Acoustic laboratories ♯6
Room modal behavior at low frequencies

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Goals: In this sessions, the students will characterize the reverberation chamber in the low-frequencies. Two phases wull be undertaken:

- a first simulation with COMSOL Multiphysics
- an experimental assessment in the frequency and time domains (using the NetdB Hardware)

Material

For the laboratory, you will use:

- COMSOL Multiphysics
- a reverberation room
- a closed-box loudspeaker
- a Bruel & Kjaer Type 2706 power amplifier
- a Doppler Laser Velocimeter (sensor Head OFV-505 + Controller OFV-5000)
- 7 PCB microphones (Type 378B02)
- a NetdB Multichannel Sound and Vibration Analyzer
- a computer with the NetdB Software Suite
- (optional) a set of active Electroacoustic Absorbers

PART I: Room modes modeling on COMSOL Mulitphysics - Eigenfrequencies

This part of the exercise is intentionally succinctly described since you are supposed to perfectly master the COMSOL modeling environment...
The first phase consists in modeling the reverberation room in the low frequency. The main outcome of this study is the retrieval of the first resonance frequencies of the room, as well as the modal decay times for the first modes.

Setting the model

Open COMSOL Multiphysics, and open a new file. Select the "Model Wizard" option, and chose the 3D "Space dimension".
Select the "Pressure Acoustics, Frequency Domain" interface, and chose the "Eigenfrequency" study. Validate by clicking "Done".

In the "Geometry" node, select "Import" and browse to search for the folder where you saved the room geometry "Reverberation_Chamber_geometry.mphbin". Click on "Build all objects" to display the room geometry.

Setting the Physics

In the Eigenmodes study, the aim is to identify the eigenvalues (resonance frequencies and damping coefficients) and eigenfunctions (mode shape across the room) of the Helmholtz equation in 3D:

$$\text{Helmholtz equation: } \nabla^2 p + k^2 p = 0$$
$$\text{with boundary conditions: } \frac{Z_n}{\rho c}\vec{\nabla} p.\vec{n} = -jkp,$$

where $\vec{k}$ denotes the wavevector ($|\vec{k}| = \frac{\omega}{c}$), and Z_n denotes the acoustic impedance of wall n. Therefore, the only input in this mode are:

- the propagating medium properties (sound celerity c , mass density ρ)
- the boundary conditions (walls properties)

The computed Eigenfrequencies are complex values $\nu_i = f_i + j\frac{\delta_i}{2\pi}$ solving the Helmholtz equation, where the real part are the actual resonance frequencies f_i, and the imaginary part depends on the damping coefficient δ_i.

First, let's set the propagation medium. Go to the "Material" node and select "Air". Assign the material "Air(mat1)" to the room domain. Define the parameters $\rho_{air} = 1.2$ kg.m^{-3} and $c_{air} = 343$ m.s^{-1} in the "Global Definitions/Parameters" node. Assign these values to the "Air(mat1)" properties.

Let's now assume the reverberation chamber walls present a homogeneous sound absorption, that is assimilated to a pure resistance, independent on frequency.

Question 1 Knowing the average absorption coefficient of the room's walls is $\alpha_{\text{walls}} = 0.01$, what boundary condition should you assign to the room walls?

Setting the mesh

Define a Tetrahedral meshing so that it presents at least 6 mesh elements per wavelength at the highest frequency (here $f_{max} = 100$ Hz).

Preliminary tests

These preliminary assessments are intended at adjusting the number of eigenmodes to analyze. Here, we want to limit the room modes analysis to the frequency bandwidth [20 - 100 Hz].

Question 2 In the Study/Step1:Eigenfrequency node, select a value (let's say 10) for "Desired number of eigenfrequencies" and start the study. Look at the real part of the last Eigenfrequencies $f_i = \Re(\nu_i)$, and iterate the "Desired number of Eigenfrequencies" until you reach $f_{N\text{max}} \approx 100$ Hz. What is the highest number Nmax?

Analysis of the room eigenfrequencies and modal decay times

We define the Modal Decay Time as the counterpart of the Reverberation Time in the modal (low-frequency) domain, ie. corresponding to a 60 dB drop of the sound pressure level after a sound source exciting a given mode i is stopped. We remind that the time dependence of each eigenfunction (ie. pressure p_i due to each modal contribution) is of the form $e^{-\delta_i.t}$.

Question 3 Find a simple relationship linking the modal decay time $\text{MT}_{60,i}$ and the damping coefficient δ_i for each mode i.

Question 4 List in a table the values of f_i, δ_i, and $\mathrm{MT}_{60,i}$.

Question 5 If possible, try to identify the category of each mode (axial / tangential / oblique mode).

Question 6 Display and comment the first 7 eigenmodes of the room (you'd rather display sound pressure levels in dB re. 20 μPa instead of sound pressure amplitudes).

PART II: Room modes modeling on COMSOL Mulitphysics - Frequency responses of the room

In this part, we will simulate the response of the room to harmonic excitations with an ideal sound source at one corner.

Setting the model

- Take the geometrical model of the former part and define a new "Frequency-Domain" Study .
- Measure the dimensions of the closed-box loudspeaker, including the loudspeaker diaphragm diameter. Model the loudspeaker cabinet as a block in the "Geometry" node, and position this box close to the corner (0,0,0) of the room, so that the longest dimension is directed towards the x axis. Finally draw a circle denoting the loudspeaker diaphragm on the face opposed to the wall. Don't forget to extract the internal volume of the loudspeaker box from the volume of the room (there should be only one volume of air in the end).
- Assign the loudspeaker diaphragm a vibratory velocity of amplitude $v_0 = 10^{-3}$ m/s.
- Place 7 points inside the room that will represent the microphones positions in the further experimental part of the lab. Make sure the chosen position are accessible experimentally. You are encouraged to chose one corner among the 7 microphones positions.
- Define the 7 sound pressures p_i measured at the 7 microphones positions as variables in the Model Definitions. In that view, you should use the "Average" function for each microphone position, and then define p_i as the average pressure at each position.
- Then you will define the transfer function between the sound source v_0 and each pressure p_i as $H_i = \dfrac{p_i}{v_0}$
- Define the study frequencies between 20 Hz and 100 Hz, with **0.1 Hz** resolution.
- Start the Study2 "Frequency Domain" (it might take $\approx$ 30 minutes).

Question 7 Display the sound pressure level distribution at the selected frequencies of the last part.

Question 8 Display the transfer function at all 7 microphones positions.

PART III: Measurement of low-frequency modes in the reverberation chamber

In this section, you will proceed to the characterization of the room modes, first in the frequency domain, then in the time domain.

Experimental protocol

The Figure 1 shows a schematic description of the experimental protocol. The sensors are limited to 8, including a Laser Doppler Velocimeter (LDV) and 7 PCB Type 378B02 microphones (which are prepolarized IEPE microphones).

- First connect all hardware together. It might be practical to place the NetdB, the LDV controller and the B&K power amplifier inside the reverberant chamber, while controlling everything from the computer outside the reverberant chamber.

Important notice: the LDV is a class 2 laser source, therefore it is dangerous for the eyes. As a rule, the operator working with the LDV should NEVER look directly to the laser source. Safety googles are available to protect your eyes.
Moreover, the sound pressure levels might be quite high in the reverberant chamber while in use. Do not forget to wear hearing protection in case you need to enter the room while in service.

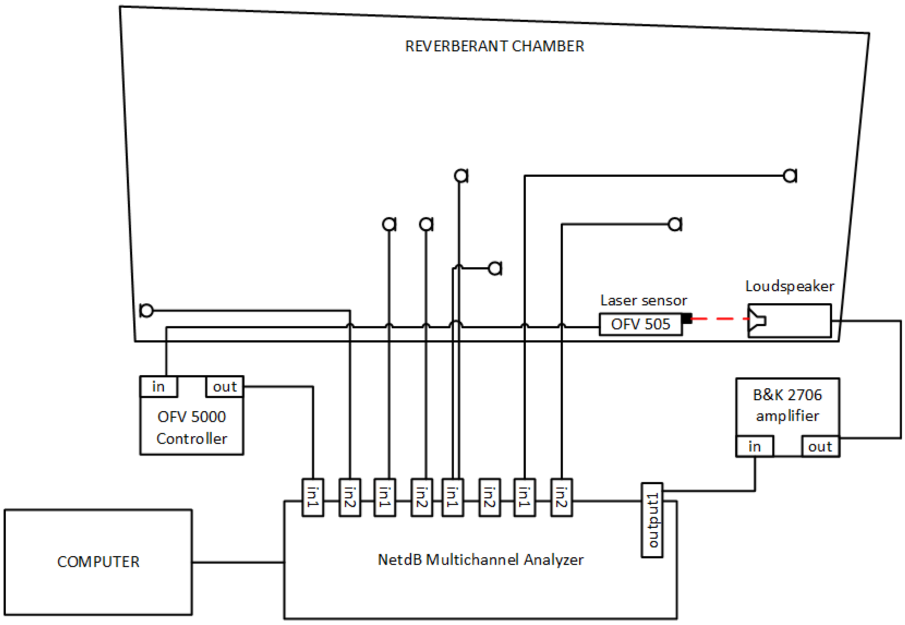

Figure 1: Experimental protocol

Setting the measurement

First of all, you will need to get used to the NetdB Hardware. It allows to do multichannel measurements (up to 12 direct input, and additional polarized input if necessary).

- Open the "dBFA Suite" software on the computer.
- Click on the "Capteurs/Calibreurs" to check that the PCB378B02 microphones (Serial numbers 128109 to 128119) are in the sensors database, and that the Nor1251 Calibrator is also present in the Calibrator database. Close the "Capteurs/Calibreurs" window
- Open the "Analyzeur" icon and select "Configuration → Nouvelle" (don't forget to regularly save the project on a folder, for instance "TP7_Room_modes.Ana").
- Click on the "Paramètres" button

Here we will set the **Acquisition** (sensors), **Génération** (sound source), **Paramètres généraux**, **Traitement** (processing), and **Visualisation** (rendering)

Paramètres généraux

You will define here the container file in which all measurements will be stored. It is important that you specify this at the very beginning of the project since you will need to retrieve all your data at this location.
Create a file "TP7_room_modes.cmg" in your work folder.

Acquisition

Here you will select the 8 first channels, and assign them the corresponding sensors:

Ch.	Actif	Nom	Description	Capteur	Unité	Freq. Echant. (Hz)	Sensibilité (V/Up)
1	✓	#1	Velocity	Laser Polytec OFV-5000	m/s	51200	20
2	✓	#2	Mic01	1/2"PCB378B02-128109	Pa	51200	1
3	✓	#3	Mic02	1/2"PCB378B02-128110	Pa	51200	1
4	✓	#4	Mic03	1/2"PCB378B02-128111	Pa	51200	1
5	✓	#5	Mic04	1/2"PCB378B02-128112	Pa	51200	1
6	✓	#6	Mic05	1/2"PCB378B02-128113	Pa	51200	1
7	✓	#7	Mic06	1/2"PCB378B02-128114	Pa	51200	1
8	✓	#8	Mic07	1/2"PCB378B02-128115	Pa	51200	1

Once you have filled the Acquisition table, you will specify the prepolarization of the PCB microphones (channel 2 to 8) by clicking on the right arrow at the top of the table. Then change the Inputmode field from "DIRECT" to "IEPE. Keep the other settings unchanged. Repeat this step for all 7 microphones.
Finalize the acquisition setting by clicking on the padlock icon. It should be locked to validate and go to the next step.

Note that by default the software has assigned a sensitivity of 20 [V/(m/s)] to the LDV. This sensitivity is linked to the output range of the controller (the blue unit attached to the LDV), which is to be adjusted, depending on the excitation of the loudspeaker. After some preliminary tests, you should be able to decide to which range you should set the laser controller.

- On the Vibrometer Controller display, check the actual sensitivity ("Range" value) and modify the value in the "Acquisition" table accordingly.

We will now calibrate the microphones.

- Plug all microphones to the corresponding inputs of the NetdB hardware (channel 2 to 8).
- Switch the Nor1251 calibrator on and fix it to the first microphone.
- Select the microphone in the Acquisition Table and click on "Contrôle des voies" (the icon with a pen on a waveform).
- Select the Nor1251 calibrator from the list.
- Click on the "Ajuster" icon (balance)
- Once stabilized, click on Valider (the blue point at the bottom)
- You should see the sensitivity updated to a new value (in the range of 50 mV/Pa).
- Repeat this step for all 7 microphones.

Paramétrage Génération

Click on the "Paramétrage Génération" icon (the second from the top in the column on the left).
In the "Paramétrage Génération" section, select channel 1 and select Pink Noise in the "Type Generation" field. Leave all the parameters unchanged (Amplitude = 1 (VRms)).

Traitement

Click on the "Traitement" icon (the 4^{th} from the top in the column on the left).
In the "Traitement" section:

- Tick the "Enreg. temporel" box (this is to record the time waveforms of all sensors)
- Chose the FFT processing and validate by clicking on the right arrow symbol

- Modifiy the parameters of the FFT processing to get the following setting
  - Hanning windowing
  - 75% overlapping
  - bandwidth 156 Hz
  - frequency resolution $\approx$ 0.1 Hz
  - Averaging type: linear
  - Duration of measurements: $\approx$ 60 s
- We also want to process transfer functions (ie. activate H_1 instead of G_{xx}) to get the function $H_i = \dfrac{\text{pressure}(mic_i)}{\text{velocity}}$. For that, set the reference channel to the velocity channel (#1).
  - Right-click on the field "Voies de réf." and tick "1" only.
  - Untick Gxy and tick $H1$

Visualisation

Click on the "Visualisation" icon (the last in the column on the left).
In the "Traitement" section:

- Select all time-domain signals (SIG_51200)
- Select all transfer functions (FFT_H1)
  - Select "Module" for displaying the amplitudes of H1
  - Select PWR unit
  - select 60 dB dynamics

Acquisition

Once the measurement parameters are set, you can validate the "Paramétrage" window by clicking on the green "tick" button at the bootom left side of the window. You should see the acquisition window containing two graphs: 'SigPlot_S' on the left displaying the instantaneous signals acquired by the 8 sensors, and 'FFT_H1_S' on the right, displaying the transfer functions between the 7 microphones and the laser velocimeter. You should also see a pop-up window with the label "Output control", displaying a loudspeaker icon and a time-count. The loudspeaker icon should be "muted".
Last, at the bottom right of the window, you should see an editable (white) field with the label "Enregistrement". You are advised to change this label for every recording sessions (eg. FRF_01, for the first attempt to measure the frequency responses functions at the 8 mic positions), so that you can easily recover the measurements once you have measured everything (this helps tagging the different measurement sessions). The green lights on the right shows the eventual saturation of the acquisition channels.
Do not forget to save the Analysis ('*.Ana') session.

Triggering the measurements

You can also select some trigger functions for the measurements, that allows setting the start and end of your measurements. It may be useless for this first part, since the Frequency responses measurements will be done with stationnary noise that you can start and end manually, but it might be useful for the last measurements in the time domain (modal decay time). The following gives you a simple example:

- Click on the icon "Triggers"
- unlock the padlock icon
- click in the "Activer" column, and then click on the '+' icon while the "Seuil" label is visible in the top-left field.

- in the Thresh_1 field, select a threshold of 0.01 m/s on Channel 1 (the loudspeaker movement will trigger the measurement)
- in the "Stop" column, select a duration of 60 s (this should be enough to stabilize the averaging of the transfer functions)
- lock the padlock

Measurement of the frequency responses at the 7 microphones positions

In the acquisition window, label the 1^{st} measurement 'FRF_01'.
The measurement session is first launched by clicking on the green "Play" button. However, the measurement won't effectively start before you click the red "Record" button. When the "Play" button is clicked, you can start the sound source by unmuting the loudspeaker icon. Once done, you can start the measurement which should last around 60 s (either manually or by setting the trigger as described before).

Security check:
Once the measurement is launched:

- look at the bottom right part of the acquisition window and ensure there is no red light. In such a case, you turn them green just by clicking on it (it adjusts the input dynamics to avoid saturations);
- look at the bottom left part of the acquisition window and check the status message reads "En cours d'enregistrement ...". If the message reads "Attente de trigger...", that means that the loudspeaker is vibrating below the trigger threshold, and you should change this either by augmenting the amplifier level, or adjusting the trigger level;
- while the measurement is occurring, a time count is running on the right-bottom side of the window;
- check that the measurement actually finishes after the set duration (in the trigger window). You should see the time count stopped at the end of the measurement (but it does not automatically shut the source off!...).

Once the measurement is finished, it should have stored the results in the TP7_room_modes.cmg file (the container declared before). Check the measurements:

- go to the main page of the dBFA Suite
- click on Post-processing
- click on the "Open" icon and select the .cmg container file (the one you have declared in the "Paramètres Généraux" section)
- you should see a series of rows, corresponding to both time recordings (Signal) of the 8 input channel and 7 transfer functions ('Fonction de transfert H1') for the 7 microphones with the velocity reference. Note that the column "Information" reads the name of the measurement ('FRF_01'). You can visualize the data by clicking on the plot icon.

Exporting to matlab

- Open the windows explorer and go to 'C:/Program Files (x86)/01dB-Metravib/dBFA Suite 4.9.0/Cmg2Mat'. You should see a matlab icon with the name cmg2mat.exe.
- Drag the TP7_room_modes.cmg file to the cmg2mat.exe icon. It should create a series of mat files in your working folder, with the following names: TP7_room_modesi.mat, where i is an integer starting from 0.
- Open Matlab, and check that the 8 first mat files correspond to 'Signal' data series, and the 7 last correspond to H1 transfer function. Check the parameters of the H1.

Question 9 Plot the frequency responses measured at all 7 microphones positions, and compare the result to the ones obtained in the COMSOL simulations. Discuss the result.

Question 10 Identify the values of the first 7 resonance frequency measured in the room. Compare these values to the COMSOL results, and provide an estimate of the relative difference.

Measurement of the modal decay times

We will now measure the time evolution of the sound pressure when the room is excited by a time-limited stationary pure tone at different discrete frequencies (corresponding to the 7 first resonance frequencies of the room). Here, we don't need the velocimeter anymore. You are advised to switch it off since the controller fan makes a significant noise that may corrupt the measurements.

The principle here is, for each room mode, to excite the room at the corresponding frequency during a certain duration $T_{\text{stationnary}}$, and then allow the sound amplitude to vanish completely during a given time T_{decay}. For that, you will need to both change the excitation setting (see section "Paramètres Génération"), the acquistion setting (we don't need H1 transfer function anymore), and the trigger setting.

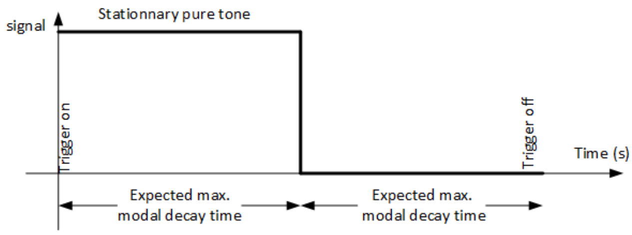

Figure 2: Schematic representation of the excitation signal for the modal decay time measurements

Question 11 Based on the computed modal decay time identified in the first session, to what maximal duration of extinction should you expect in the room?

- set the "output" so that it generates a pure tone at desired frequency, during a limited time $T_{\text{stationnary}}$
- untick the FFT processing in the "Traitement" section;
- Set the trigger so that the measurement spans at least over $T_{\text{stationnary}} + T_{\text{decay}}$. There is no more LDV now, so you need to use another trigger start (eg. a microphone or alternatively you can set the start manually and discard the trigger start).

Now you can launch the time decay measurement (do not forget to change the measurement label to 'MT60'!). Follow the same steps as in the preceding section. You should have 7 measurements in file TP7_room_modes.cmg.

Question 12 Propose a method to retrieve the modal decay time out of the time-series of the sound amplitude decay in the room. Present the results in a table and compare to the ones obtained with the COMSOL simulations.

PART IV: (optional) Measurement of low-frequency sound absorbers in the reverberation chamber

Repeat the room mode measurement of PART III with 4 Active Sound Absorbers dispatched at the 4 corners of the reverberant chamber.

Question 13 Present the measurement results and comment on the performance of the Active Sound Absorbers.