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Data Article

Acoustic recordings data from an echoic environment and a toolkit for its analysis

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ABSTRACT

The primary data are the impulse responses that were recorded in an echoic environment, using a set of twelve loudspeakers and a microphone. They were used as a part of an acoustic calibration process of large environments, as presented by Kazakov and Nelken (DOI: 10.1016/j.jneumeth.2018.08.025; Kazakov and Nelken, 2018). The impulse responses can be also used to localize the microphone in 3D (multi-lateration). The required audio files and the MATLAB code allows a complete reproduction of the experiment.

Specifications table

Subject area	Biology
More specific subject area	Computational Neuroscience
Type of data	Figures, Sound files for impulse production, the recorded Golay sequences, MATLAB code, a MATLAB data structure with the impulse responses

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2352-3409/

How data was acquired	Sound production: The sounds (complementary Golay pair) were generated by a single multichannel sound card (RME M16AD) and fed through programmable attenuators (PA5, TDT). Sounds were presented through TDT MF1 speakers (Tucker-Davis Technologies). The speakers were driven by power amplifiers (SA1, TDT). Sound recording: The sound was recorded by a calibrated microphone (Brüel&Kjær model 4939) at 192 kHz. Automated sound sampling: Spatial sampling was automated using a custom-built robot, based on model RB-Rbo-33 by RobotShop (http://www.robotshop.com).
Data format	Raw, analyzed
Experimental factors	A trigger was produced and recorded in a parallel channel, marking the precise timing of the sound onset. Using this trigger the discrete Golay sounds were cropped out from the continuous sound recording
Experimental features	At each one of the 10 locations, each of the 12 speakers played the Golay complementary pair, with a silence buffer of 0.2 s between successive sound presentations. Upon completion, the microphone was horizontally shifted by 8.5 cm, followed by the next round of sound playing. During the post-processing, the recordings of the two Golay sequences were identified using the trigger that was stored in a separate channel. The sounds recorded in response to each of the two complementary sequences were cross correlated with their original sequence, and the two added up to produce the estimate of the impulse response.
Data source location	Edmond and Lily Safra Center for Brain Sciences, The Hebrew University of Jerusalem, Jerusalem, Israel. Location: 31°46'19.8"N 35°11'49.3"E
Data accessibility	The data is supplied within this article. See README.txt for further instruction regarding the code. Additional sound samples are available upon request (the full data is not uploaded due to large volumes)

Value of the data

- The complementary Golay pair (Ga16.wav, Gb16.wav) can be used in impulse response analysis, and additional Golay pairs of any length can be generated using the provided code (generate_golay.m).
- The recorded impulse responses illustrate the type of acoustic distortions that are expected to occur in lab conditions due to the geometry of the arena. The geometry of the arena in which that data was collected is described here and in Kazakov and Nelken [1].
- The impulse responses can be used to develop advanced sound onset identifiers, for precise distance measurements that are based on the time-of-arrival of a sound wave.
- In particular, these data allows 3D localization of the microphone by using distance measurements from multiple loudspeakers.

1. Data

This data article provides the required audio and code files to conduct an impulse response analysis, based on Golay complementary sequences, followed by an acoustic 3D localization of the microphone. The impulses were produced using a complementary pair of Golay sequences [2], supplied as sound files `Ga16.wav` and `Gb16.wav`. The code for generating the complementary pair is provided as `golay.m`. The impulse responses are provided as a single file `sound_samples.wav`, that contains a consecutive 120 recordings from 10 horizontally aligned recordings locations, 12 speakers at each location. Additionally, we supply code for the impulse response analysis (`parse_calib_file.m`) and acoustic 3D localization of the microphone (`localize_points_in_3D.m`).

2. Experimental design, materials, and methods

2.1. The arena

The sounds were recorded in a circular environment shown in Fig. 1, and thoroughly described in Kazakov and Nelken [1] (Section 2.1). The environment was assembled from 18 black rectangles of clear cast acrylic (Perspex) sheet (Fig. 2).

2.2. Impulse response measurements using Golay sequences

The impulse response analysis is presented in Kazakov and Nelken [1], Sections 2.2–2.5. The code that generates the Golay sequences is provided as `generate_golay.m`, and the resulting sequences of length 2^{16} are provided as `Ga16.wav` and `Gb16.wav`. The computation of the impulse from the complementary pairs is described by Fig. 3.

2.3. Analysis of the acoustic recordings

Once a sound file was recorded, the onset of the presentation of each sequence was located using a custom written script (`parse_calib_file.m`, attached to this article). The script locates the triggers that indicate sound onset and selects the segment of the recording that follows the trigger. Each pair of recordings is cross-correlated with the Golay sequences that were presented and summed to produce a single IR (as in Fig. 3). The two Golay sequences and the IR are stored in an instance of a `singleLoc`class, that represents the analyzed sound from a specific speaker and a recording location. For the `sound_sampling.wav` file, this script creates the `horizontal_line.mat` that holds an array of 120 `singleLoc` objects.

2.4. Distance measurements using impulse responses

An impulse response can be used to estimate the distance of the microphone from the speaker. To do so, we identified the onsets of the sound at both the speaker and the microphone locations: The

Fig. 1. The echoic environment. An image of the echoic arena. 12 speakers are spread in pairs around the arena wall.

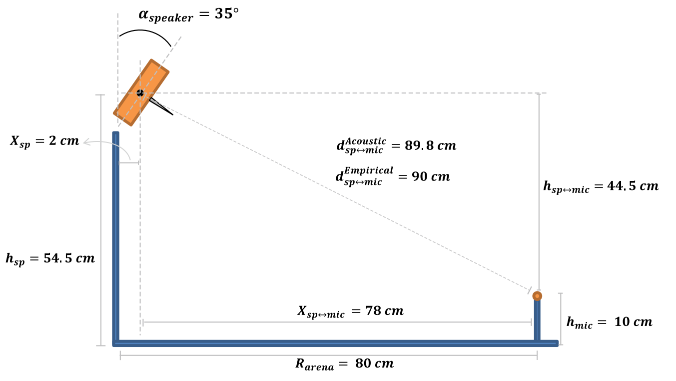

Fig. 2. : Control for the precision of acoustically evaluated distances. A side view of the environment (speaker – orange rectangle, microphone – orange circle). The manually measured distances: R_{arena} is the radius of the arena. X_{sp} is the distance of the speaker's membrane from the wall, due to the 35° tilt. $X_{sp\leftrightarrow mic}$ is the horizontal distance of the microphone from the loudspeaker. h_{sp} is the height of the speaker's membrane center. h_{mic} is the height of the microphone. $h_{sp\leftrightarrow mic}$ is its vertical distance of the microphone from the speaker. Using the Pythagorean theorem we calculate $d^{Empirical}_{sp\leftrightarrow mic}$, the distance between the speaker's membrane and the microphone, based on the manual measurements. $d^{Acoustic}_{sp\leftrightarrow mic}$ is the acoustically evaluated distance (89.8 cm), deviating 0.2 cm (0.22%) from the manually measured one (90 cm) (For interpretation of the references to color in this figure, the reader is referred to the web version of this article).

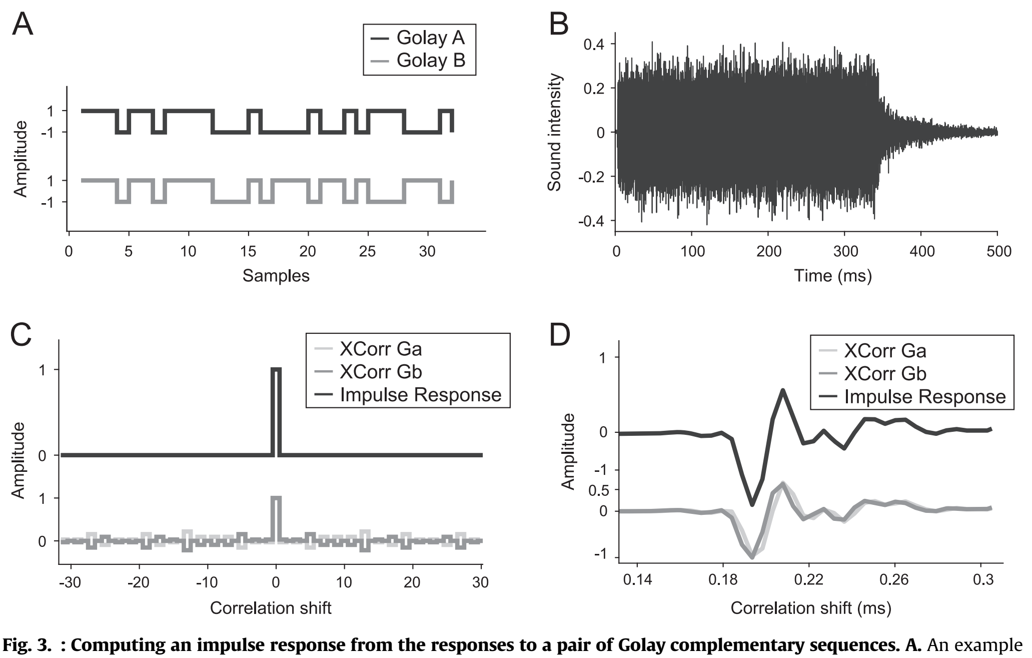

Fig. 3. : Computing an impulse response from the responses to a pair of Golay complementary sequences. A. An example of two complementary Golay sequences of length $2^5 = 32$ bits. **B.** The two auto-correlations of Golay codes (bottom two lines) sum to an impulse at the origin. **C.** Sound recording of a single Golay sequence. **D.** The responses to each Golay sequence is cross-correlated with its original Golay sequence, producing a single cross-correlation. The two cross correlations are shown by bottom plots. When summed, they form an impulse response (the upper dark line).

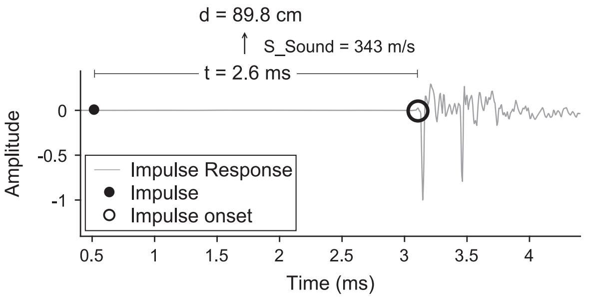

Fig. 4. Impulse response encodes the distance of the speaker to the microphone. An impulse response was recorded at the center of the arena (gray line). Black point represents sound onset, black empty circle is the estimated sound onset at the microphone's location. Time difference of these two points is called time-of-arrival of the sound, in this case it is 2.62 ms. It can be converted to distance by multiplication with the speed of sound: 0.00262 x 343 = 89.8 cm.

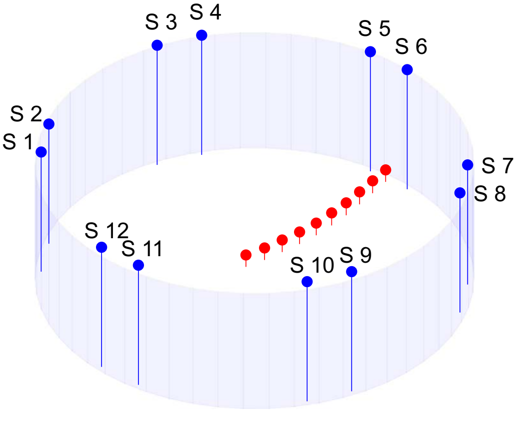

Fig. 5. The sampling location. The acoustic localization of the microphone in the ten locations along a horizontal line. At each location an impulse response was calculated for each of the twelve speakers. The microphone continuously acquired the sounds along all locations, producing a single sound file `sound_samples.wav`.

first onset time is marked by the synchronization trigger, while the second onset was identified in the IR using a custom MATLAB code (provided in the `get_imp_onset()` method of the `singleLoc` class). The algorithm searched for the first large amplitude deflection following the low-amplitude region at the beginning of the IR, which reflected the travel time of the sound (Fig. 4, black circle). The difference between the two onsets was denoted as 'timing of arrival' (ToA) and was converted to the traveling distance of the sound wave (distance = ToA $\times$ speed of sound at 20° C, considered as 343 m/s).

2.5. Localization of a speaker in 3D

The acoustic 3D localization was performed according to the protocol described in Kazakov and Nelken [1] (Sections 2.5–2.6). The calculation of the distance between a speaker and the microphone is depicted in Fig. 4.

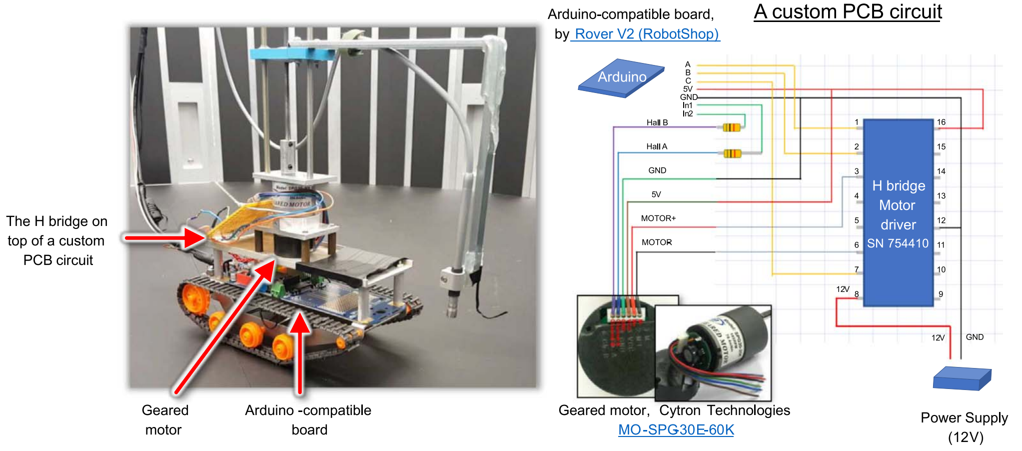

Fig. 6. : Electrical circuitry that enables the vertical movement of the microphone. The bottom part of the robot was assembled from the kit Rover V2 (RobotShop). A geared motor (MO-SPG-30E-60K) was installed on top of it, fixed inside a custom made frame. The motor controlled the elevation of the extension arm, that held the microphone. An Arduino-compatible board (part of the RobotShop kit) controlled the robot's tracks and the geared motor. The later was connected to the board using a custom made PCB circuit, utilizing an H bridge driver.

The script `localize_points_in_3D.m` uses this method to locate the 3D coordinates of the 10 sampling points from the file `horizontal_line.mat` (the result depicted in Fig. 5). The multilateration solution is implemented in the `find_point.m` function.

2.6. *Sampling of the sounds*

Spatial sampling was automated using a custom-built robot, as described in Kazakov and Nelken [1] (Section 2.4) A high-precision geared motor (Cytron Technologies, MO-SPG-30E-60K) was used to set the elevation of the microphone arm and was operated by an encoder (H bridge motor driver SN754410, Texas Instruments Inc.) connected to the Arduino board on the robot (Fig. 6).

For example, for the attached sound samples (`sound_samples.wav`), the robot was placed between two adjacent speakers with the microphone pointing towards the wall, and its arm was lowered such that the microphone membrane was 5.5 cm above the floor. The robot moved horizontally 8.5 cm between each of the 10 sampling locations (Fig. 5), towards the arena center. In that experiment, all 12 speakers were sampled at each of the 10 locations, resulting in a set of 120 Golay pairs, all stored in the single file `sound_samples.wav`.

Acknowledgements

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Transparency document. Supporting information

Transparency data associated with this article can be found in the online version at https://doi.org/10.1016/j.dib.2018.10.148.

Appendix A. Supporting information

Supplementary data associated with this article can be found in the online version at https://doi.org/10.1016/j.dib.2018.10.148.

References

[1] A. Kazakov, I. Nelken, Acoustic calibration in an echoic environment, J. Neurosci. Methods 309 (2018) 60–70. https://doi.org/10.1016/j.jneumeth.2018.08.025.

[2] B. Zhou, D.M. Green, J.C. Middlebrooks, Characterization of external ear impulse responses using Golay codes, J. Acoust. Soc. Am. 92 (1992) 1169–1171. https://doi.org/10.1121/1.404045.