Test Signal Generation and Accuracy of Turntable Control in a Dummy-Head Measurement System*

GYÖRGY WERSÉNYI, *AES Member*, **AND ANDRÁS ILLÉNYI,** *AES Member*

University of Technology and Economics, Békésy György Acoustical Research Laboratory, Budapest, Hungary

To measure the variations of the head-related transfer function (HRTF) caused by the acoustical environment near the head, a precisely controlled measurement setup with increased signal-to-noise ratio is needed. Based on the conclusions of an earlier investigation, a fully automatic dummy-head measurement system was installed in an anechoic room. An alternative way of generating a pseudorandom test signal is described as well as the rebuilding of the Brüel & Kjaer turntable for accurate settings of azimuth.

0 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,

$$\mathrm{HRTF} = \frac{P_1(\mathrm{j}\omega)}{P_2(\mathrm{j}\omega)} \tag{1}$$

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].

* Manuscript received 2000 October 4; revised 2002 October 16.

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° and +90°. 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,

$$\mathrm{HRTF}(\mathrm{j}\omega) = \frac{H_{\mathrm{outerears}}(\mathrm{j}\omega)}{H_{\mathrm{reference}}(\mathrm{j}\omega)}\,. \tag{2}$$

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 mea-surement 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,

$$\mathrm{Re}_{\mathrm{avg}}[i] = \sqrt{\frac{1}{N} \sum_{j=1}^{N} \left(\mathrm{Re}_{\mathrm{j}}^2[i] + \mathrm{Im}_{\mathrm{j}}^2[i] \right)} \tag{3}$$

$$\mathrm{Im}_{\mathrm{avg}}[i] = 0 \tag{4}$$

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,

$$\text{crest factor} = P_n = \frac{n_{\mathrm{peak}}(t)}{n_{\mathrm{rms}}(t)} . \tag{5}$$

Here

$$n_{\mathrm{rms}}(t) = \lim_{T \to \infty} \left\{ \frac{1}{T} \int_0^T x^2(t)\, \mathrm{d}t \right\}^{1/2} \tag{6}$$

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

$$t = \frac{1}{f} \tag{7}$$

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

$$N > \frac{T}{t} . \tag{8}$$

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 Re_{avg} calculated in Eq. (3). Let the phase Im_{avg} be a random variable with uniform distribution over the interval 0, … , 2π. The IFFT of this spectrum satisfies the first two requirements but does not satisfy the third,

$$n(t) = \mathrm{IFFT}\left\{\mathrm{Re}_{avg} + \mathrm{j\,Im}_{avg}\right\} . \tag{9}$$

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

$$P_n = \frac{n_{peak}(t)}{n_{rms}(t)} \tag{10}$$

$$Q_n = \sqrt{P_n} . \tag{11}$$

3) If there are sample whose absolute values exceed

$$Q_n' = n_{rms}(t)\, Q_n \tag{12}$$

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

$$q_n = 20 \log Q_n . \tag{13}$$

If there are spectral components whose absolute values exceed q_n dB, the corresponding component of the spectrum at the start, cut these amplitudes so that they will be exactly q_n dB greater than the corresponding component without affecting the phase information. Similarly, modify the component if the absolute value is more than q_n dB less than the according component of the starting spectrum.

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 characteristics 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

The authors would like to thank Péter Berényi and Richárd Tátrai for their ideas, useful comments, and help during the setup and programming.

6 REFERENCES

[1] J. Blauert, *Spatial Hearing* (MIT Press, Cambridge, MA, 1983).

[2] E. A. G. Shaw, "Transformation of Sound Pressure Level from the Free-Field to the Eardrum in the Horizontal Plane," *J. Acoust. Soc. Am.*, vol. 56, pp. 1848–1861 (1974 Dec.).

[3] S. Mehrgart and V. Mellert, "Transformation Characteristics of the External Human Ear," *J. Acoust. Soc. Am.*, vol. 61, pp. 1567–1576 (1977 June).

[4] D. Hammershøi and H. Møller, "Free-Field Sound Transmission to the External Ear; a Model and Some Measurement," in *Proc. Fortschritte der Akustik DAGA'91*, pt. A (Bochum, Germany, 1991), pp. 473–476.

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

H. Møller, "Head-Related Transfer Functions: Measurements on 40 Human Subjects," in *Proc. 6th Int. FASE Conf.* (Zürich, Switzerland, 1992), pp. 225–228.

[6] H. Møller, M. F. Sørensen, D. Hammershøi, and C. B. Jensen, "Head-Related Transfer Functions of Human Subjects," *J. Audio Eng. Soc.*, vol. 43, pp. 300–321 (1995 May).

[7] D. Hammershøi and H. Møller, "Sound Transmission to and within the Human Ear Canal," *J. Acoust. Soc. Am.*, vol. 100, pp. 408–427 (1996).

[8] W. M. Hartmann, "How We Localize Sound," *Phys. Today*, pp. 24–29 (1999 Nov.).

[9] H. Møller, M. F. Sørensen, C. B. Jensen, and D. Hammershøi, "Binaural Technique: Do We Need Individual Recordings?," *J. Audio Eng. Soc.*, vol. 44, pp. 451–469 (1996 June).

[10] J. C. Middlebrooks, "Narrow-Band Sound Localization Related to External Ear Acoustics," *J. Acoust. Soc. Am.*, vol. 92, pp. 2607–2624 (1992).

[11] J. Blauert, "Sound Localization in the Median Plane," *Acoustica*, vol. 22, pp. 205–213 (1969/1970).

[12] J. C. Middlebrooks and D. M. Green, "Sound Localization by Human Listeners," *Ann. Rev. Psychol.*, vol. 42, pp. 135–159 (1991).

[13] C. I. Cheng and G. H. Wakefield, "Introduction to Head-Related Transfer Functions (HRTFs): Representations of HRTFs in Time, Frequency, and Space," *J. Audio Eng. Soc.*, vol. 49, pp. 231–249 (2001 Apr.).

[14] M. D. Burkhard and R. M. Sachs, "Anthropometric Manikin for Acoustic Research," *J. Acoust. Soc. Am.*, vol. 58, pp. 214–222 (1975 July).

[15] H. Møller, "On the Quality of Artificial Head Recording Systems," in *Proc. Inter-Noise 97*, vol. II (Budapest, Hungary, 1997), pp. 1139–1142.

[16] P. Maijala, "Better Binaural Recordings Using the Real Human Head," in *Proc. Inter-Noise 97*, vol. II (Budapest, Hungary, 1997), pp. 1135–1138.

[17] H. Møller, D. Hammershøi, C. B. Jensen, and M. F. Sørensen, "Evaluation of Artificial Heads in Listening Tests," *J. Acoust. Soc. Am.*, vol. 47, pp. 83–100 (1999).

[18] D. R. Begault, E. M. Wenzel, and M. R. Anderson, "Direct Comparison of the Impact of Head Tracking, Reverberation, and Individualized Head-Related Transfer Functions on the Spatial Perception of a Virtual Speech Source," *J. Audio Eng. Soc.*, vol. 49, pp. 904–916 (2001 Oct.).

[19] P. Minnaar, S. K. Olesen, F. Christensen, and H. Møller, "Localization with Binaural Recordings from Artificial and Human Heads," *J. Audio Eng. Soc.*, vol. 49, pp. 323–336 (2001 May).

[20] M. Cohen and E. Wenzel, "The Design of Multidimensional Sound Interfaces," in *Virtual Environments and Advanced Interface Design*, W. Barfield and T. A. Furness III, Eds. (Oxford University Press, New York, 1995), pp. 291–346.

[21] D. J. Kistler and F. L. Wightman, "Principal Component Analysis of Head-Related Transfer Functions," *J. Acoust. Soc. Am.*, vol. 88, p. 98 (1990).

[22] F. L. Wightman and D. J. Kistler, "Headphone Simulation of Free-Field Listening I–II," *J. Acoust. Soc. Am.*, vol. 85, pp. 858–878 (1989).

[23] H. Møller, "Fundamentals of Binaural Technology," *Appl. Acoust.*, vol. 36, pp. 171–218 (1992).

[24] P. Schöne, "Zur Nutzung des Realisierungsspielraums in der kopfbezogenen Stereofonie," *Rundfunktech. Mitt.*, vol. 24, pp. 1–11 (1980).

[25] K. Genuit, "Bestimmung strukturgemittelter Außenohrübertragungsfunktionen," in *Proc. DAGA'84* (Darmstadt, Germany, 1984), pp. 667–670.

[26] A. Illényi and G. Wersényi, "Discrepancy in Binaural Tests and in Measurements of Sound Field Parameters," in *Proc. Békésy Centenary Conf. on Hearing and Related Sciences* (Budapest, Hungary, 1999), pp. 160–165.

[27] G. Wersényi and A. Illényi, "Evaluation of Differences in Dummy-Head HRTFs Caused by the Acoustical Environment Near to the Head," to be published by the *Acta Acoustica*.

[28] P. Berényi and A. Illényi, "What Does It Mean for an HRTF Not to Have the Minimal Phase Property?," in *Proc. Inter-Noise 96*, vol. VI (Liverpool, UK, 1996), pp. 2127–2130.

Table 1. Data sheet of measurement system.

Source	
Elevation	[−45°, +90°], 5° steps, 0.78% precision
Azimuth	[0°, 360°], 1° steps, 1.14% precision
Distance	1.8 m
Signal processing	
Sampling frequency	50 kHz
Conversion	16 bit maximum, 9.78 bit on average
FFT	4096 points
Channels	2 channels, simultaneous
Averaging	>768 frames
Loudspeaker	
Diameter	0.12 m
Transfer function	Constant within ±6°
Stimuli	Non-MLS 81.92-ms pseudorandom noise
SPL at origin	74.6 dB (max)
Validity of transfer function	Above 200 Hz
Overall SNR	>89 dB, independent of frequency
Deviation in transfer functions	<0.5 dB

[29] T. Behrens, H. Prante, and C. Maschke, "Untersuchungen zur Summenlokalisation in der Medianebene," in *Proc. DAGA'94* (1994), pp. 1157–1160.

[30] G. Wersényi, "Measurement System Upgrading for More Precise Measuring of the Head-Related Transfer Functions," in *Proc. Inter-Noise 2000* (Nice, France, 2000), pp. 1173–1176.

[31] W. G. Gardner, *3-D Audio Using Loudspeakers* (Kluwer, Boston, MA, 1998).

[32] V. Mellert, K. F. Siebrasse, and S. Mehrgardt, "Determination of the Transfer Function of the External Ear by an Impulse Response Measurement," *J. Acoust. Soc. Am.*, vol. 56, pp. 1913–1915 (1974).

[33] S. Müller and P. Massarani, "Transfer-Function Measurement with Sweeps," *J. Audio Eng. Soc.*, vol. 49, pp. 443–471 (2001 June).

[34] U. P. Svensson and J. L. Nielsen, "Errors in MLS Measurements Caused by Time Variance in Acoustic Systems," *J. Audio Eng. Soc.*, vol. 47, pp. 907–927 (1999 Nov.).

[35] J. C. Middlebrooks, "Virtual Localization Improved by Scaling Nonindividualized External-Ear Transfer Function in Frequency," *J. Acoust. Soc. Am.*, vol. 106, pp. 1493–1510 (1999).

[36] J. V. Hundeboll, K. A. Larsen, H. Møller, and D. Hammershøi, "Transfer Characteristics of Headphones," in *Proc. FASE'92* (1992), pp. 161–164.

[37] J. Borish and J. B. Angell, "An Efficient Algorithm for Measuring the Impulse REsponse Using Pseudorandom Noise," *J. Audio Eng. Soc.*, vol. 31, pp. 478–488 (1983 July/Aug.).

[38] K. Genuit and N. Xiang, "Measurements of Artificial Head Transfer Functions for Auralization and Virtual Audio Environment," in *Proc. ICA'95* (Trondheim, Norway, 1995), pp. 469–472.

[39] D. D. Rife and J. Vanderkooy, "Transfer-Function Measurement with Maximum-Length Sequences," *J. Audio Eng. Soc.*, vol. 37, pp. 419–444 (1989 June).

[40] S. Carlile and D. Pralong, "The Location-Dependent Nature of Perceptually Salient Features of the Human Head-Related Transfer Functions," *J. Acoust. Soc. Am.*, vol. 97, pp. 3445–3459 (1994).

[41] B. Zhou, D. M. Green, and J. C. Middlebrooks, "Characterization of External Ear Impulse Responses Using Golay Codes," *J. Acoust. Soc. Am.*, vol. 92, pp. 1169–1171 (1992).

[42] S. Foster, "Impulse Response Measurements Using Golay Codes," in *Proc. IEEE Conf. on Acoustics, Speech, and Signal Process.*, vol. 2 (1986), pp. 929–932.

[43] K. Hartung, "Modellalgorithmen zum Richtungshören, basierend auf den Ergebnissen psychoakustischer und neurophysiologischer Experimente mit virtuellen Schallquellen," Ph.D., dissertation, Ruhr-Universität, Bochum, Germany (1997). (Shaker Verlag, Aachen, Germany, 1999).

[44] E. Terhardt and W. Aures, "Wahrnehmbarkeit der periodischen Wiederholung von Rauschsignalen," in *Proc. DAGA'84* (Darmstadt, Germany, 1984), pp. 769–772.

[45] J. C. Makous and J. C. Middlebrooks, "Two-Dimensional Sound Localization by Human Listeners," *J. Acoust. Soc. Am.*, vol. 87, pp. 2188–2200 (1990 May).

[46] S. E. Boehnke and D. P. Phillips, "Azimuthal Tuning of Human Perceptual Channels for Sound Location," *J. Acoust. Soc. Am.*, vol. 106, pp. 1948–1956 (1999 Oct.).

[47] D. S. Brungart and W. M. Rabinowitz, "Auditory Localization of Nearby Sources: Head-Related Transfer Functions," *J. Acoust. Soc. Am.*, vol. 106, pp. 1465–1479 (1999).

[48] G. Wersényi, "HRTFs in Human Localization: Measurement, Spectral Evaluation and Practical Use in Virtual Audio Environments," Ph.D. dissertation, Brandenburg Technical University, Cottbus, Germany (2002).

THE AUTHORS

G. Wersényi

A. Illényi

György Wersényi was born in Györ, Hungary, in 1975. He received a M.Sc. degree in electrical engineering from the Technical University of Budapest in 1998. Since then he has been a full-time Ph.D. student there in the Békésy György Acoustic Research Laboratory of the Department for Telecommunication and Telematics and after the academic year 2000–2001 at the Brandenburg Technical University in Cottbus, Germany, he received a Ph.D. degree in 2002.

He is also a lecturer in audio engineering at the Széchenyi István University.

Dr. Wersényi is a member of the Hungarian Acoustical

Society (OPAKFI) and the AES. His areas of interest include spatial hearing, acoustic measurements, psychological acoustics, auditory modeling, acoustical information decoding, and transmission.

●

András Illényi was born in Budapest, Hungary, in 1932. He graduated from the Faculty of Physics and Mathematics of Loránd Eötvös University of Natural Sciences in Budapest in 1956. At present he is head of the George Békésy Acoustic Research Laboratory, Department of Telecommunication and Telematics, at the Technical University of Budapest.

His activities are related to many fields of the acoustical sciences and technical applications. His most important fields of interest are directional and binaural hearing, noise and vibration control, and practical applications. He has more than 80 publications in the fields of physical acoustics, measuring techniques, sound quality, HRTFs and binaural hearing, holophony, and room acoustics. He is the owner of elaborated licences in more than 50 countries; three in the United States.

Dr. Illényi is a member of the Acoustic Complex Committee of the Hungarian Academy of Sciences and was head of the committee from 1995–1998. He is a member of the following societies: Acoustical Society of America, Audio Engineering Society, Deutsche Arbeitsgemeinschaft für Akustik, Akustischer Verein Österreich, and a board director of I-INCE. He is president of the National Noise Committee in the Bureau of Standardisation in Hungary and of the OPAKFI (Scientific Society for Optics, Acoustics, Motion Pictures and Theatre Technology). He holds the “Excellent Inventor” gold medal (1972, 1973, 1975), the George Békésy medal (1983), the “Pro Silentio” medal (1990), and the Kempelen Farkas prize (1994).