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THE RANDOM INCIDENCE SENSITIVITY OF MEASUREMENT MICROPHONES

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Abstract

The random-incidence sensitivity of a microphone is defined as the ratio of the output voltage to the sound pressure that would exist at the position of the acoustic center of the microphone in the absence of the microphone in a sound field with incident plane waves coming from all directions. Although the measurement process seems to be straightforward, some practical and fundamental problems arise: i) reflections from the mounting rig contaminate the measured frequency response, and whereas some of these reflections can be removed using a time-selective technique, others coincide with the direct impulse response and consequently cannot be removed; ii) the accuracy of the estimate is heavily dependent on the rotational symmetry of the microphone and on the angular resolution. The directivity factor of a number of laboratory standard microphones has been determined experimentally. Considerations on the influence of the angular resolution are presented. Although the procedure has so far only been applied to laboratory standard microphones, it is not restricted to such microphones and may be applied to other types of measurement microphones.

INTRODUCTION

The sensitivity of a microphone depends on the sound field where it is immersed. In a calibration, the microphone can be subjected to a uniform sound pressure over the diaphragm, a plane wave in a free field, or a diffuse field. The first two cases have been extensively studied. This has led to the development of a number of standards [1, 2, 3] and further investigations continue [4, 5]. The diffuse-field sensitivity has always been considered equivalent to the random-incidence sensitivity, which is defined in terms of the response of the microphone to a number of plane waves that impinge successively onto the microphone from all possible directions [6, 7].

Directivity factor

If we assume that the microphone is rotationally symmetrical, the directivity factor can be determined using the following expression [8]

$$Q(f) = 2\left[\int_{\Omega} f^2(\theta) d\Omega\right], \tag{1}$$

where $f(\theta)$ is the fractional response at the angle θ. This is defined as the ratio of the frequency response at the angle θ to the frequency response at normal incidence, i.e., $f(\theta) = H(\theta)/H(0)$. This expression can be used when an analytical form of the fractional response is available. However, this is not the case when the frequency response of a measurement microphone is determined experimentally. In such a case, the integral in Eq. (1) must be replaced by a discrete series

$$Q(f) = \frac{2H^2(f, \theta_0)}{\sum_{n=1}^{\pi/\Delta\theta} H^2(f, \theta_n) \sin\theta_n \Delta\theta}. \tag{2}$$

Needless to say such a discretization is made under the assumption that it has no significant effect on the final accuracy of the estimated sensitivity. A standard concerned with the determination of the diffuse field calibration of sound level meters recommends that each segment should not be larger than 3% of the total measurement area [6]. Finally, the directivity index is the directivity factor in logarithmic form [9]. The random-incidence sensitivity can be derived from the directivity index and the frontal-incidence free-field correction. The random-incidence correction can be determined from the directivity index and the frontal-incidence free-field correction.

The experimental determination of the random incidence sensitivity is far from a simple and straightforward process. Measuring the frequency response at all incidence angles is not trivial. The measurements must be carried out in an anechoic environment, but even in the best anechoic room the accuracy of the measurement will be degraded by reflections from the walls and from the measurement rig. Hence a technique that can remove such effects is needed. A time-selective technique that has been developed for the reciprocity calibration of microphones in a free field can be applied with advantage [10]. Another time-selective technique, time delay spectrometry [11] can also be used advantageously.

Measurement set-up

The measurement set-up is composed of the measurement rig and the measurement instrumentation. The former is mounted inside the anechoic room and the latter can be placed in a control room. The measurement rig has to comply with two criteria: i) it has to be as reflection-free as possible, and ii) the rod where the microphones are mounted has to be long enough to be considered semi-infinite and it has to have the same cross-section as the microphone. An scheme of the measurement rig is shown in Fig. 1. The rig is in the middle of a large anechoic room with a free volume of nearly $1000\,\mathrm{m}^3$. The hollow rod where the microphones are mounted is about $80\,\mathrm{cm}$ long, and it has the same diameter of the microphone under test. The cross which hangs from the ceiling is about 1 m away from the microphone rod, and the piano wires are less than 1 mm in diameter. The microphone and microphone rod are aligned

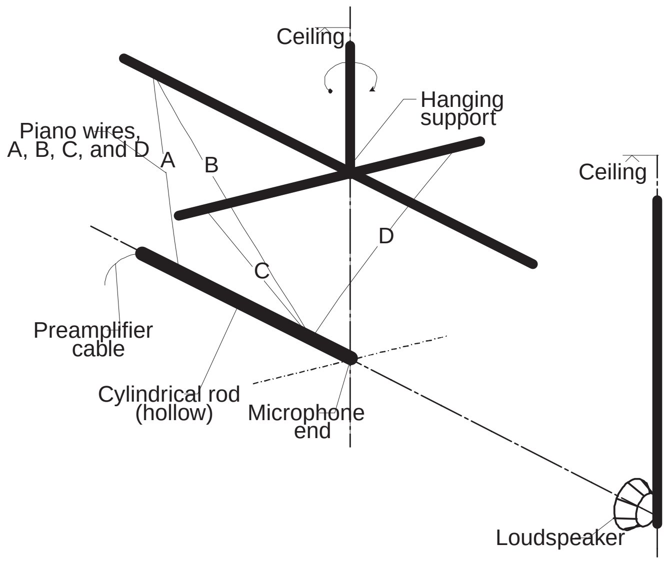

Figure 1: Schematics of the mounting rig inside the anechoic room. The drawing is not scaled

in such a way that the rig rotates around the diaphragm of the microphone. The loudspeaker is about 1.7 m away from the microphone. A *quarter-inch* monitor microphone is placed in front of the loudspeaker. The loudspeaker is a modified tweeter that has a flat frequency response up to 40 kHz. Determining the frequency response of the test microphone at a given angle involves measurement of the output voltage of the monitor and the test microphone.

Cleaning technique

Even if the measurement rig has been carefully designed in order to minimize any reflections that could contaminate the measured frequency response, some elements of the rig will reflect sound back to the microphones. In most cases these unwanted reflections can be removed using a time-selective technique. This kind of technique has been tried before in free-field reciprocity calibrations [10].

The procedure applied in this case is very similar: (a) The frequency response has been measured as the transfer function between the signals of the two microphones; (b) the missing lower portion of the frequency response is filled with theoretical data; (c) the high frequency response is tapered smoothly to zero by a low-pass filter; (d) an inverse Fourier transform is calculated; (e) a time-selective window is applied to the resulting impulse response in order to clean it from reflections; and (f) a Fourier transform is applied to the cleaned impulse response. Figure 2 shows the impulse response determined from measurements at several angles of incidence.

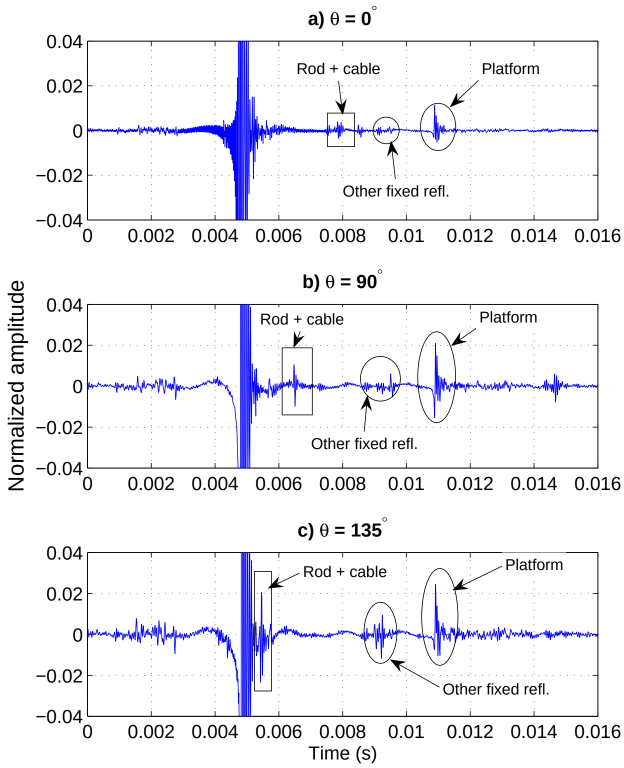

Figure 2: Impulse response of the original rig at different angles of incidence.

THE INFLUENCE OF THE ANGULAR RESOLUTION

The determination of the directivity factor by using Eq. (1) would yield the exact value of the factor; however, such an expression cannot be used in the case of the microphones. Instead, the directivity factor must be approximated by Eq. (2). Therefore, the influence of the angular step on the final estimate of the quantity sensitivity should be analyzed.

A suitable procedure would consist in determining the directivity factor using different angular resolutions and compare whether the estimate converges to a unique value as the resolution gets finer. Although it is intuitively evident that the finer the resolution the better the result, there is no actual indication of how accurate the estimate determined from the finest resolution will be. On the other hand, Eq. (1) can be implemented on a case where the analytical solution of the scattering problem exists, such as the case of a sphere. Once this estimate is obtained it can be compared with the outcome of the implementation of Eq. (2) for the same case. The analytical and the discrete estimates are analyzed below.

The case of a sphere

If harmonic variation with the $e^{-i\omega t}$ sign convention is assumed, the sound pressure scattered by a solid sphere of radius a, centered at the origin of a spherical coordinate system, can be calculated from the expression [12]

$$p_s(r,\theta) = -A \sum_m \left\{ (2m+1) i^{m+1} e^{-i\delta_m)} \sin \delta_m P_m(\cos\theta) h_m(kr) \right\}, \tag{3}$$

where A is the amplitude of a incident plane wave coming from the direction θ, $h_m(x)$ is the spherical Hankel function of first kind and order m, P_m is the Legendre function of order m, k is the wave number, r is the distance to the observation point which in this case is the radius of the sphere, and the angle δ_m is defined as

$$\delta_m = \arctan \left\{ \frac{(m+1) j_{m+a}(ka) - m j_{m-1}(ka)}{m n_{m-1}(ka) - (m+1) n_{m+1}(ka)} \right\}, \tag{4}$$

where j_m and n_m are spherical Bessel and Neumann functions of order m. The total pressure on the surface of the sphere is determined by adding the incident plane wave

$$p_t(r,\theta) = A \left\{ \sum_{m=0}^{\infty} (2m+1) i^m P_m(\cos\theta) \left[j_m(ka) - i e^{-i\delta_m} \sin \delta_m h_m(ka) \right] \right\}. \tag{5}$$

Equation (5) can be then integrated as in Eq. (1) or it can be estimated at discrete values of θ and then an estimate of the directivity factor can be estimated using Eq. (2).

Figure 3a shows the difference between an estimate of the directivity index obtained by numerical integration and estimates obtained from the discrete summation of the function at discrete values of θ, 20°, 10°, and 5°.

The case of a microphone

Unlike the case of the sphere, there is no analytical expression for the diffraction of an impinging sound wave on a microphone. Therefore, it is only possible to apply Eq. (2) to the ratio of the frequency response measured at discrete values of the angle θ to the frequency response at $\theta = 0°$. Thus, it is not possible to establish an exact reference of the directivity factor that can be compared with the discrete estimate. However, the differences between discrete estimates obtained using increasing angular steps can be compared with the case of the sphere. Figure 3b shows the difference between discrete estimates obtained using 5°, 10°, 20°and 30°.

DISCUSSION

As can be seen in Figure 2, in some cases, the reflections cannot be separated from the impulse response. This occurs at incidence angles larger than 120°. This is because at such an angle perturbations from any element behind the microphone will be a part of the direct wave. This

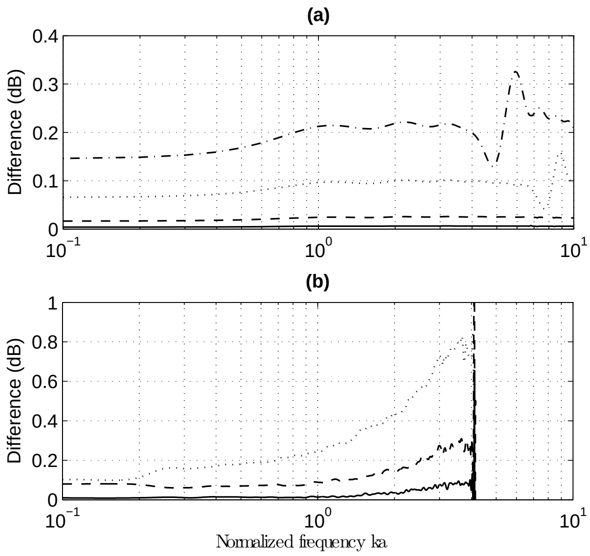

Figure 3: The influence of the angular resolution on the directivity index. (a) Difference between the analytical solution and the discrete estimate for a sphere using angle steps of 5°(solid line), 10°(dashed line), 20°(dotted line), and 30°(dash-dotted line); (b) difference between an estimate of the directivity index of microphones determined with steps of 5°and estimates obtained using steps of 10°(full line), 20°(dashed line), and 30°(dotted line)

is a fundamental problem of the method: unless the cable and the hanging wires are removed, there is no easy way to get rid of their reflections. In spite of this, the time-selective technique proves to be useful in eliminating other strong reflections from elements of the set-up.

The difference between the analytical and the discrete estimates of the directivity index of the sphere is nearly constant in the whole frequency range for all angular steps, whereas the directivity index of the microphone increases with frequency, and the slope changes as the angular resolution coarsens. The reason is that microphones are much more directional than spheres. However, the differences between the discrete estimates obtained using 5°and 10°are of the same order of magnitude for the microphone and the sphere, about 0.01 dB at low frequencies; this is also the case for the difference between 10°and 30°, about 0.06 dB at low frequencies. Thus, the difference between the analytical estimate and the discrete estimate for the sphere gives an idea of the accuracy that can be reached using different angular resolutions. A resolution of 5°seems to be accurate enough, because the difference is less than 0.01 dB; using a resolution finer than 5°would improve the accuracy very little while increasing the measurement time because of the additional measurements needed.

The directivity index of LS1 and LS2 microphones has been determined experimentally. Four LS1 microphones were used in the investigation, and one of them was measured four times, whereas another was measured twice. This gives a total of eight measurements.

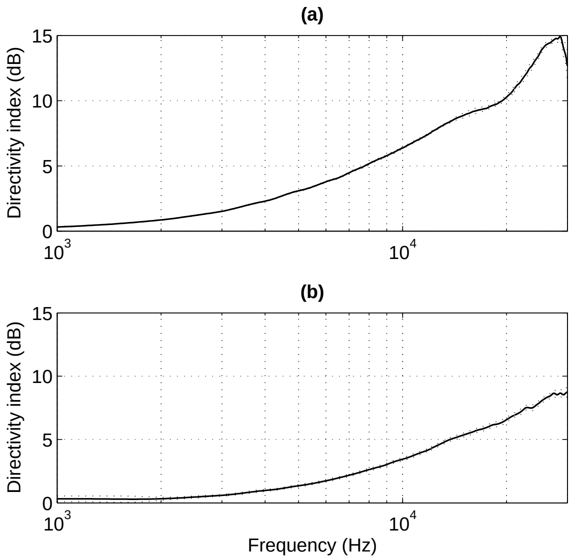

Figure 4: Experimental directivity index of: a) LS1 microphones, and b) LS2 microphones. Solid line: average; dotted lines: average plus and minus one standard deviation

These are shown in Fig. 4a. Six LS2 microphones were used in the investigation as well. Two of them were measured twice; this is a total of eight measurements. The results are shown in Fig. 4b. The reproducibility of the directivity index is better for LS1 microphones than for LS2 microphones. This is not unexpected because LS2 microphones have a lower sensitivity. The good behavior of LS1 microphones makes it possible to observe a significant deviation between 14 kHz and 18 kHz, reaching a maximum of about 16 kHz. Two phenomena occur in this frequency range: (a) A resonance of the back cavity of the LS1 microphones occurs at around 16 kHz. This resonance may be excited in different ways depending on how the oblique incidence modifies the movement of the diaphragm. Such a behavior cannot be detected in the LS2 case. A reason may be that the diaphragm of the LS2 is less compliant and more damped as well; (b) a non-axisymmetric mode in the cavity occurs at about 18 kHz and it may be excited by the oblique impinging wave; LS2 microphones have a cavity shorter than LS1 microphones, thus minimizing the influence of the non-axisymmetric mode. The differences between microphones may be caused by the differences in the resonance frequency of the microphones.

CONCLUSIONS

The directivity index of laboratory standard microphones has been determined experimentally. Reflections from the room and the measurement rig were removed from the frequency responses using an FFT-based, time-selective procedure.

The influence of the angular resolution has been studied by determining the directivity index using different resolutions. The case of the diffraction of the microphone was compared with the case of the diffraction of a sphere. The comparison showed that a resolution of 5°represents a good compromise between accuracy and measurement time.

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