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Impulse Measurements of Headphones on Ear Simulators and on Head and Torso Simulators

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Summary

In this study, a method for measuring the impulse responses of headphones on an ear simulator and on several head and torso simulators (HATS) is described. The method is based on maximum-length sequences with a period of 16383 and on the Hadamard transformation. The latter procedure is used for the correlation of the measured sequence with the maximum-length sequence to calculate the impulse response, which takes only 7 s on IBM-AT-compatible personal computers. The s/n ratio is increased by 42 dB compared with a measurement with a single pulse. Four different types of headphones were measured on three types of HATS. The impulse responses were transformed into the frequency domain using a fast Fourier transformation. Free-field sensitivities and coupler sensitivities were determined. The results agree very well with the results obtained with other measurement techniques using other signals (e.g. one-third octave filtered random noise). In particular, the differences from results of very precise measurements with pure-tones are less than a few hundredth parts of a decibel. The measured impulse responses can be used in digital filter systems.

Impulsmessungen von Kopfhörern an Ohrsimulatoren und an Kunstköpfen

Zusammenfassung

In dieser Arbeit wird eine Methode zur Messung von Impulsantworten von Kopfhörern an künstlichen Ohren und an Kopf- und Rumpfsimulatoren (Kunstköpfen) beschrieben. Die Meßmethode beruht auf Maximalfolgen mit einer Länge von 16383 und auf der schnellen Hadamard-Transformation. Letztere wird zur Berechnung der Kreuzkorrelation der gemessenen Sequenz mit der Maximalfolge benutzt, um die Impulsantwort zu berechnen. Dies dauert nur etwa 7 s auf IBM-AT kompatiblen Personal-Computern. Der Signal-Rauschabstand wird bei dieser Methode gegenüber dem einer Einzelimpuls-Messung um 42 dB erhöht. Vier verschiedene Kopfhörer wurden an drei Kunstköpfen gemessen. Die Impulsantworten wurden mittels FFT in den Frequenzbereich transformiert. Es wurden Freifeld-Übertragungsmaße und „Kuppler"-Übertragungsmaße bestimmt. Die Ergebnisse stimmen sehr gut mit denen anderer Meßtechniken überein (z. B. mit terzgefiltertem Rauschen). Insbesondere die Abweichungen zu Ergebnissen aus Präzisionsmessungen mit Sinustönen liegen bei einigen hunderstel Dezibel. Die so gemessenen Impulsantworten können in Systemen zur digitalen Filterung verwendet werden.

Mesures de réponses impulsionnelles d'écouteurs sur oreille artificielle et sur têtes de mannequins

Sommaire

Dans cette étude, on décrit une méthode de mesure des réponses impulsionnelles d'écouteurs sur oreille artificielle, et sur des ensembles imitant la tête et le torse humains (HATS). La méthode est basée sur l'utilisation de suites maximales d'une longueur de 16383, et sur la transformation rapide d'Hadamard. Cette transformation sert au calcul des réponses impulsionnelles à partir de la corrélation entre les séquences mesurées et les suites maximales. Cela ne prend que 7 s sur un mini-ordinateur de type IBM-AT compatible. Par rapport à une mesure faite sur une impulsion unique, le rapport signal/bruit est augmenté de 42 dB. On a examiné quatre types d'écouteurs différents, sur trois types de mannequins HATS. Une transformation de Fourier rapide des réponses impulsionnelles dans le domaine des fréquences permet de déterminer les fonctions de transfert en champ libre et sur coupleur. Les rèsultats sont en trés bon accord avec ceux obtenus par d'autres techniques; celle utilisant des bruits filtrés par tiers d'octave par exemple. En particulier, les différences d'avec les mesures de précision en sons purs sont de l'ordre de quelques centièmes de dB. Il est possible d'utiliser les réponses impulsionnelles ainsi mesurées dans des systèmes de filtrage numérique.

1. Introduction

Exact measurements of the transfer functions of headphones are of fundamental importance in audiometry. At the moment, these calibration measurements are mainly performed using an analogue technique with pure-tone signals. The phase response of the headphones is not taken into account. The headphone calibrations could be expanded if an impulse measurement technique was applied. However, if it is to be used for calibration purposes, the impulse technique should be as accurate as the conventional measurement technique. An improved method for measuring

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headphones is needed for the description and calibration of signals of short duration (clicks) for objective audiometry (electrical response audiometry ERA [1]). The transient behaviour of the headphone has a crucial influence on the shape of the click. Thus the impulse response of the headphone mounted on an ear simulator must be measured.

Besides the field of audiometry there is, of course, an interest in measurement techniques with impulses for the development and construction of headphones for the reproduction of music and speech. For these applications free-field sensitivities (or diffuse-field sensitivities) must be determined on head and torso simulators or on test subjects.

2. Measurement method

The method described in the following is applicable to linear time invariant (LTI) systems. Since measurements of LTI systems with a single pulse have the well-known disadvantage of a low signal to noise ratio caused by the limited maximum amplitude of transducers, another signal is used. As a result of these limited amplitudes an increase in the effective signal intensity is not achieved by simple amplification but by temporal spreading with subsequent additional signal processing. The signal used can be a maximum-length sequence $m(t)$ [2]. It is a pseudo-random, periodical (period N) and deterministic binary sequence with an autocorrelation function $\Phi_m(t)$, which is a very good approximation of the N periodical sequence of Dirac pulses. The increase in signal energy compared with a single pulse is $10 \lg N$ dB.

Let us assume that the impulse response $h(t)$ of a linear time invariant system is to be measured. During a measurement we have first a measured sequence $s(t)$ which is the convolution of the impulse response $h(t)$ of the measured system (in this case the headphone on the ear simulator) with the input maximum-length sequence. The impulse response of the system is calculated by cross-correlating the measured sequence with the input sequence. The cross-correlation can be replaced by a convolution with the time-inverted maximum-length sequence.

As a result of the periodicy of $\Phi_m(t)$, $h(t)$ must fade away within the measuring period to avoid "time-aliasing". Thus for one period we have ("$*$" denoting convolution):

$$s(t) = h(t) * m(t) \tag{1}$$

$$s(t) * m(-t) = h(t) * m(t) * m(-t) \tag{2}$$

$$= h(t) * \Phi_m(t) \approx h(t), \tag{3}$$

if N is large.

The procedure of convolution or cross-correlation can be programmed as a discrete algorithm, as a fast Fourier transformation and multiplication in the frequency domain, or as the Hadamard transformation, which is the most efficient method [3], [4], [5], [6]. In this study a period $N = 16383$ is used. The increase in signal energy is 42 dB, and is obtained at the expense of the performance of the Hadamard correlation procedure, which takes 7 s on IBM-AT-compatible personal computers. With a clock frequency of 50 kHz the measuring time is 328 ms. In comparison, the impulse response of a headphone on an ear simulator or a manikin lasts a few milliseconds at the most, so that the time-aliasing is negligible.

The measurement technique with maximum-length sequences and Hadamard transformation is already established, and first versions of its hardware and software are available commercially. These will certainly soon be improved and made faster to make real-time applications possible. More information on the Hadamard transformation is given in [4], and information on the measurement system used here, including the noise generator, A/D converter and software, is given in [5].

3. Headphone measurements on an ear simulator

Fig. 1 shows the elements of the measuring equipment including a head and torso simulator (manikin) fitted with a headphone. The manikin can be replaced by an ear simulator. The maximum-length signal generator (Gen.) also creates clock and trigger signals for the A/D converter card (A/D) of the personal computer. The trigger signal is used for marking the beginning of each sequence to enable group delay measurements to be carried out. Two channels of the A/D converter card are fed simultaneously with the input voltage of the headphone (here called the U signal) and with the output signal of the ear simulator (called the p signal

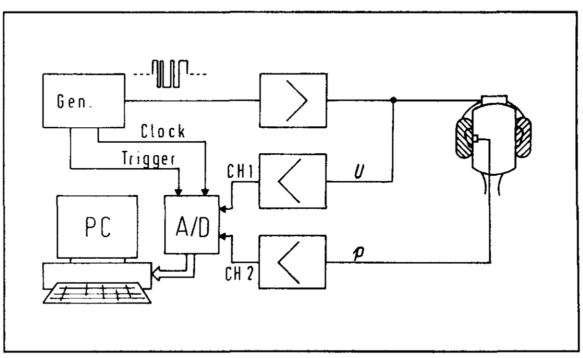

Fig. 1. Elements of the measuring arrangement.

specifying a sound pressure signal). Thus the frequency response and the sensitivity (in dB re 1 Pa/V) of the headphone are directly known by Fourier transformation and complex division of the p spectrum by the U spectrum. A 4096 point FFT algorithm was used. The frequency curve of the pressure sensitivity of the ear simulator microphone is assumed to be flat (see section 4).

The following headphone types were used in this study:

a) Beyer DT 48, audiometric headphone (supra-aural, closed, flat cushions),
b) Sennheiser HD 430, Hi-Fi headphone (circum-aural, open),
c) Sennheiser HD 50, Hi-Fi headphone (supra-aural, open),
d) Sony MDR-E464(B), Hi-Fi headphone (intra-concha, open).

The following ear simulators or head and torso simulators were used.

a) artificial ear, Brüel & Kjær type 4153 (IEC 318 [7]),
b) manikin, type KEMAR, Knowles electronics,
c) manikin, Brüel & Kjær type 4128,
d) manikin, type HMS II.4, HEAD acoustics.

For these measurements, all manikins were equipped with occluded ear simulators according to [8]. As described in [9], they fulfil the most important requirements of the only existing standards concerning HATS in particular, the technical report IEC 959 [10] and the standard ANSI S. 3.36 [11].

3.1. Calibration of audiometric headphones

An audiometric headphone, Beyer DT 48, was mounted without cushions on a B & K 4153 artificial ear using an adapter ring rather similar to that described in [12]. The artificial ear B & K 4153 includes a half-inch microphone. Originally, the DT 48 headphone had to be calibrated on an ear simulator according to IEC 303 (i.e. B & K 4152) which is equipped with a one-inch microphone. However, in our case a more broadband type of ear simulator was preferred to enable measurements of short impulses (see below) to be performed.

The headphone was measured with the impulse measurement technique described above. From experience of repeated measurements with removal of the headphone, it is known that the maximum spread of the results with this combination of headphone and ear simulator is $\pm$ 0.04 dB in a frequency band from 63 Hz to 12.5 kHz. Fig. 2 shows the impulse response (a) and the frequency response (b) of the DT 48 headphone calculated by a 4096 point FFT. For absolute

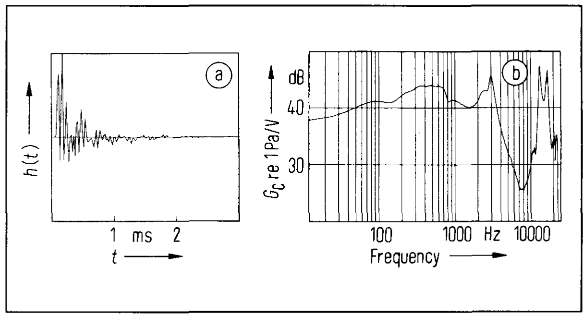

Fig. 2. a) Impulse response $h(t)$ and b) frequency response G_c of a Beyer DT 48 headphone (without cushions) mounted on a B & K 4153 "artificial ear" with ring adapter (rather similar to standardized method in [12]). Deviations from results of highly precise calibrations of headphones with pure-tone signals are 0.05 dB at the most between 100 Hz and 4 kHz.

calibration, the sensitivity was measured at 250 Hz with a pure-tone signal using a B & K 4220 sound calibrator (pistonphone). The frequency response was then shifted to fit the calibration level at 250 Hz.

To perform a comparison, another measurement technique based on pure-tone signals with analogue narrow-band analysis was applied using the same ear simulator and headphone. A comparison of the sensitivity levels of the two techniques yielded $\pm$ 0.05 dB, at the most, within a frequency range from 100 Hz to 4 kHz. This difference includes the uncertainty caused by removal of the headphone from the ear simulator. Under these circumstances the differences between the two measurement techniques are impressively small. Above 4 kHz the difference increases to 1.5 dB at the most. This difference could not be immediately explained, but it later turned out that the increasing uncertainty towards higher frequencies is mainly due to differences between the two amplifiers of the U and the p signals (see Fig. 1). Of course, this uncertainty could easily be avoided in several different ways.

Excluding the time for mounting the headphone on the coupler, the total measuring time using pure-tone signals is at least 2 min. Using the broad-band measurement method with maximum-length sequences the measuring time is approximately 15 s, including sampling of the signal and the subsequent signal processing by the Hadamard transform and FFT.

3.2. Transmission of ERA-clicks

A knowledge of the impulse response allows further evaluations to be made. If a rectangular "reference click", as described in a standard draft for the calibration of electrical response audiometers [1], is fed to the headphone, the ear simulator output signal $r(t)$ of course is not a simple rectangle in shape, but has a time response that is extremely difficult to describe mathematically. According to the theory of linear time invariant systems, the output signal $r(t)$ is expected to be the convolution of the input (rectangular) signal with the impulse response $h(t)$ (see the result in Fig. 3a).

To check the theory, a measurement with a real rectangular click was carried out. Fig. 3b shows the result of a measurement using the real signal from a pulse generator, and a digital storage oscilloscope for receiving the ear simulator response. If we compare the curves of Fig. 3, again a very good agreement can be observed. Hence the validity of applying the LTI system theory to headphones on ear simulators is evident. Checking the validity of the measurement technique must be taken very seriously, since the requirements of linearity and time invariance must be fulfilled to avoid artifacts [6].

Yet there are still unsolved questions with respect to standardization of calibration techniques. How to describe the signal shape shown in Fig. 3? Can an optimum signal shape or at least an optimum signal flank be found for ERA in clinical work? One step towards finding satisfactory answers to these questions is to consider digital filtering. If a special signal shape is required, digital filtering is the best way of forming well-defined signals.

By subsequently applying an FFT, complex division and the inverse FFT, it is possible to calculate a time response which has a reciprocal complex spectrum of the form of the impulse response shown in Fig. 2. This response can be included in the signal path by means of a digital (finite impulse response filter to equalize the frequency response of the headphone. This method worked very well for loudspeakers ([13], even in real-time [14]). In our case the finite impulse response filter was inserted between the pulse-generator and the headphone. The transmission of a corrected rectangular pulse was then measured as shown in

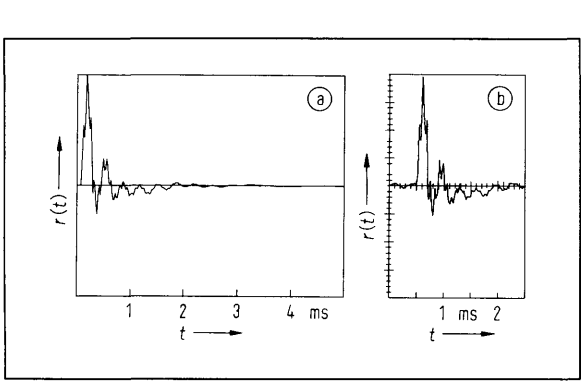

Fig. 3. Reproduction $r(t)$ of a 100 μs rectangular click from a DT 48 headphone on an ear simulator. a) calculated by convolution, b) measured with a digital storage oscilloscope.

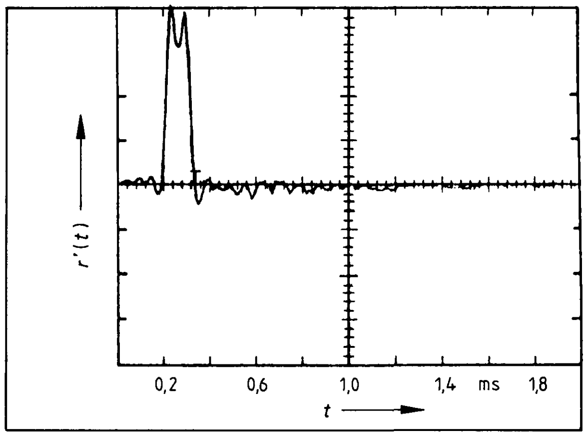

Fig. 4. Reproduction $r'(t)$ of a 100 μs rectangular click from a DT 48 headphone on a ear simulator with an FIR filter measured with a digital storage oscilloscope.

Fig. 4. This pulse serves well for use as a reference. Perhaps it is also advantageous in practical work.

It is true that there is not evidence indicating a need to equalize audiometric headphones with digital filters correcting amplitude and phase distorsions, but it seems worthwhile to investigate this field. Improvements could be achieved in speech audiometry, for masking noise spectra in bone conduction audiometry, and probably in the transmission of ERA clicks in clinical work. Of course, an equalization of headphones is extremely important if HATS recorded spatial sound (music, speech, noise, etc.) is to be reproduced naturally for hearing tests of for domestic use, perhaps even individually for each person.

4. Measurements of free-field sensitivities of headphones with head and torso simulators and test subjects

The free-field sensitivity of a headphone is defined as the voltage to sound pressure transducer sensitivity in units of a free sound field [15]. For instance, it can be measured using a subjective method by adjusting loudness levels of headphones in comparison with the loudness of a loudspeaker in a free field. If a headphone is equalized in free-field units, its free-field sensitivity is flat and its colouration to sound signals is equal to the colouration caused by the filtering and the diffraction of sound at ear canal, pinna, head and torso. In speech audiometry, the free-field sensitivity of headphones is used to equalize the speech signal for realizing natural conditions of speech perception.

In the free-field sensitivity G_{FF} of a headphone is measured by an objective method, the ear simulator should be replaced by a manikin or by a test person fitted with probe microphones. The headphone sensitivity must first be measured as described above for coupler sensitivities. However, besides this the frequency response of the headphone H_H must be divided by the free-field response (or better, the complex free-field transfer function H_{FF}) of the manikin used. Thus $G_{FF} = H_H/H_{FF}$ is to be measured.

The free-field transfer function of a manikin or a test person is defined as the frequency-dependent transformation of sound pressure from the free field to the eardrum. For detailed information on H_{FF} see the measurements and requirements in one-third octave representation in [9], [10], [11]. In this study, the transfer function H_{FF}, was measured using the impulse technique in an anechoic chamber and stored in the computer before starting the headphone measurements. An example is shown in Fig. 5.

In H_{FF} of Fig. 5 the transfer function of the loudspeaker H_{LS} (and other electrical elements of the equipment) was eliminated using a reference measurement with a free-field microphone (with microphone transfer function H_{FM}) prior to the manikin test.

When headphones were measured on test persons, the latter were fitted with probe microphones (cylindrical electret type, diameter 5 mm, length 7 mm). The method of measuring and evaluating the free-field sensitivities of the headphone is similar to the method described above for the headphone test on manikins. The influence of the position of the probe microphone in the ear canal is eliminated when calculating the free-field sensitivity of the headphone. The measurements are performed in three steps:

1.) Measurement of the headphone fitted to the manikin or test person as a function of the frequency f. The result is the "coupler sensitivity" $A(f)$ (the eardrum sound pressure of the manikin or ear canal sound pressure of the test person divided by the headphone voltage (see Fig. 1). $A(f)$ includes the microphone sensitivity H_{MM} of the manikin microphone or of the probe microphone.

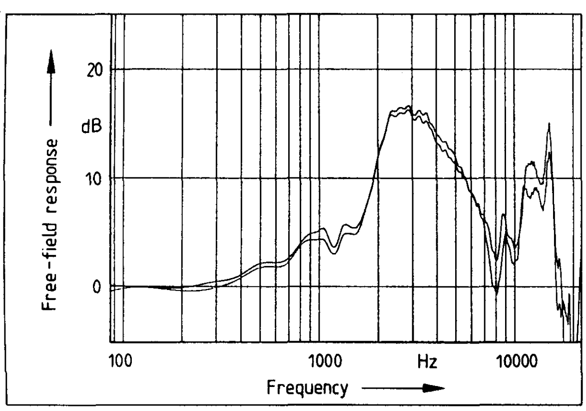

Fig. 5. Free-field response of the HATS B & K 4128 (right and left ear).

2.) Measurement of the function $B(f)$ of the sound pressure of the free sound field to the "eardrum" of the manikin or into the ear canal of the test person. With regard to test persons the probe microphone must not be moved during measurements 1.) and 2.). $B(f)$ includes the microphone sensitivity H_{MM} of the manikin microphone or of the probe microphone, and the characteristic of the loudspeaker H_{LS}.

3.) Reference measurement of $C(f)$ (frequency curve of the sound pressure at the reference point (position of the head) with a free-field microphone, loudspeaker reference measurement). No manikin or test person is in the sound field. Besides the loudspeaker characteristic, $C(f)$ includes the sensitivity of the free-field microphone H_{FM}.

According to eqs. (4) to (6), the frequency curves of the sensitivities of the manikin microphones or the probe microphones H_{MM} are eliminated by dividing the results of the measurements of the headphone $A(f)$ (when fitted to the manikin or test person) by the results $B(f)$ of the measurements of the manikin (or test person) in the free field.

$$\begin{array}{llccc} \text{Step:} & 1.) & 2.) & 3.) & \\ & \downarrow & \downarrow & \downarrow & \\ G_{FF}(f) = & A(f) & \dfrac{1}{B(f)} & C(f) & (4) \\ & = H_{MM} H_H & \dfrac{1}{H_{LS} H_{MM} H_{FF}} & H_{LS} H_{FM} & (5) \end{array}$$

Thus,

$$G_{FF} = H_{FM} H_H/H_{FF}. \quad (6)$$

In both cases, in measurements on manikins and on test subjects, the resulting frequency curve of the headphone free-field sensitivity G_{FF} is correct, provided that the free-field sensitivity H_{FM} of the microphone (Brüel & Kjær type 4165) used in 3.) is flat, and that the test person's probe microphone has not been moved during the measurements 1.) and 2.).

In all the results obtained, it is assumed that the frequency curve of the free-field microphone is flat. This is valid with an uncertainty of less than $\pm$ 0,2 dB within a frequency range up to 3 kHz and within $\pm$ 1 dB up to 16 kHz. It is impossible to take the microphone characteristic more accurately into account until a broad-band primary microphone phase and amplitude calibration can performed instead of the pure-tone reciprocity calibration now used. If sensitivities at fixed frequencies (i.e. typical frequencies for audiometry) are to be determined and if the microphone sensitivity is known from primary pure-tone reciprocity calibration, the frequency response of the

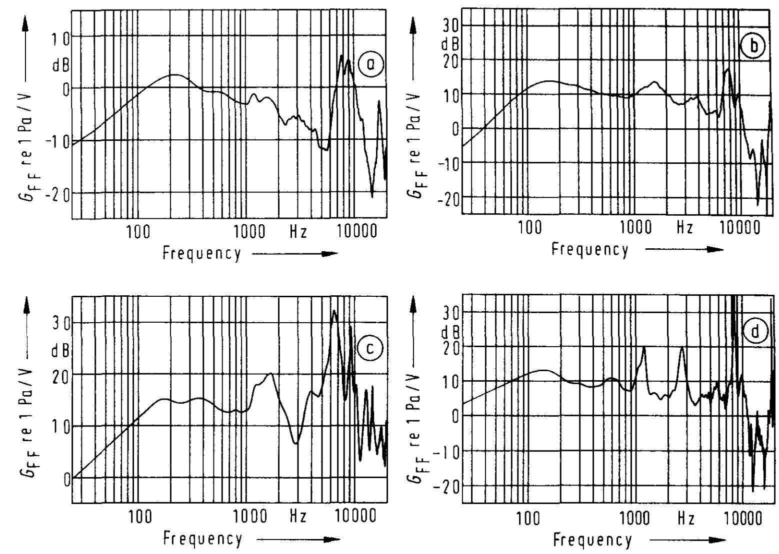

Fig. 6. Examples of free-field sensitivities G_{FF} of headphones (only left side). a) Sennheiser HD 430 on B & K 4128, b) Sennheiser HD 50 on KEMAR, c) Sony MDR-E464(B) on HMS II.4, d) HD 50 on a test person (one individual).

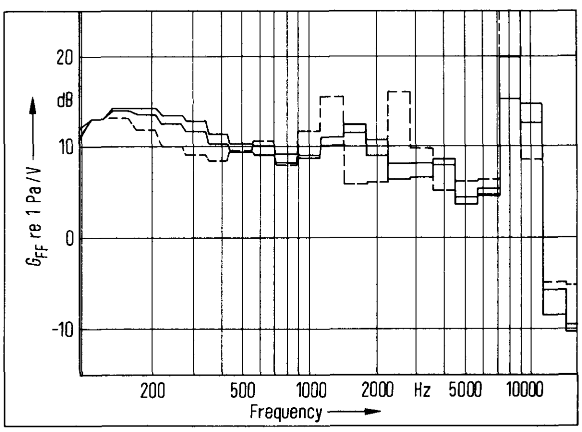

Fig. 7. One-third octave filtered free-field sensitivity G_{FF} of Sennheiser HD 50 (only left side) on KEMAR respectively on B & K 4128, dotted line on test subject (one individual).

microphone can be accounted for by correction values. The same applies to measurements of headphones on ear simulators.

Free-field sensitivities of open-type headphones are shown in Figs. 6 and 7. These types were chosen as examples, since the results achieved with manikins agree well with corresponding results achieved with test persons [16]. For instance, the curves in Fig. 7 can be compared with curves KHM in Fig. 5 of [16]. It shows the free-field sensitivities from one-third octave noise measurements of the headphone Sennheiser HD 50 on manikins KEMAR and B & K 4128 and on test persons. With regard to the manikins, the difference between the two measurement methods is $\pm$ 0.5 dB at the most, within a frequency range from 100 Hz to 6.4 kHz. This is approximately the uncertainty due to removing the headphone from the manikin and re-measuring with one-third octave noise. No additional uncertainty appears to result from the application of the impulse method.

All curves in Fig. 6 show much more detail than can be found in conventional one-third octave representations. A comparison of Fig. 6 b) and d) reveals differences in the results on HATS and on individual test subjects. Fig. 7 indicates that the individual frequency response of the headphone on a test person may differ by 10 dB from the curve obtained with HATS, even in a one-third octave plot. The reason for this is assumed to be the averaged simulation of the ear canal and eardrum impedance by the occluded ear simulator. The device described in IEC 711 [8] is intended to be an average ear simulator without representing any individual. The standard deviation in results from measurements with test persons are from 3 dB up to 6 dB for frequencies from 1 kHz to 10 kHz [16]. This is due to anatomical differences between humans.

5. Conclusion

In this study, it has been shown that the impulse measurement technique with maximum-length sequences is as accurate as conventional precision measurement techniques with pure-tone or narrow-band signals. For some applications the use of measurement techniques that obtain more detailed information and especially more resolution in frequency curves than one-third octave spectra is recommended. At least one-sixth octave spectra are preferable for the evaluation of frequency responses of headphones or head and torso simulators for frequencies above 8 kHz. Resonances with steep flanks and high peaks may occur which cannot be interpreted by a coarse frequency resolution.

The effects of the frequency responses of measuring microphones are small in most cases. For calibration purposes, however, they must be taken into account. Particularly in precision measurements with uncertainties of less than 0,1 dB, the amplitude characteristics of the microphones are not negligible. At present, the influence of the phase characteristics is almost unknown, but its effect on transient responses is obvious. To improve the situation, a broad-band phase and amplitude calibration of microphones is needed. It seems worthwhile to investigate reciprocity calibration of the pressure sensitivity of microphones with maximum-length sequences.

Besides the judgement of the frequency curve under free-field or ear simulator conditions, characteristics described by the impulse response are expected to be useful for the further development of headphones, ear simulators and HATS. In particular, measurements with test subjects compared with those with HATS seem to be good methods for improving of HATS, ear simulators and headphones. There is still a need for an ear simulator type for circumaural closed headphones and also for telephone sets. So far, the devices used now allow much less reproducibility than can be obtained with measurements of open type headphones.

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References

[1] IEC 645, Audiometers, Part 3, Method for the specification of acoustical test signals of short duration for audiometric and neuro-otological purposes. Draft Nov. 1991.

[2] MacWilliams, F. S., Sloane, N. J. A., Pseudo-Random Sequences and Arrays. Proc. IEEE **84** [1976], p. 1715.

[3] Alrutz, H., Über die Anwendung von Pseudorauschfolgen zur Messung an linearen Übertragungssystemen. Dissertation Universität Göttingen, Germany 1983.

[4] Borish, J., Angell, J., An efficient algorithm for measuring the impulse response using pseudorandom noise. J. Audio. Eng. Soc. **31** [1983], 478.

[5] Schmitz, A., Vorländer, M., Messung von Außenohrstoßantworten mit Maximalfolgen-Hadamard-Transformation und deren Anwendung bei Inversionsversuchen. Acustica **71** [1990], 257.

[6] Xiang, N., Mobiles universelles Meßsystem für die binaurale raumakustische Modelltechnik. Dissertation Ruhr-Universität Bochum, Germany 1991.

[7] IEC 318, An IEC artificial ear of wide-band type, for the calibration of earphones used in audiometry. 1970.

[8] IEC 711, Occluded ear simulator for the measurements of earphones coupled to the ear by ear inserts. 1981.

[9] Vorländer, M., Freifeld- und Diffusfeld-Übertragungsmaße von natürlichen Köpfen und von Kunstköpfen. Acustica **74** [1991], 192.

[10] IEC TR 959, Provisional head and torso simulator for acoustic measurements on air conduction hearing aids. 1990.

[11] ANSI S.3.36, Specification for a manikin for simulated in situ airborne acoustic measurements. 1986.

[12] ISO 389, Acoustics: Standard reference zero for the calibration of pure tone air conduction audiometers. 1988.

[13] Leckschat, D., Heyder, U., Verbesserung von Lautsprecher-Impulsantworten durch digitale Signalaufbereitung. Acustica **70** [1990], 55.

[14] Goertz, A., Leckschat, D., Entzerrung von Lautsprechern mit einem Signalprozessorsystem in Echtzeit. Fortschritte der Akustik – DAGA 90, DPG-GmbH, Bad Honnef, 1990, p. 995.

[15] IEC 268-7, Sound system equipment: Headphones and earphones. Draft May 1991.

[16] Richter, U., Übertragungsmaße von Kopfhörern aus Mini-Kasettengeräten. PTB-MA-18, Physikalisch-Technische Bundesanstalt, Braunschweig 1990.