Selected Topics in Electroacoustical Calibrations, Measurements and Standards

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Abstract: Modern instruments including multi-channel DSAs (dynamic signal analyzers) capable of FFT (fast Fourier transform) and other signal processing, as well as improved concepts for applying these instruments to particular electroacoustical measurement and standardization issues, are essential tools in modern electroacoustical metrology. The overall uncertainty of a measurement, and its suitability for its intended purpose, may involve components of uncertainty that are individually and collectively difficult to evaluate. Consequently, the capacity to perform the same measurement in different ways can be especially valuable. Different measurements exploiting different source signals, signal processing implementations, and apparatus can provide insight supporting the validity of measurement results and interpretations.

Examples selected from recent and ongoing work at National Institute of Standards and Technology (NIST) include calibration and characterization of microphones and microphone arrays in a free field, and the effects of imperfections in anechoic chambers, instabilities, and nonlinearities in electroacoustical systems. Recent international standardization work and comments involving different methods for examining nonlinearities in the electroacoustical performance of hearing aids should elicit further discussion of the relative advantages and disadvantages of these methods.

MICROPHONE CALIBRATION AND CHARACTERIZATION

International Electrotechnical Commission (IEC) standards published in the 61094 (formerly numbered 1094) series deal with the specification of critical mechanical and electroacoustical characteristics of LS (laboratory standard) and WS (working standard) microphones, and the primary methods for calibrating the pressure and free-field sensitivities of LS microphones by reciprocity techniques (1-4). Many metrology laboratories are refining the implementation of these methods. Newer microphone types are involved in numerous issues, such as determining the values of the frequency-dependent acoustic center positions for IEC Type LS2 microphones, especially at frequencies above 20 kHz. Methods for calibrating the free-field sensitivities of measurement microphones by comparison with a LS or WS microphone of known sensitivity are not yet the subject of international standardization activity, but are of great practical significance.

Modern electronic instruments such as the best available DSAs incorporating a variety of source signals and signal processing techniques are sufficiently accurate to be used in these comparison methods, especially for instances in which source transducer characteristics and the imperfections of anechoic chambers usually contribute significant, and sometimes dominant, components of the overall measurement uncertainties. DSA source signal characteristics and microphone output signal acquisition and processing techniques can be selected to exhibit different degrees of sensitivity to the effects of source transducer limitations and anechoic chamber imperfections. Such selection often leads to valuable insights regarding the influences of particular transducer limitations and chamber imperfections on calibration uncertainties. At frequencies from about 2 kHz to 40 kHz, a variety of source signal types, source transducers, and microphone output signal acquisition and processing techniques have been used to determine relative free-field sensitivity levels of a pair of IEC type WS2 working standard microphones by comparison calibration in the large, general purpose NIST anechoic chamber that agree within a few tenths of a decibel with the relative sensitivity levels of these microphones obtained from primary calibration by the reciprocity technique in the small NIST anechoic chamber.

DIRECTIONAL ARRAYS AND HEARING AID MEASUREMENTS

Directional arrays of microphones, including those incorporating conventional, adaptive, and hybrid adaptive beamforming techniques, offer promise of improved hearing aid performance in noisy and reverberant acoustical environments (5-8). Cosmetic acceptability to users is usually required for the commercial success of new hearing aids, so that the possible incorporation of directional arrays into eyeglass frames is of considerable interest.

The designer of an array frequently assumes that its microphone elements are point receivers in the free field, but the actual (in situ) performance of an array placed on a person will be influenced to varying degrees by diffraction,
scattering, and reflection of sound caused by structures including the array mounting apparatus and the human pinna, head, and torso. In principle, for a given placement position of the array on a person or, more conveniently, on a HATS (head and torso simulator, also called an anthropometric manikin), measured characteristics can be used to modify the design in attempts to improve the in situ performance. In practice, such attempts may prove difficult or even hopelessly impractical if a given placement is characterized by excessively large spatial gradients of sound pressure over the array that are also strongly frequency-dependent (and possibly also strongly dependent on the angle of incidence of sound from the source), resulting in large, strongly frequency-dependent differences in sound pressure between microphone elements that change significantly if the array changes its position slightly (as could happen, for example, when a person removes and replaces the hearing aid, or turns his or her head). Problems would also occur if differences between an individual user and the simulator prove to be significant.

A straight-line array of four microphone elements uniformly spaced at intervals of 2.54 cm (array length 7.6 cm) recently was provided by its designers to NIST for attachment to the earpiece of eyeglasses placed on a well-known HATS (KEMAR) in the large anechoic chamber, and subsequent measurement (among other characteristics) of transfer functions between microphone outputs. For frontal incidence of sound on this simulator, initial results demonstrate that some positions are much better than others: in the worst case, a given between-microphone transfer function exhibited strongly frequency-dependent variations of more than 25 dB in absolute value (relative to 0 dB, the value indicating essentially equal output levels from the two microphones) in the approximate frequency range 3 kHz to 6 kHz. In the best case, for each of three such transfer functions these variations in relative level were less than 9 dB at all frequencies up to 6.4 kHz, the upper limit of measurement, and exhibited more gradual dependence on frequency. Clearly, effects of array position (for this type of eyeglass frame placement, at least) on simulated in situ performance must be considered carefully in efforts to optimize array performance on hearing aid users. There also can be substantial differences between these measurements and the corresponding transfer functions measured with the array mounted on a different pair of eyeglasses on KEMAR.

**MEASUREMENT AND CHARACTERIZATION OF HEARING AID NONLINEARITIES**

Recent standardization work involving application of the coherence function to examine effects of nonlinearities in the electroacoustical performance characteristics of hearing aids has elicited some comments proposing an alternative method involving the Hilbert transform. There sometimes seem to be differences in the objectives (not always clearly stated or understood) toward which these methods are applied in different countries. More balanced discussions of the relative advantages and disadvantages of these (and perhaps yet other) methods would appear to be useful, especially involving the advocates of Hilbert transform methods.

**REFERENCES**