P&D Information Systems and Signal Processing

Week 2: February 17-21, 2020
Linear and head-mounted microphone array recordings

Giuliano Bernardi, Randall Ali, Santiago Ruiz[1], Marc Moonen

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Prelimenary remarks:

- The aim of this session is to record a set of signals for a linear microphone array and a head-mounted microphone array. For each, two target audio sources (loudspeakers) will be placed at two different angles. Both target audio sources have to be recorded individually for each microphone array.
- The recording session will be carried out in the audio lab (room ELEC 02.12). A computer room is also booked (Check online) for the same time slot so you can work on your code while waiting for your turn to record.
- The group will be divided in three groups of 4-5 students.
- Student are advised to read an introduction to the digital audio workstation (DAW) software used in the lab (Logic Pro X), however this will be briefly cover in the general introduction to the lab that will be given to each group.
- The signals to be recorded, the angles and responsible group are given in the following table.

File name	**angles**	**Responsible**
part1_track1_dry.wav	-90°	GROUP 1
part1_track2_dry.wav	90°	
part2_track1_dry.wav	-60°	GROUP 2
part2_track2_dry.wav	60°	
part2_track1_dry.wav	-90°	GROUP 3
part2_track2_dry.wav	90°	

  Each group must upload their files to Toledo after the measurement session so that everyone has all data.

- All the equipment in the lab is very delicate and expensive, so please treat them responsibly.

Part 1: Linear microphone array measurements

1. A 4-microphone array is set up. Measure the inter-microphone distance and the distance from the floor. The convention for enumerating the microphones is as shown in Figure 1

[1]For questions and remarks: santiago.ruiz@esat.kuleuven.be.

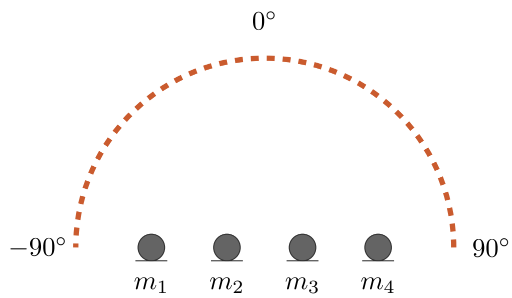

Figure 1: Convention used to enumerate the microphones in the linear microphone array.

2. Connect the XLR cables of the microphones to the corresponding inputs in the wall. Make sure to do it systematically so that everyone in the group can quickly understand the signal flow.

3. Place the loudspeaker(s) at the correct angle and 1m from the center of the microphone array. Write down the height of the microphone array and place the loudspeaker at the same height.

4. In the computer go to the folder "P&D_ISSP_2020" in Documents, there you will find the files needed for the recording. Open the DAW and create a new project from the template PD_issp_template.

5. Assign the inputs and outputs to the audio tracks according to your signal flow.

6. Set the gains of the pre-amplifiers of microphones in the array in the audio interface to 45. Set the loudspeaker(s) gain(s) to -25dB in the software Totalmix.

7. Import the white noise signal in the folder and record it. This will serve as a calibration procedure.

8. Record the assigned signals individually using all the microphones at the same time. Assess if a second recording is needed.

9. Listen to the recorded signals to verify they were recorded correctly.

10. Export (bounce) the recorded signals in WAVE format and name each file following this convention

    LMA_Mn_angle.wav
    LMA_Mn_angle.wav

    where n is the microphone number according to Figure 1, angle stands for the angle of the loudspeaker.

11. Make sure to copy these files and upload them to Toledo.

Part 2: Head-mounted microphone array measurements

1. There is a mannequin head set up in the lab (KU 100). attach the hearing aids to the ears of the head and connect them to the pre-amplifier.

   Remark: This pre-amplifier is battery powered, hence it de charges very fast. For longer battery life turn it on only when necessary. There are two rechargeable batteries, if one of the runs out please recharge it for the next group.

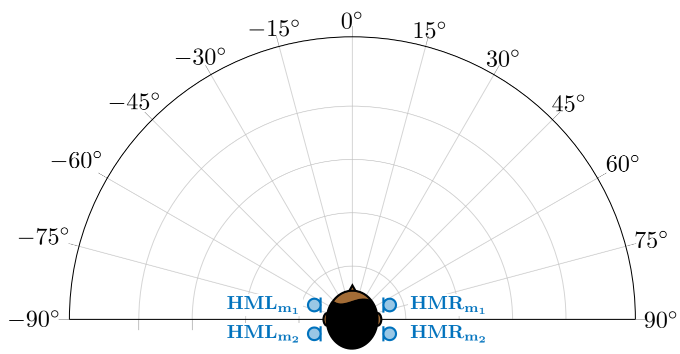

Figure 2: Convention used to enumerate the microphones in the head-mounted microphone array.

2. Connect the outputs of the pre-amplifier to the inputs in the wall. Make sure to do it systematically so that everyone in the group can quickly understand the signal flow.
3. Place the head where the linear microphone array is and the loudspeaker(s) at the correct angle and 1m away from the center of the head. Measure the height of the mannequin head and make sure the same height is used for the loudspeaker.
4. Set the gains of the head-mounted microphones to 25 in the audio interface.
5. Assign the inputs and outputs to the audio tracks in the DAW according to your signal flow.
6. Perform the calibration procedure with the white noise signal again for the head-mounted microphone array.
7. Record the assigned signals individually using all the microphones at the same time.
8. Listen to the recorded signals to verify they were recorded correctly.
9. Export (bounce) the signals in WAVE format and name each file following this convention

   HML_Mn_angle.wav
   HMR_Mn_angle.wav

   where L represents the microphones on the left ear and R the ones on the right ear.
10. Make sure to copy these files and upload them to Toledo.