Applied Acoustics, Chalmers

Introduction to Audio Technology & Acoustics

VTA 137

January 5, 2022, 8.30 – 13.30, SB-building

Instructors: Jens Ahrens, Wolfgang Kropp, Pontus Thorsson

Questions during the exam: Jens Ahrens, Ph 2210 (will visit the examination room at 10.00 and 12.30)

Solutions will be posted on the department's bulletin board on Jan 7, 2022.

Preliminary exam results will be posted on the department's bulletin board latest Jan 25, 2022.

Questions on corrected exams can be discussed in the time slot announced together with the preliminary results.

Final results submitted to the university latest Jan 27, 2022.

Permitted material at the exam: Mathematical tables in books like the Physics Handbook, Tefyma, Beta, or similar books, acceptable calculators as defined below, the "Useful Formulae" - collection for Audio Technology and Acoustics, distributed at the course and available at the department's web site and the audio formula collection distributed at the course and available at the department's web site

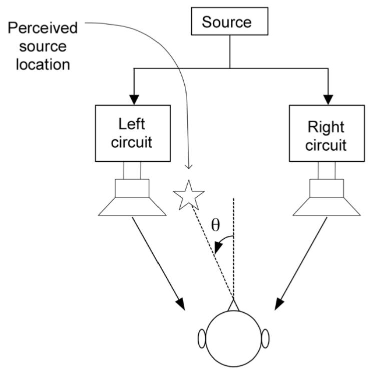
An acceptable calculator is a calculator without (or cleared) text memory.

Grades: 12p = 3, 18p = 4, 24p = 5

Do not forget to specify the assumptions made in your solutions to the problems!

PROBLEM 1

a) A common stereo system setup is shown in the figure below. If the same source signal is fed to both loudspeakers, listeners perceive the sound as coming from the position between the loudspeakers (at $\theta=0°$). In such a system, it is however possible to change the perceived source angle by introducing circuits which alter the signal before being sent to the loudspeakers. What two types of circuits can one use to change the perceived source location? Explain briefly how these circuits should be adjusted so that the source is perceived as coming from e.g. more to the left (as shown). (1p)



b) We want to implement a simple "surround" system by placing a third loudspeaker behind the listener in the figure above. A requirement on the system is that the sound still should be localized to the front stage (somewhere between L and R loudspeakers) after introducing this third loudspeaker. Explain what one could do in order to fulfil this requirement. (2p)

c) What is meant by "frequency masking" as applied to our hearing? Explain why intermodulation distortion should be more audible than harmonic distortion for a loudspeaker according to the following example:

Harmonic distortion means that a pure sine tone is reproduced together with weaker overtones. Assume that the loudspeaker, when emitting a sine tone at 1 kHz at a level of 80 dB, also produces overtones at 2000 Hz (at a level of 40 dB) and 3000 Hz (at a level of 35 dB).

Intermodulation distortion means that when two sine tones are reproduced, difference frequencies are also produced (in addition to the overtones described above). If the loudspeaker reproduces 9 kHz and 10 kHz at 80 dB, then 1 kHz is also produced at a level of 40 dB. (2p)

PROBLEM 2

Matilda has to close her nightclub temporarily due to the new covid19 restrictions. She decides to use this time period to improve the acoustics in her nightclub and asks an acoustic consultant for help. The consultant measures the reverberation time in the 500-Hz octave band (T=2.9 s) and recommends that it should be lowered to 0.7 s. After inspecting the walls and the floor he is confident that the absorption coefficient of these surface is approximately 0.05, but he is unsure about the ceiling.

Dimensions of the nightclub: length 15 m, width 10 m, height 3 m

a) What is the absorption coefficient of the ceiling? (2p)

b) To lower the reverberation time, the acoustic consultant decides to install absorber panels covering 2/3 of the ceiling. Which absorption coefficient α_{abs} do they need to have to obtain the desired reverberation time? (2p)

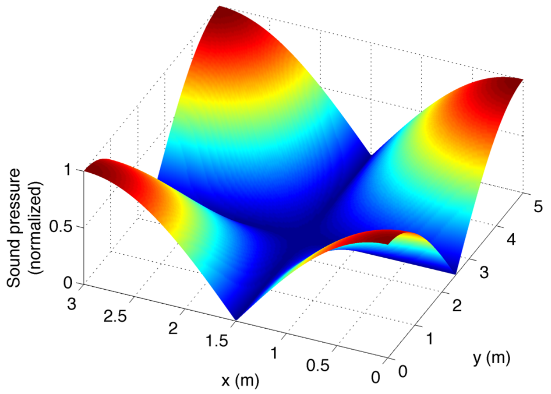
c) What is the reduction in sound pressure level in the reverberant field in comparison to the original situation when the absorbers with α_{abs} have been installed as described in b)? (1p)

(Speed of sound in air: c = 343 m/s)

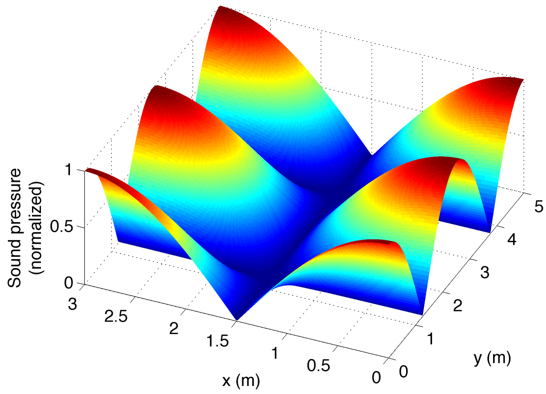
PROBLEM 3

In the control room of Albert's recording studio, the two modes shown in the figure below are particularly problematic. They cause a clearly audible drop in the bass region in Albert's usual position (at x=1.6 m, y= 2.5 m).

a) What are the eigenfrequencies belonging to these two modes? (1p)



Mode I

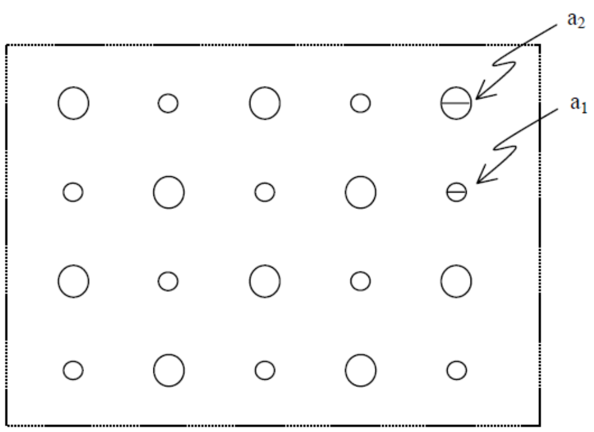


Mode II

Note: For both modes, there are no sound pressure variations in the z-direction.

b) If only mode II in the figure above is excited, what is the sound pressure level in Albert's position (at x=1.6 m, y= 2.5 m), relative to the sound pressure in one of the corners? (1p)

c) Albert's friend (who studied acoustics) provides Albert with the perforated gypsum board shown below so that Albert can build a resonant absorber that effectively damps both mode I and mode II. At what distance from the wall should Albert mount the board? (3p)



<u>Perforated gypsum board</u>

Thickness = 3.5 cm

Hole diameters:
a_1= 1.26 cm ; a_2 = 1.76 cm

Distance between center of holes: 12.5 cm

(Speed of sound in air: c=343 m/s)

PROBLEM 4

You are making a recording, and you have chosen a microphone with a sensitivity of 15 mV / Pa. It happens so that your pre-amplifier is set to provide 50 dB of gain. The output signal appears at a line level of +6 dBu. Note that the dBu reference is $\sqrt{0.6}$ Vrms.

a) You decide to replace the microphone with a more sensitive one, but you don't know what the sensitivity is exactly. With the new microphone, you reduce the per-amplifier gain to 45 dB, and then you obtain the line level +15 dBu. What is the sensitivity of the new microphone?

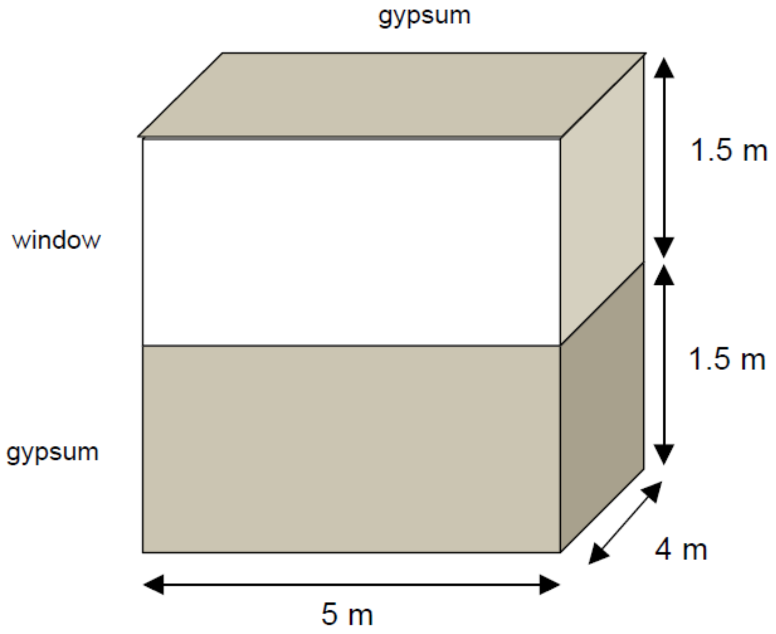
(4 p)

b) If the maximum line level is +24 dBu, what would be the maximum SPL at the new microphone before clipping with this gain setting?

(1 p)

PROBLEM 5

A small office cabin has been set up inside a large factory hall. Half of the surface of the cabin's walls are windows, half are gypsum panels, and its roof is also made of gypsum (see figure). The floor is perfectly rigid. The reduction indices for the different materials are given in the table below. The reverberation times inside the office cabin as well as the sound pressure levels $L_{p,factory}$ in the factory hall (measured in the reverberant field) are also given in the table.



f (Hz)	500	1k
R_{window} (dB)	29	35
R_{gypsum} (dB)	32	38
$Lp_{factory}$ (dB)	75	74
T_{office} (s)	0.5	0.5

a) What is the total A-weighted sound pressure level inside the office cabin, caused by the factory noise from the hall outside? (3 p)

b) A computer inside the office radiates noise with a sound power level of (52; 55) dB at (500 Hz; 1 kHz). What is the total A-weighted sound pressure level at 2 m distance from this source, combined with the noise calculated in a)? Assume that the computer radiates sound omnidirectionally. (2 p)

(Speed of sound in air: c = 343 m/s)

PROBLEM 6

A stereo recording is made with a Blumlein pair, i.e. two figure-of-8 microphones are placed at 90° angle to each other, ±45° to the frontal direction.

a) Assume that the recording is made in an anechoic environment and that the source is placed at 30° angle to the left of the frontal line and at 2 m distance from the microphones. What is the output level difference between the Left and Right channels? The microphone is placed in the same horizontal plane as the source. (2 p)

b) What will the level difference be if the environment is a room with volume 11880 m^3 and reverberation time 5.3 s? Assume that the output level from the microphone can be calculated as

$$L_{\mathrm{MIC}} = G + 10\log\left[\frac{\mathrm{DF}_{\mathrm{SRC}}\mathrm{DF}_{\mathrm{MIC}}}{4\pi r^2} + \frac{4}{\mathrm{A}}\right] \quad \mathrm{dBu}$$

where G is a constant, $\mathrm{DF}_{\mathrm{SRC}}$ and $\mathrm{DF}_{\mathrm{MIC}}$ are the directivity factors of the source and the microphone, respectively. The source is a point source.

(3 p)

Hint: The output from a figure-of-8 microphone is proportional to cos θ, where θ is the angle between the on-axis direction of the microphone and the angle of sound incidence. The Directivity Index is 4.8 dB for $\theta = 0$. Furthermore, $\mathrm{DI} = 10\log_{10}(\mathrm{DF})$.

(Speed of sound in air: c = 343 m/s)