table 1. The spatial resolution is 5° from \(-45°\) to \(+90°\) in the median plane and 1° in the horizontal plane. The direction of the sound source can be set with a precision of 0.8% elevational and 1.14% azimuthal, calculated for a 1.8-m source distance. The signal processing includes a sampling frequency of 50-kHz, a 16-bit resolution simultaneously in two channels, and a 4096-point FFT. Any random noise effects are decreased during long-time averaging. The overall SNR of about 89 dB is frequency independent.

The applied non-MLS pseudorandom noise stimuli can be used with this system for all transfer function measurements. The algorithm is able to generate the proper stimuli easily and quickly for any system other than ours. A detailed description of the experimental apparatus and the measurement results are given in [30], and [48].

5 ACKNOWLEDGMENT

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6 REFERENCES


[5] C. B. Jensen, M. F. Sørensen, D. Hammershøi, and


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### Table 1. Data sheet of measurement system.

<table>
<thead>
<tr>
<th>Source</th>
<th>Elevation</th>
<th>Azimuth</th>
<th>Distance</th>
<th>Signal processing</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>[-45°, +90°], 5° steps, 0.78% precision</td>
<td>[0°, 360°], 1° steps, 1.14% precision</td>
<td>1.8 m</td>
<td>Sampling frequency</td>
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<td>Conversion</td>
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<td>Transfer function</td>
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<td>Validity of transfer function</td>
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<td>Overall SNR</td>
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<td>Deviation in transfer functions</td>
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<td>Constant within ±6°</td>
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<td>Non-MLS 81.92-ms pseudorandom noise</td>
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<td>Above 200 Hz</td>
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<td>&gt;89 dB, independent of frequency</td>
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<td>&lt;0.5 dB</td>
</tr>
</tbody>
</table>


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