Chapter 12
Manfred Schroeder and Acoustical Impedance

Roland Kruse and Volker Mellert

Abstract Manfred Schroeder was a man with many interests. Between the years 1966 and 1967, he invented devices for measuring the surface impedance of materials as well as the vocal tract impedance. At the time, he would only consider normal incidence; however, he already anticipated that the behavior of materials at grazing incidence would have relevance to room acoustics, a field of research he is well known for.

To achieve this aim, an experimental set-up to determine the characteristic impedance of absorbers at grazing incidence has been developed. The sample is placed on the bottom of a rectangular impedance tube and the horizontal wave number above the sample is calculated from the pressure transfer function between two microphones. From this wave number, the absorber properties can be deduced. While the method works reasonably well for highly absorbing samples, the non-ideal sound field in the tube—as confirmed by finite element simulations—limits its usefulness in case of less absorbent materials. Improvements of the method are suggested.

12.1 Introduction

Manfred Schroeder was a man with many interests. Between the years 1966 and 1967, he invented devices for measuring the surface impedance of materials as well as the vocal tract impedance. At the time, he would only consider normal incidence; however, he already anticipated that the behavior of materials at grazing incidence would have relevance to room acoustics, a field of research he is well known for.

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12.2 System for Determining Acoustic Reflection Coefficients

The first sign of Manfred Schroeder's research on measuring impedances is his patent number 3,346,067 [10], granted in October 1967 but filed already in March 1966, which deals with a "system for determining acoustic reflection coefficients" which equals, as we have plane wave propagation, a device for impedance determination. Before having a detailed look at this invention, let us have a look at how reflection/absorption coefficients of materials (at normal incidence) were measured at that time: using the Kundt's tube. This meant that for each frequency of interest, one had to find—by moving the microphone along the tube—the minimum and maximum of the sound pressure. In addition, if one was interested in the complex reflection coefficient, one had to locate the position of the first minimum with respect to the sample's surface. Obviously, that was a time-consuming task, and for materials with high absorption, like porous absorbers at high frequencies, it was difficult to locate the extrema of the sound pressure.

Manfred Schroeder sought to overcome this problem by using broadband signals and two microphones, relying on the definition of the reflection coefficient: "The reflection coefficient, an important parameter in determining the acoustic characteristics of a material, is defined as the ratio of the amplitude of a reflected pressure wave to the amplitude of an incident pressure wave, both amplitudes being measured at the face of the material" [10]. In other words, we try to separate incident and reflected wave and to predict the sound pressure at the sample's surface from measurements done at some distance to it.

Figure 12.1 shows a sketch of the system. It shows what would nowadays be called a two-microphone impedance tube, the only difference being that the sound source is not located opposite the sample but at the side of the tube.

From the measured pressure amplitudes at the two locations, first the amplitudes of the incident and reflected wave were derived with Eq. (12.1), then the (complex) reflection coefficient: Eq. (12.2). Analog electronics (delays, multipliers, etc.) were used to perform these operations, as this was the established solution for such a

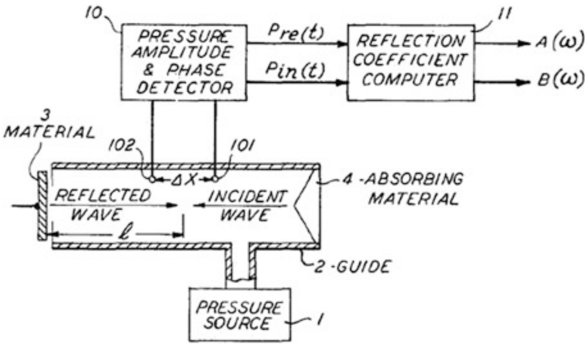

Fig. 12.1 Sketch of the impedance measurement system invented by M.Schroeder in 1966 [10]

problem at that time (the term "reflection coefficient computer" is rather misleading from today's point of view on digital signal processing). Broadband processing was achieved by using a bandpass filter bank and having multiple "computers" evaluating Eq. (12.2) simultaneously. Hence, one could obtain the frequency-dependent, complex reflection coefficient almost instantaneously.

$$P_{\text{in}}(\omega) = \frac{1}{e^{i\omega\Delta t} - e^{-i\omega\Delta t}}\left[P_1(\omega)e^{i\omega\Delta t/2} - P_2(\omega)e^{-i\omega\Delta t/2}\right]$$
$$P_{\text{re}}(\omega) = \frac{1}{e^{i\omega\Delta t} - e^{-i\omega\Delta t}}\left[P_2(\omega)e^{i\omega\Delta t/2} - P_1(\omega)e^{-i\omega\Delta t/2}\right] \tag{12.1}$$

$$\Re(R) = \frac{\overline{P_{\text{re}}(t)P_{\text{in}}(\omega_j, t)}}{\overline{P_{\text{in}}^2(\omega_j, t)}}\,\Im(R) = \frac{\overline{P_{\text{re}}(t)\left(P_{\text{in}}(\omega_j, t)e^{i\pi/2}\right)}}{\overline{P_{\text{in}}^2(\omega_j, t)}} \tag{12.2}$$

Thinking about the work of Manfred Schroeder one wonders how this invention fits into his line of work, as it seems to be neither related to room acoustics (surprisingly, the patent does not refer to the use of reflection/absorption coefficients in this area) nor to speech processing or perception. A closer look reveals that it is most likely a spin-off from his research on speech (re-) production.

12.3 Determination of the Geometry of the Human Vocal Tract

In 1967, Manfred Schroeder submitted a manuscript [11] to JASA, published the next year, in which he describes a procedure involving, first, the determination of the frequency-dependent impedance of the vocal tract and, second, the derivation of the vocal tract geometry using an appropriate model. Reading the motivation behind this study, it becomes clear why Schroeder was interested in such a topic: "Knowledge of vocal-tract configuration (tongue and jaw position, lip rounding, etc.) is basic to a better understanding of the physical and physiological processes involved in the human speech process—one of man's important means of communication. Present lack of knowledge in this area is epitomized by the variable quality and limited vocabulary exhibited by present speaking machines. . ..." No, Manfred Schroeder was not interested in a mere measurement technique but had a much more visionary goal in mind, something that was also part of the zeitgeist with science fiction movies featuring talking robots and "computers".

There're several factors making this publication particularly interesting: on the one hand, the setup used to measure the impedance is very similar to the one in the patent, with the material sample replaced by the subject's mouth, and on the other hand, because of the extensive use of digital processing: the two microphone signals are digitized right away, the impedance is calculated, poles and zeros of the impedance curves (representing resonances in the vocal tract) are located and from them the geometry (cross-sectional area versus distance) is estimated by numerical inversion of Webster's horn equation, a process involving an iterative procedure which digital computers are well suited for. We may, consequently, guess that the system filed for patent is a scaled-down version of the system presented in this chapter, making use of established components.

This in not the first occasion demonstrating Schroeder's interest in digital computers, he also used them for measuring room-acoustical parameters [12] and for simulating the effect of such parameters, e.g., the frequency-dependent reverberation time, on sound perceived in a "virtual" room [13]. And all this despite the limited processing power available in the Sixties and the enormous price of digital computers. Obviously, someone believed in the future of these machines.

Now where do we stand today? The two-microphone impedance tube (with digital processing) is a standardized method (ISO 10534-2) for measuring the normal-incidence reflection coefficient. In addition, in situ methods are used to determine the impedance at arbitrary angles of incidence [14, 15]. What one may miss is a laboratory method for determining the impedance at grazing incidence.

Manfred Schroeder gives us a hint on what this can be useful for: in his paper on measurements in the Philharmonic Hall [16], he and his colleagues made an interesting observation, the seat effect: "This lack of low frequencies in the first overhead reflection revealed another low-frequency deficiency that ad hitherto gone unnoticed: A progressive attenuation of low frequencies in the direct sound as it grazes across the rows of seats. (This 'seat effect' must exist in many other halls, but it is usually masked by the presence of low-frequency components in the early overhead reflections)."

The acoustical properties of materials or structures can vary with the angle of incidence (non-local reaction), and for a correct simulation of the room acoustics data for other than normal incidence is needed.

Therefore, we shall now outline how such data can be obtained using a standing-wave tube.

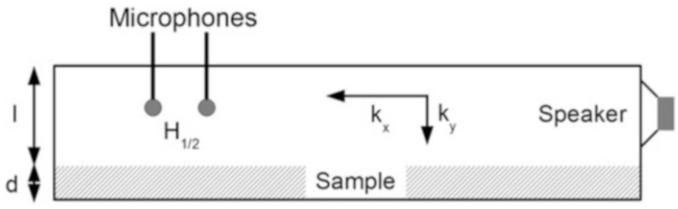

Fig. 12.2 A standing-wave tube with grazing sound incidence

12.4 Standing-Wave Tube at Grazing Incidence

Figure 12.2 shows an overview of the proposed setup. It is similar to a two-microphone impedance tube, but the sample is now placed at the bottom of the tube, which has a square cross-section of $10 \times 20\ \text{cm}^2$ and a length of 91 cm. It is made from 20 mm thick aluminum to guarantee a sufficient bending stiffness. The upper cover is height adjustable and allows the channel height to be increase by up to 8 cm. The loudspeaker at the right radiates broadband noise, which is recorded by two microphones having the same height above the sample's surface. The quantity to be measured is the transfer function $H_{1/2}$ between the microphones. To allow for differences between the sensitivities and phase responses of the microphones, the switch-microphones calibration technique [17] has been used.

This technique is an advancement of a technique described by [18] who used a Kundt's tube-like approach with a movable microphone with an otherwise identical setup.

The general effect of the (absorbing) sample at the bottom of the channel is that waves traveling above it are attenuated and do not remain plane waves. This can be exploited to find the acoustical properties of the material under test.

12.4.1 Theoretical Background

The sound field inside the tube is assumed to be two-dimensional as the depth of the channel is only 10 cm. The field can then be described by a horizontal wave number k_x and a vertical wave number k_y. In the first step, k_x is deduced from the measured transfer function $H_{1/2}$ and the distances x_1, x_2 of the microphones with respect to the hard wall at the end of the tube. We shall assume that this wall is perfectly reflecting. Equation (12.3) can then be used, keeping in mind that the microphones are located at the same height y above the sample; it can be solved numerically for k_x.

$$H_{1/2} = \frac{p_1}{p_2} = \frac{e^{-ik_x x_1} + e^{ik_x x_1}}{e^{-ik_x x_2} + e^{ik_x x_2}} = \frac{\cosh(ik_x x_1)}{\cosh(ik_x x_2)} \tag{12.3}$$

The wave number k_x is linked to the wave number k_a and characteristic impedance Z_a of the material and the geometry of the problem. The theory of Scott [19] as outlined by Eq. (12.4) has been employed, which basically states that the vertical wave components are to be continuous at the sample's surface. It is important to note that this theory (and the similar one by Mechel [20]) assumes an infinitely long channel; especially, it does not take any additional effects, i.e. the boundary condition of a source into account.

$$\sqrt{k^2 - k_x^2} \tan\left(\sqrt{k^2 - k_x^2} l\right) = \frac{1}{Z_a' k_a'} \sqrt{k_a^2 - k_x^2} \tan\left(\sqrt{k_a^2 - k_x^2} d\right) \tag{12.4}$$

Equation (12.4), too, can be solved numerically for the absorber properties. Because this would require deriving two complex quantities from a single one, the solution may be non-unique. In addition, just like in the case of in situ impedance deduction, pretests indicate that it is prone to small measurement errors [21]. Therefore, we will assume that the absorber's impedance and wave number can be described by, e.g., the Delany–Bazley model [22], so that only the single parameter "flow resistivity" has to be deduced. Other absorber models would serve as well.

12.4.2 Results and Discussion

To verify the functionality of setup and data processing, measurements were first done with an empty tube, so that the expected value for k_x is $(1 + 0i)$. Figure 12.3 shows the actual results. At low frequencies, the wave number deviates from the expected value, likely due to the (relative) proximity of the loudspeaker and hence the initially non-plane wave field. Unexpectedly, the wave number is also incorrect around 800 Hz (and at other, higher frequency ranges). In addition, within these frequency ranges, the coherency between the microphone signals (not shown) was reduced due to the sound pressure at one of the microphones being very low. Assuming plane waves, the lowest frequency a pressure minimum is expected to occur at a microphone 10 cm in front of a hard wall is 850 Hz but the minimum actually occurred at the microphone 2 cm in front of the wall.

A finite element simulation of the sound propagation inside a rigid channel with known geometry and position of the sound source was done to better understand how the sound field looks like. In Fig. 12.4, the pressure distribution inside the empty tube is shown for four frequencies. At 500 and 700 Hz, one sees the non-plane field close to the source, whereas at the left side (where the microphones are located), the wave fronts are mostly flat. On the other hand, at 850 Hz, there's a

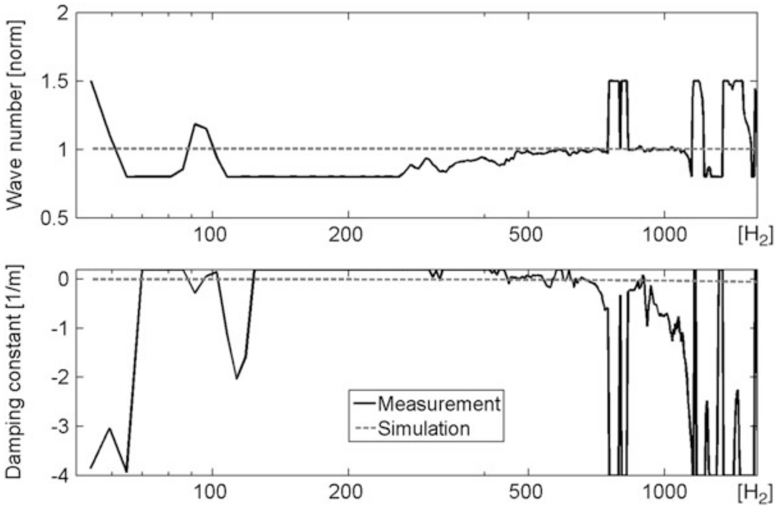

Fig. 12.3 Measured horizontal wave number in an empty tube with the microphones 2 and 10 cm in front of the hard backing. Expected value is $k_x = (1 + 0i)$

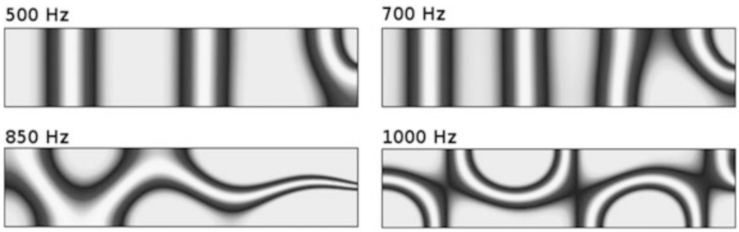

Fig. 12.4 Sound pressure distribution inside a channel of $20 \times 91\ \text{cm}^2$ with rigid walls and a 8 cm loudspeaker at the *right side*. *White indicates* low, black high pressure

pressure minimum in the top left corner leading to the low coherency of the microphone signals. Though this is unexpected from a room acoustics point of view, where one would always expect pressure maxima in the corners, this is simply the result of the source not being small compared to the channel dimensions. By use of a probe microphone, the course of the pressure minimum in this corner has been verified to agree well with the FE simulation.

Further simulations show that this unwanted "source effect" can be reduced by placing a second loudspeaker at the right wall, thus making the excitation more symmetric, and by reducing the height of the channel above the sample.

With the given setup, another option was to place the microphones further away from the reflecting wall, as then the propagation loss due to the absorbing material would reduce the interference between incident and reflected wave, hence increasing the pressure in the mimima. One must be aware that this places the microphones closer to the source and might reduce the accuracy of the method at low frequencies.

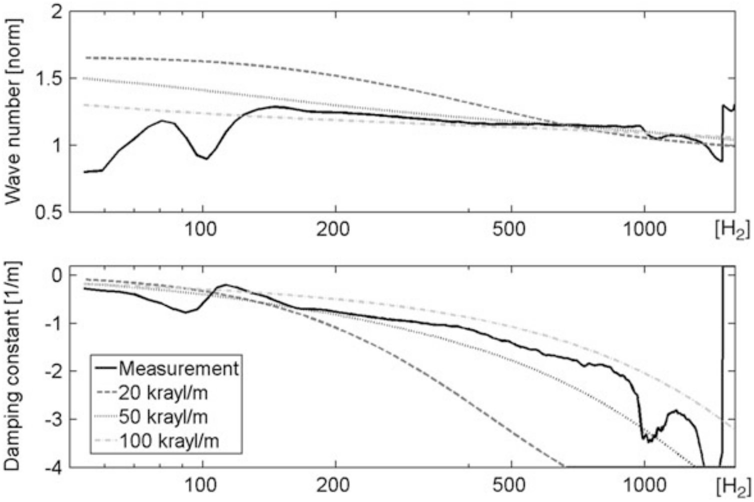

Fig. 12.5 Measured wave number above 9 cm polyurethane foam in comparison with the wave number predicted for a Delany–Bazley type absorber with three different flow resistivities

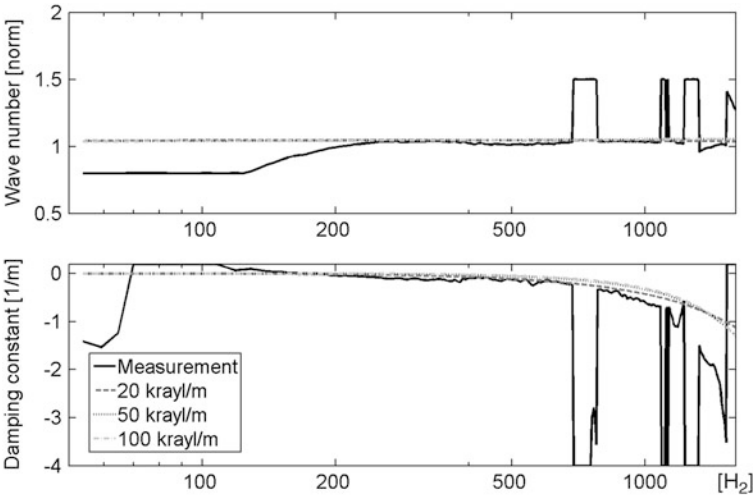

Fig. 12.6 Measured wave number above 1 cm polyurethane foam in comparison with the wave number predicted for a Delany–Bazley type absorber with three different flow resistivities

Figures 12.5 and 12.6 show the measured wave number in comparison with the wave number expected for a Delany–Bazley type absorber of same thickness. The microphones were now located 22 resp. 31.5 cm in front of the left wall. For the 9 cm thick foam, the (average) course of the wave number above 150 Hz agrees well with the one expected for a porous absorber with a flow resistivity between

50 and 100 krayl/m. Due to the high propagation loss, the sharp peaks as observed with an empty tube have disappeared.

The results for the otherwise comparable 1 cm thick foam do not prove satisfactory. On the one hand, due to the reduced absorption, the undesired extrema have reappeared. On the other hand, even the predicted courses of the wave number do hardly vary for flow resistivities of 20–100 krayl/m because a 1 cm thick layer simply does not produce a significant propagation loss within the investigated frequency range.

12.4.3 Summary and Conclusion

In 1966, Manfred Schroeder demonstrated his interest in measurement of the acoustic impedance through a publication in JASA, dealing with the impedance of the human vocal tract, and through a patent describing a two-microphone impedance tube. The knowledge of the (surface) impedance of materials and structures is required for room-acoustical simulations. The standard procedure ISO 10534-2 measures the impedance for normal incidence, though observations by Schroeder in relation to the seat effect indicate that a measurement at grazing incidence can be desirable.

In our study, we have advanced an existing but hardly known and inconvenient technique to gain a procedure which is similar to the ISO standard. The tested material is placed on the bottom of a rectangular impedance tube and the wave number in the air space above it is deduced from the sound pressure transfer function between two microphones. By use of a model for the sound propagation in such a situation, the absorber characteristics can be deduced. Measurements show that the new technique does still have its limitations: the sound field deviates from what the model predicts because the existence of the source of finite size is not taken into account. As a result, at low frequencies the incident field is not reasonably flat; additionally, pressure minima leading to poor coherency between the microphone signals occur at other locations than expected. Finite element simulations verify these findings. By careful choice of the microphone positions, credible results could be achieved for absorbent material samples. But for samples with low absorption, like a 1 cm thick open cell foam, the method is not sensitive enough to distinguish between materials with largely different flow resistivities, at least with the current measurement setup and geometry.

All in all, though improvements are still required, the new method has demonstrated its ability to determine the impedance at grazing incidence in a well-defined laboratory setting allowing for the comparison between different materials.

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Biography

Dr. **Roland Kruse** received his Ph.D. degree in 2009 from the University of Oldenburg, under the supervision of Prof. Volker Mellert, for his work on the in situ measurement of the acoustical ground impedance. From 2009 to 2010 he continued his research at the Acoustics Group, The Open University, Milton Keynes studying the influence of the environmental conditions on source localization outdoors. After returning to the Acoustics Group at Oldenburg University, he became a lecturer at the Technical University of Braunschweig in 2011. His current research is aimed at finite element technologies and material parameter identification for biological and other composite materials.

Prof. **Volker Mellert** was head of the Acoustics Group at the Institute for Physics of Oldenburg University and retired in 2009. He received his Ph.D. in physics in Göttingen 1972 and became a Professor for Applied Physics in Oldenburg in 1976. He was President of the German Association for Acoustics (DEGA) and of the European Association for Acoustics (EAA). He served as board member of the International Commission of Acoustics and as Editor-in-Chief for ACUSTICA. He received the German Medal of Merit for his contributions in acoustics and the Helmholtz-Medal of the DEGA. He is Fellow of the Acoustical Society of America.