Loudspeaker Directivity Database Documentation

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Summary

This document describes the methodology and data format for our database of measured anechoic loudspeaker impulse responses. The measurements were made in the anechoic chamber of the 3D Audio and Applied Acoustics (3D3A) Laboratory at Princeton University.

1 Measurement Procedure

We conduct measurements in accordance with the prescriptions of the AES standard on loudspeaker polar radiation measurements [1], with one exception: instead of measuring impulse responses (IRs) over the entire sphere, we measure IRs only along horizontal and vertical orbits around the loudspeaker, as is commonly done by loudspeaker manufacturers [2]. These measurements are conducted in an anechoic chamber, which has dimensions of $3.6 \times 2.35 \times 2.55$ m ($l \times w \times h$) and anechoic wedges 8 inches deep, which corresponds to one quarter-wavelength at ~ 425 Hz. The measurement procedure has been described in detail in a previous publication [3].

Briefly, we place, in the anechoic chamber, the loudspeaker on a computer-controlled turntable (Outline ET250-3D), aligned such that the point of rotation coincides with the center of the high-frequency transducer, by default (the precise alignment may vary by loudspeaker). The loudspeaker is placed upright on the turntable for the horizontal orbit, and sideways for the vertical orbit. We then place the measurement microphone (B&K Type 4189-A-021) at a (nominal) distance of 1.6 m from the loudspeaker, and align the barrel of the microphone with the measurement axis.[1]

With the loudspeaker initially on-axis with the microphone, we generate and send to the loudspeaker an exponential sine sweep (ESS) [4] (more recently we have adopted the so-called "phase-controlled" ESS [5]) and record the resulting microphone signal. All measurements are conducted at a sampling rate of 96 kHz and the ESS signals are generated with a nominal frequency range of 20 Hz to 48 kHz and a duration of 5 seconds. The loudspeaker is then rotated in 5° increments and the above steps are repeated until the orbit is completed.

1.1 Impulse Response Extraction

We first deconvolve the recorded sweep by the input sweep, via the fast Fourier transform (FFT), to yield the measured IR. For each IR, we detect the onset via thresholding, with the threshold level set to the smallest multiple of 5% (of the peak amplitude) which exceeds the peak noise amplitude of the IR. Once the threshold-crossing (onset) is found, we compute and store the relative delay (in samples) between the onsets of the current and on-axis IRs, and then time-align the current IR to the on-axis IR. We then truncate the IR to N samples (typically $N = 16,384$ at a sampling rate of

[1] The measurement axis is defined as the line that passes through the measurement microphone and the point of rotation [1].

96 kHz) with the onset occurring at $N/4$ samples. These IRs, the threshold levels, and the delay values are provided in the database, as described in Section 2.

1.2 Normalization

The measurement microphone is calibrated (using a B&K Type 4231 calibrator) such that −11 dBFS corresponds to 94 dBSPL. Consequently, to express the loudspeaker frequency responses in dBSPL, we first compute the discrete Fourier transform (DFT) of the IR, where the DFT and its inverse are given by:

$$H[k] = \sum_{n=0}^{N-1} h[n] e^{\frac{-2\pi i n k}{N}} \quad \iff \quad h[n] = \frac{1}{N} \sum_{k=0}^{N-1} H[k] e^{\frac{2\pi i n k}{N}}. \tag{1}$$

We then rescale to dBSPL by applying the following normalization:

$$|H_{\text{SPL}}| = \frac{|H|}{10^{-5.25}}, \tag{2}$$

or, equivalently,

$$20 \log_{10} |H_{\text{SPL}}| = 20 \log_{10} |H| + 105. \tag{3}$$

2 Data Format

The measured data for each loudspeaker is stored in (up to) two MAT-files[2] with the naming conventions `SPKR_H_IR.mat` and `SPKR_V_IR.mat`, where `SPKR` is a short-hand for the loudspeaker (e.g., "SandersM11" for the Sanders Sound Systems Model 11), and `H` and `V` correspond to the horizontal and vertical orbits, respectively. Inside each file there are four variables:

1. the sampling rate (samples/second) (stored as `fs_H`, `fs_V`),
2. a $72 \times N$ matrix of IRs (`IR_H`, `IR_V`),
3. a 72×1 column-vector of delay values (samples) (`d_H`, `d_V`), and
4. a 72×1 column-vector of threshold levels (%) (`th_H`, `th_V`).

As in Section 1.1, N is the number of samples in each IR, and the 72 rows correspond to different measurement angles (0° through 355° in increments of 5°). Note that rows for which no measurements were made are filled with zeros. The delay values (in conjunction with the threshold levels used to obtain them – see Section 1.1) may be used to extract the original arrival times of the IRs, relative to the on-axis measurement.

[2] http://www.mathworks.com/help/matlab/import_export/load-parts-of-variables-from-mat-files.html

References

[1] AES56-2008: AES standard on acoustics - Sound source modeling - Loudspeaker polar radiation measurements, 2008 (reaffirmed 2014).

[2] Floyd E. Toole. *Sound Reproduction: Loudspeakers and Rooms*. Focal Press, 2008.

[3] J. G. Tylka, R. Sridhar, and E. Y. Choueiri. A Database of Loudspeaker Polar Radiation Measurements. In *Audio Engineering Society Convention 139*, October 2015. URL `http://www.aes.org/e-lib/browse.cfm?elib=17906`.

[4] A. Farina. Simultaneous Measurement of Impulse Response and Distortion with a Swept-Sine Technique. In *Audio Engineering Society Convention 108*, February 2000. URL `http://www.aes.org/e-lib/browse.cfm?elib=10211`.

[5] K. Vetter and S. di Rosario. ExpoChirpToolbox: a Pure Data implementation of ESS impulse response measurement. In *4th Pure Data Convention*, 2011.