Vom Helmholtz-Integral bis zur Helmholtz-Medaille

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Abstract

This paper is a summary of the presentation that the author gave – in German, with the above title - at the occasion of his honourful awarding with the Helmholtz medal. In this presentation, the basics and applications were summarized of array technology in the field of room acoustics and sound reproduction, which is probably the most important contribution of the TU Delft acoustics group to the development of modern technology in the rather traditional world of acoustics. It started with the entry of A.J.(Guus) Berkhout as the head of the Delft acoustics laboratory in 1976, succeeding prof. C.W. Kosten after a 4 year interim period. Prof. Berkhout came from the oil industry, where since long it is common use to explore the earth bottom with arrays of sound sources and receivers, yielding 3D images of the earth interior giving insight in its spatial structure and, hence, the location of fossile fuels. Berkhout'promoted the – by then rather revolutionary - application of array technology in the domain of 'audible acoustics' in order to obtain insight in the spatial structure of sound fields. The author had the privilege and pleasure to participate in these developments from the beginning, and to contribute to the promotion of this approach across the borders – also, and not in the last place, in Germany.

Microphone arrays in room acoustics measurements

Traditionally, in room acoustics research impulse responses were measured at individual receiver positions, being 'representative' for a certain area of the space considered. Fig.1 shows an example of such a response, measured with modern equipment in a small lecture hall at TU Delft. Even with a higher education and much experience in room acoustics, it is difficult to draw relevant conclusions from this image.

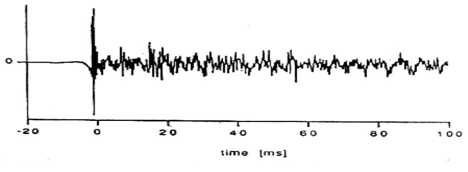

Fig.1. Impulse response measured in a small lecture hall.

As an experiment, in the early 1990s similar impulse responses were measured in the same hall (rectangular, dimensions ab.9m (l), 6m (w), 3m (h), furniture removed), where a loudspeaker was positioned at the center ab.2 meters from the front wall. A microphone was moved (by hand!) over the width of the hall along a line ab.4 meters in front of the loudspeaker with 5 cm interspacing – i.e., along a *linear array* of microphone positions.

The measured impulse responses are shown in Fig. 2.

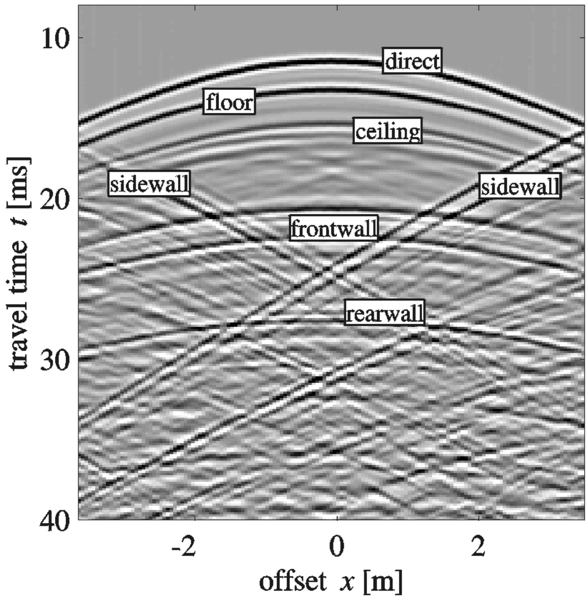

Fig.2. Multi-trace impulse response data set. Vertical axis: travel time. Horizontal axis: lateral microphone position *re* the center of the hall. Same hall as considered in fig.1.

The responses are plotted with vertical time axis, as a function of the position of the microphone, *offset* meaning the lateral distance to the hall center. This way, it is seen that the individual responses together form a picture of the sound *waves* propagating in the hall, which can easily be identified.

Due to the success of these introductory experiments, similar measurements have been done in theatres and concert halls, where the microphone was moved over a rail by means of an automated device [1].

An advantage of microphone array measurements is that they can be extrapolated to other, virtual array positions by signal processing. For this purpose, the data set is usually decomposed, by spatial Fourier Transformation, into plane waves, which can be easily extrapolated by a simple phase shift. After extrapolation to the desired virtual array position, inverse spatial FT yields a space-time representation similar to that shown in fig. 2.

This extrapolation is accurate for plane wave components incident parallel to the array, and increasingly less precise for components with increasingly oblique angle of incidence.

A solution for this problem is to perform an array measurement not only over the width of the hall, but also in the length direction, thus forming a *cross array* of microphone positions. Plane wave components incident unacceptably oblique on one cross array branch have an acceptable incidence angle to the other one, and *vice versa*, such that all components can be extrapolated with sufficient accuracy. An example is given in fig.3 where, from such a cross array measurement, impulse responses have been reconstructed for a square audience region of the Amsterdam Concertgebouw. From the resulting data, the Lateral Energy Fraction LEF – a parameter predicting the spatial impression of music – has been calculated for all positions in the area considered. (Note: it is seen that, in this fully symmetric geometry, the LEF results are asymmetric. Apparently, the atmospheric conditions in the hall have changed during the measurements.)

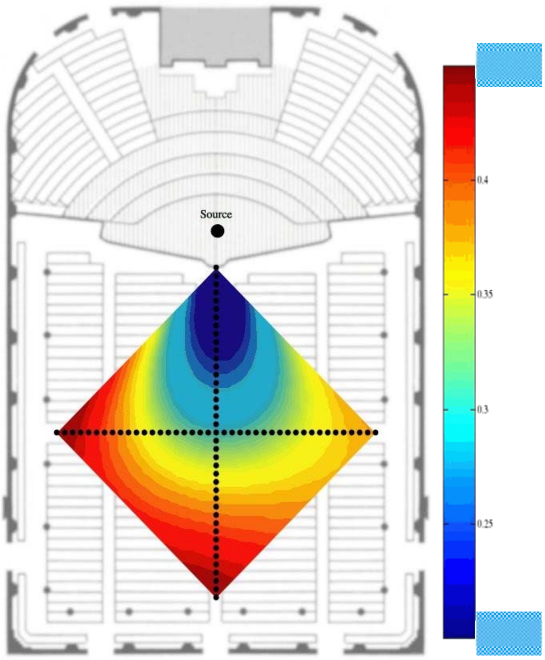

Fig.3. Lateral Energy Fraction values in a square audience region of the Amsterdam Concertgebouw, calculated from impulse responses extrapolated from cross array measurements.

A hall-wide linear array measurement takes several hours. As an alternative, *circular array* measurements can be applied, where a hypercardioid microphone mounted on a rod is rotated on a turntable in about 12 minutes. Analysis and extrapolation of the resulting data set can be done by processing based on *circular harmonics* theory [2]. This way, measurements can be done for several source positions (e.g., along an array) in the usual time available. Fig.4 shows the measurement setup configured in the Amsterdam Concertgebouw.

As a logical follow-up of the linear and circular array, a *spherical array* has been developed, Here, a microphone is moved along a so-called Lebedev distribution of positions on a spherical surface, enabling the discrimination of wave field components in three dimensions.

Microphone arrays in sound recording

Linear microphone arrays can also be applied for music recording. Then, instead of one microphone moving along an array of positions, a real microphone array has to be applied.

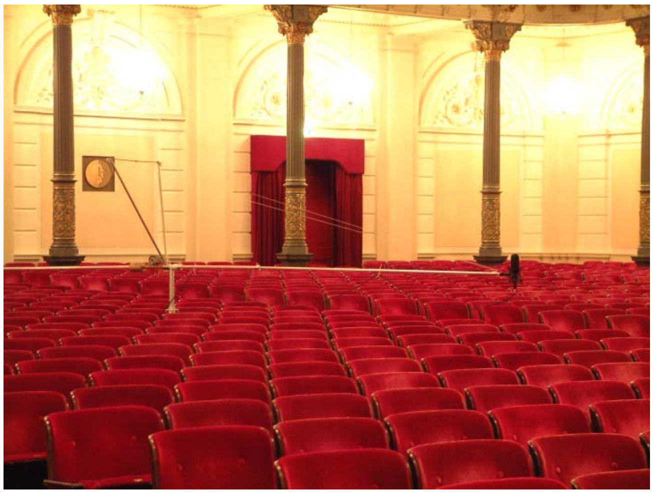

Fig.4. Circular array measurement setup in the Amsterdam Concertgebouw.

Such an array has the advantage that, with appropriate signal processing, *beamforming* can be applied, i.e., the recording can be focussed into a certain direction to select certain voices or instruments. The longer the array in comparison with the wave length considered, the narrower the beam and the sharper the focussing. When the distance between the microphones is larger than the (apparent) wavelength considered, unwanted spatial *aliasing* (backfolding) takes place. This means that, for an aliasing-free broadband recording, many microphones are required.

An economic solution is the application of a *logarithmic* microphone array, as schematically depicted in fig.5 [3].

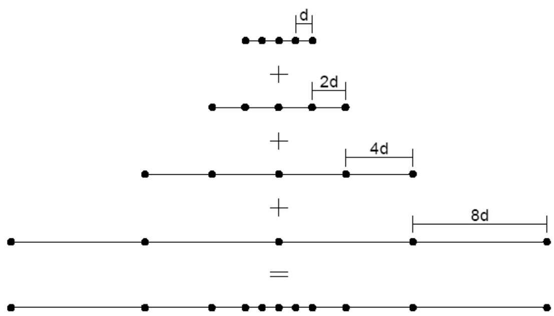

Fig.5. Schematic construction of a logarithmic array.

For a highest frequency band determined by the recording requirements, an array is designed with sufficient length L for appropriate beamforming and a microphone distance d avoiding spatial aliasing. For a frequency band a factor a lower (in fig.5: a =2), an array is designed with length aL and microphone distance ad. This is repeated until the lowest frequency band is reached. Combination of all these subarrays yields an array with an equidistant central part and outer parts where the distance between successive microphones increases with a factor *a*. This way, an array is constructed having focussing properties that are independent of frequency.

Fig.6 shows a recording session with a logarithmic microphone array performed in the Audimax of TU Ilmenau during a Sommerwerkstatt in the framework of the author’s guest professorship. The individual instruments were selected by focussing for later reproduction in Wave Field Synthesis format.

Fig.6. Ensemble „On the Fritz“ recorded with a logarithmic microphone array in the Audimax of TU Ilmenau.

Recording experiments have also been carried out with *circular* microphone arrays [4] (fig.7), also in cooperation with TU Ilmenau.

Fig.7. Orchestra recording with a circular microphone array in the ‘Frits Philips‘ music hall in Eindhoven, The Netherlands.

Also here, the theory of circular harmonics forms the basis of focussing and further post processing.

Loudspeaker arrays in sound reproduction: Wave Field Synthesis

Wave Field Synthesis (WFS) is a sound reproduction format aiming at giving all listeners a natural acoustical perception in time and space including a correct localization of the sources. For this purpose, arrays of closely spaced loudspeakers create a copy in time and space of the sound field generated by the sources. Fig.8 illustrates the difference with the traditional two-channel stereo format. Here, the spatial reproduction of the sound field and, hence, the source localization is only correct in a so-called ‘sweet spot‘. At other positions, the spatial perception is determined by the directivity characteristics of the loudspeakers. In WFS, the array loudspeakers correctly reproduce the sound field of the source in the full listeners area.

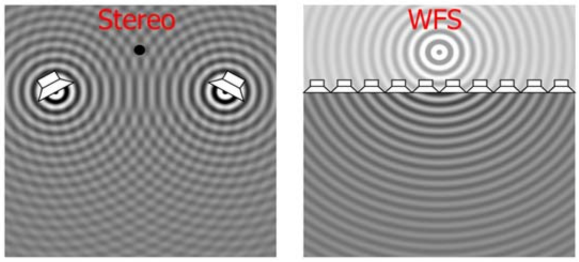

Fig.8. Illlustration oft the basic difference between two-channel stereo and Wave Field Synthesis.

Wave Field Synthesis finds its basics in Huygens‘s principle [5] which formulates that a light (and also a sound) source emits wave fronts which can be interpreted as continuous distributions of secondary sources with the strength of the local value of the primary source field, together creating the next wave front. This is illustrated in fig.9.

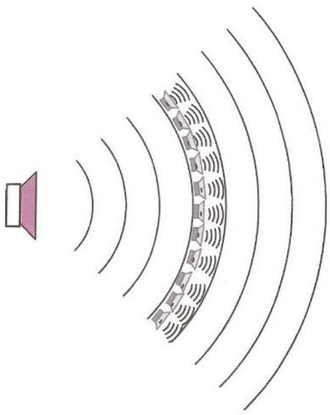

Fig.9. Illustration of Huygens‘s principle.

In order to make this principle applicable for practical audio purposes, we replace the poorly defined wave fronts by a fixed surface and consider the geometry sketched in fig.10. Sound sources are present outside a closed surface *S* with a local normal vector **n** at each position $\mathbf{r}_S$. Surface S includes a source-free volume with receiver positions **r**. The sources outside S generate a pressure P and a normal particle velocity component V_n on S, both considered in the space-frequency domain.

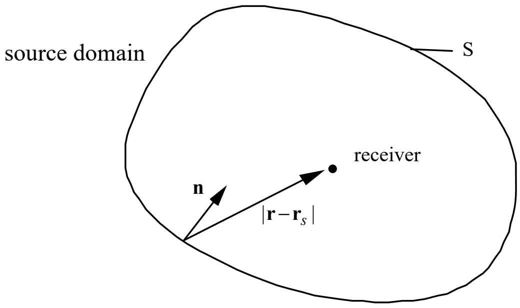

Fig.10. Geometry for adaptation of Huygens‘s principle to practical applicability.

After substantial mathematical manipulation in which the theorems of Gauss and Green are applied, the Kirchhoff-Helmholtz representation theorem is found, reading:

$$P(\mathbf{r},\omega)=\frac{1}{4\pi}\oint\!\!\oint_S\left[P(\mathbf{r}_S,\omega)\frac{1+jk|\mathbf{r}-\mathbf{r}_S|}{|\mathbf{r}-\mathbf{r}_S|}\cos\phi\frac{e^{-jk|\mathbf{r}-\mathbf{r}_S|}}{|\mathbf{r}-\mathbf{r}_S|}+j\omega\rho V_n(\mathbf{r}_S,\omega)\frac{e^{-jk|\mathbf{r}-\mathbf{r}_S|}}{|\mathbf{r}-\mathbf{r}_S|}\right]dS \qquad (1)$$

It says that, when the sound pressure P *and* the normal component of the particle velocity V_n are known at each position on surface S, integration of these quantities over S after weighting with certain functions yields the sound pressure at any receiver point within S.

Closer observation shows that the weighting function of V_n represents the field at **r** of a unity *monopole* source at $\mathbf{r}_S$, and the weighting function of P the field at **r** of a unity *dipole* source at $\mathbf{r}_S$. This means that surface S can be considered as covered with a continuous distribution of virtual monopoles and dipoles with the strength of the local field values of the primary sources, together ‘creating‘ the pressure field at receiver positions **r** inside S.

Eq.(1) provides basic possibilities to calculate sound pressure fields in a source-free area, but it has practical disadvantages. First, knowledge of the primary sound field on a 3D closed surface is required, in terms of pressure as well as normal velocity. A step in a more practical direction could be that only one of the field quantities has to be known on, say, a planar surface. Therefore, we modify the surface S into a (basically infinite) plane, completed to a closed surface by a hemisphere that is infinitely far away, or at least such far that it delivers no contribution to the sound field in the relevant listeners area. Using smart choices of Green functions, the so-called Rayleigh integrals can be derived for this situation:

$$P_A=\frac{j\omega\rho}{2\pi}\int_S V_n\frac{e^{-jk\Delta r}}{\Delta r}dS \qquad \text{Rayleigh I} \qquad (2)$$

$$P_A=\frac{jk}{2\pi}\int_S P\frac{e^{-jk\Delta r}}{\Delta r}\cos\varphi\, dS \qquad \text{Rayleigh II} \qquad (3)$$

Eq.(2) shows the Rayleigh I integral stating that, when the normal component of the particle velocity generated by sources at one side of a plane S is known at this plane, the sound pressure at any point A in the halfspace at the other, source-free side can be calculated. According to the Rayleigh II integral, Eq. (3), this is also the case when at plane S the sound pressure generated by the primary sources is known. Here, plane S can be considered as covered with a distribution of virtual monopole sources for Rayleigh I and with a virtual dipole source distribution for Rayleigh II. In both cases, the virtual sources ‘create‘ the pressure field in the source-free halfspace. This is illustrated in fig.11 as a modification of the Huygens‘s principle scheme in fig.10.

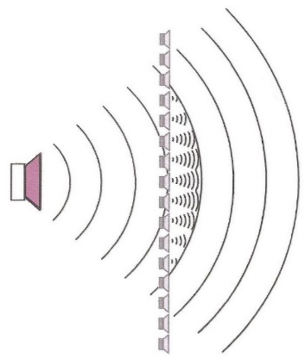

Fig.11. Illustration of the Rayleigh integrals configuration.

Now we come to practical applications. If we cover a plane surface with real loudspeakers having monopole or dipole characteristics and drive these with signals equal to, respectively, the local normal velocity or pressure values generated by the primary sources, we create a perfect copy, in time and space, of the original sound field – which is the optimal way of sound amplification.

If, as suggested by Berkhout, we leave out the primary sources and just drive the loudspeakers with the signals that should have been generated by the sources if they had been present, we simulate (reproduce) the sound field of the now virtual sources, to be perceived by any listener in the audience space with correct source localization. *This is the principle of Wave Field Synthesis.*

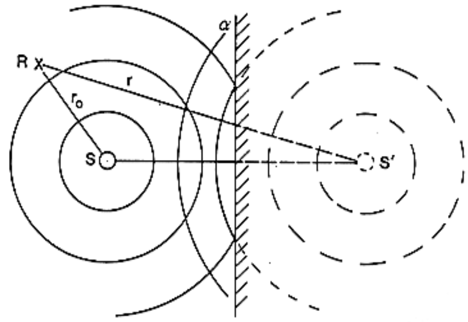

Fig.12. A room reflection can be interpreted as the direct sound of a mirror image source.

Since a room reflection can be considered as the direct sound of an image source mirrored in the reflecting boudary (see fig.12), also these reflections (i.e., the acoustics of the hall in which the reproduced music was performed) can be simulated by WFS. Fig.13 shows a schematic setup where listeners experience such recording: a frontal loudspeaker array reproduces the direct signals from stage, surrounding arrays simulate the room reflections, i.e., the acoustics..

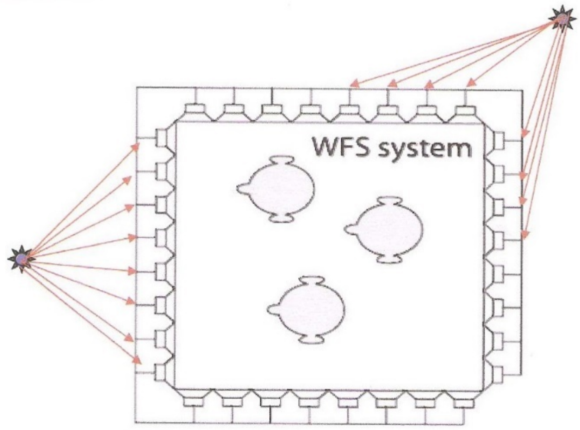

Fig.13. Schematic setup of a surround WFS system reproducing sources at stage and the acoustics of the hall.

For practical applicability, many adaptions had to be developed. In practice, instead of infinite planes covered with loudspeakers, linear or circular arrays have to do the job. Loudspeakers have finite dimensions such that spatial aliasing has to be coped with. Loudspeakers never have monpole or dipole characteristics for the full frequency range of interest. Room acoustics is generated by millions of mirror image sources which never can be individually reproduced, etc. etc. By further research, using the limited capacity of the human hearing mechanism in noticing simplifications and minor shortcomings, all these problems have been appropriately solved. See, e.g., [6], [7], [8]. Wave Field Synthesis appeared to work!

History of Wave Field Synthesis

In 1988, Berkhout published his first paper on Wave Field Synthesis [9] – then called Acoustic Holography which was soon changed, in order to avoid confusion with other techniques, into Wave Front Synthesis and shortly after into Wave Field Synthesis - and in 1989 a shorter version was presented by the author at the AES Convention in Hamburg [10]. At the same time, at the TU Delft laboratory a DSP-driven experimental setup was installed – see fig.14 - with which many demos were given in order to convince interested people of the promising possibilities of WFS. After a period of disbelief and hesitation, the interest increased, and in 1998 an invitation came from the Verband Deutscher Tonmeister VDT to transport the setup to Karlsruhe for demonstrations at the Tonmeistertagung in that city and to present papers on the topic [11], [12]. This was repeated in 2000 at the Tonmeistertagung in Hannover [13]. These presentations highly contributed to a broader appreciation of WFS.

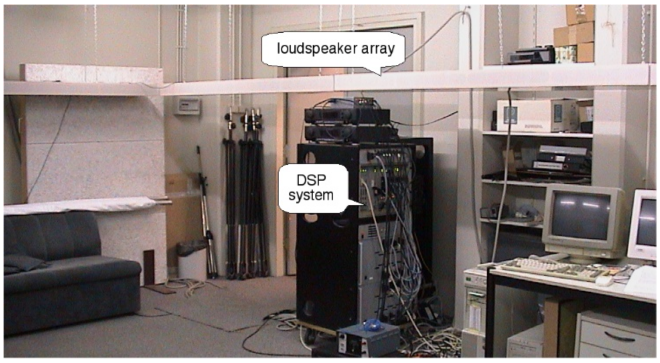

Fig.14. First WFS demo setup at the TU Delft acoustics laboratory. The white bars contain the array loudspeakers.

The latter visit resulted in the invitation to the author for an Edgard Varèse guest professorship at TU Berlin in 2001. In the scope of this professorship, the Delft WFS setup was installed in the large electronic music studio of the Fachgebiet Audiokommunikation, supervised at the time by Folkmar Hein. Sixteen demo sessions were presentated to a variety of interested target groups. Besides, the author gave lectures on room acoustics and array technology. The cooperation was the start of ample application of WFS in electronic music education and composition practice at TU Berlin, since 2004 under the supervision of Prof. Stefan Weinzierl.

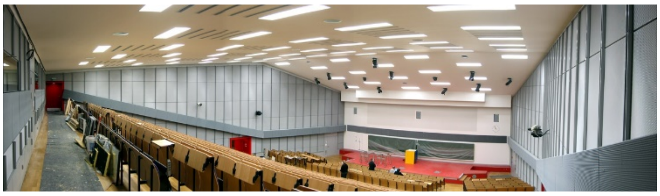

Fig.15. Grosser Hörsaal at TU Berlin with WFS system.

A former master student of the author, Marije Baalman, who is also a composer of electronic music, did her WFS-oriented PhD research at TU Berlin [14] and made a significant contribution of the development of an extensive WFS system (more than 2000 channels) in the Grosser Hörsaal of TU Berlin (fig 15).

Also in 2001, the EU-funded CARROUSO project had started, the name being an acronym phonetically reminding to the name of the great tenor singer Enrico Caruso, standing for 'Creating, Assessing and Rendering in Real time Of aUdio-visual environmentS in MPEG-4 cOntext', where the rendering should be performed in WFS format. Ten European industries, research organisations and universities participated, among which Fraunhofer IDMT in Ilmenau – coordinating the project under supervision of Thomas Sporer – TU Delft, IRT in Munich and IRCAM in Paris. Presentations and demonstrations were given at several conferences, such as the 22nd Tonmeistertagung in Hannover in 2002, where an entire paper session including a panel discussion moderated by the author was devoted to the CARROUSO project [15]. Fig.16 shows the CARROUSO demo setup at this Tagung, where the slide show was projected on a flat panel loudspeaker (MAP, Multi Actuator Panel) developed in the scope of the project [16].

Fig.16. CARROUSO demo at the 2002 Tonmeistertagung in Hannover. MAP loudspeakers also served as slide projection screens.

In this demo, the performance of MAPs was compared with that of traditional loudspeakers. The project ended in 2003. Most of the goals had been reached. Again, WFS had successfully been promoted.

Fig.17. WFS system in cinema,'Lindenlichtspiele' in Ilmenau.

Meanwhile, Fraunhofer IDMT installed the first commercially operating system in the local cinema 'Lindenlichtspiele' in Ilmenau, see fig.17.

In 2004, the author fulfilled a guest professorship at TU Ilmenau, where again he gave lectures and organized, in close cooperation with Fraunhofer IDMT and highly supported by TU staff member Bernhard Albrecht, workshops on array technology, also in the form of Sommerwerkstätte during the following years. At IDMT, WFS research and development strongly proceeded in this period, leading to the installation of powerful WFS research systems – see fig.18 - and a commercial offspring in the form of the IOSONO company.

When in 2009, a year before his retirement, the author wrote a monography on WFS for the Audio Engineering Society AES [17], he counted WFS research systems at 15 universities and research institutes in Austria, France, Germany (7!), The Netherlands, Spain, Switzerland and the United Kingdom. Besides, there were 20 WFS systems in use worldwide for public performances in theatres, cinemas and entertainment halls.

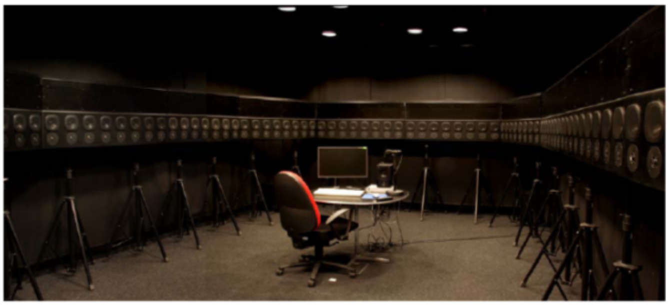

Fig.18. WFS system at the Fraunhofer IDMT laboratory in Ilmenau.

After his retirement in 2010, the author worked three years as a part time guest profesoor at RWTH Aachen, where he participated in the first phase of the DFG-funded SEACEN project - the acronym standing for Simulation and Evaluation of ACoustical Environments – by supervising the PhD research of Rob Opdam on an array-based method for the simulation of sound fields in rooms [18].

Outlook

Since the retirement of the author in 2010 and that of his colleague M.M. (Rinus) Boone two years later, TU Delft ended the research on 'audible acoustics', including the application of array technology in this field. Fortunately, the author forsaw this in time. He is very happy that, with full support of the TU Delft authorities, he got the opportunity to transfer the TU Delft knowledge and experience on array technology to Germany through the guest professorships described above. With pleasure he observes that, at the universities and institutes where he had the privilege to work, research on array technology fruitfully proceeds.

Due to the CARROUSO project and other international cooperation networks, array technology R&D activities have also been initiated in other parts of Europe and the world. It is doubtful if, due to its requirement of a large amount of loudspeakers, WFS in its pure form will conquer the world, but the many attempts made nowadays to simplify the concept with preservation of the high-quality spatial perception make a good chance. As long as the researchers understand the physical basics of array technology and the working of the human hearing mechanism, and use this knowledge in a smart combination, solutions must be within reach!

To conclude, the author wants to mention the activities of Prof. Sascha Spors, who organized the DAGA 2019 where the author received his Helmholtz medal. As a talented PhD student from Erlangen, Sascha Spors participated in the CARROUSO project. After finishing his PhD dissertation, he started, in cooperation with TU Berlin, a strong WFS research group at Deutsche Telekom. Meanwhile, he updated the 'old' WFS theory to a 2.0 version. Nowadays, he is a full professor at Rostock University, where he continues his fruitful research on array technology and, no doubt, inspires his staff and students to follow him on that way. Yes, the application of array technology in room acoustics and sound reproduction has good perspectives!

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