# Audio Technology & Acoustics Exam 2022-01-26

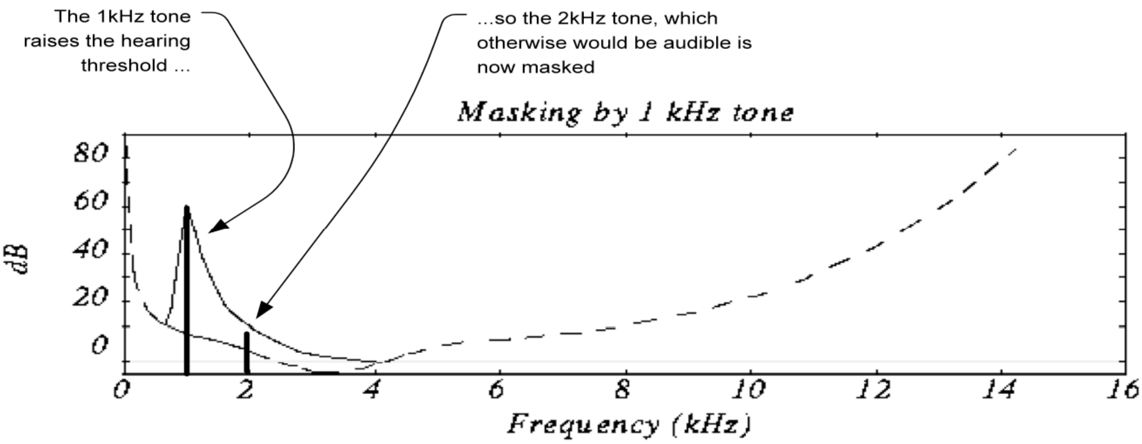
## Problem 1 a)

Changing the angle of the so-called phantom sound source in a stereo system can be achieved by using either amplitude or time differences between the channels or a combination of both. (This is mainly due to the fact that the human localization mechanism is based on interaural level and time differences). The circuits shown in the figure could therefore be either amplifiers or time delay units. A phantom sound source located more to the left could be achieved by either lowering the amplitude of the right loudspeaker / increasing the left loudspeaker amplitude ('amplitude panning') or by delaying the right loudspeaker. The amplitude panning technique is the far most common panning technique in stereo systems (amplifiers, mixing consoles, etc.).

## Problem 1 b)

Assuming that the same signal is sent to all three loudspeakers, we would need to make sure that the rear loudspeaker's sound arrives after the front stage loudspeakers' sound, because of the "law of the first wavefront". The law of the first wavefront states that humans localize the sound to the origin of the sound which first arrives at our ears, under the condition that the later arriving sound is not much louder than the first sound. Therefore, we would either have to place the third loudspeaker at a "safe" distance from the listener or introduce a delay unit which makes sure that the front stage sound arrives first. Delaying the rear sound too much (about 20ms) may however cause it to be perceived as an echo, which is probably not a wanted feature of the system. (In Dolby Pro-Logic surround systems, the rear sound is both delayed and band limited to avoid rear localization artefacts)

## Problem 1c)




Frequency masking means that a tone having a certain frequency (masker) raises the hearing threshold around the tone's frequency so that another, weaker tone, having a frequency close to he masker's frequency, presented at the same time will not be heard. The threshold shift caused by the masker is greater above the masker's frequency than for frequencies below the masker's frequency as shown in the figure. In the example, intermodulation distortion should be more audible because the distortion's frequency component has, first of all, a frequency which is not close to the undistorted sound and second, a frequency which is lower than the undistorted sound (masking less efficient).

## Problem 2a

Assume diffuse sound field

Volume: $V = l \cdot b \cdot h = 15\text{ m} \cdot 10\text{ m} \cdot 3\text{ m} = 450\text{ m}^3$

Surfaces: floor $S_f = l \cdot b = 15\text{ m} \cdot 10\text{ m} = 150\text{ m}^2$

ceiling $S_c = S_f = 150\text{ m}^2$

walls $S_w = 2 \cdot (l \cdot h + b \cdot h) = 2 \cdot (15\text{ m} \cdot 3\text{ m} + 10\text{ m} \cdot 3\text{ m}) = 150\text{ m}^2$

$S_{wf} = S_w + S_f = 300\text{ m}^2$

$S = S_w + S_f + S_c = 450\text{ m}^2$

$$T = \frac{55\,V}{c\,A}$$

$$A = \frac{55\,V}{c\,T} = \frac{55 \cdot 450\text{ m}^3}{343\,\frac{\text{m}}{\text{s}} \cdot 2.9\text{ s}} = 24.88\text{ m}^2$$

Sabine's formula: $A = S \cdot \alpha_m$ ; $\alpha_m = \frac{A}{S} = \frac{24.88\text{ m}^2}{450\text{ m}^2} = 0.0553 < 0.3$ (Sabine ok)

$$\alpha_\text{m} = \frac{\sum_i \alpha_i S_i}{S} = \frac{\alpha_\text{wf} \cdot S_\text{wf} + \alpha_\text{c} \cdot S_\text{c}}{S}$$

$$\alpha_\text{c} = \frac{\alpha_\text{m} \cdot S - \alpha_\text{wf} \cdot S_\text{wf}}{S_\text{c}} = \frac{0.0553 \cdot 450\text{ m}^2 - 0.05 \cdot 300\text{ m}^2}{150\text{ m}^2} = 0.0659$$

## Problem 2b

Desired reverberation time: $T_\text{d} = 0.7$ s

$$S_\text{abs} = \frac{2}{3} S_c = \frac{2}{3} 150\text{ m}^2 = 100\text{ m}^2$$

$$A_\text{d} = \frac{55\,V}{c\,T_\text{d}} = \frac{55 \cdot 450\text{ m}^3}{343\,\frac{\text{m}}{\text{s}} \cdot 0.7\text{ s}} = 103.08\text{ m}^2$$

Sabine's formula: $A = S \cdot \alpha_m$ ; $\alpha_m = \frac{A}{S} = \frac{103.08\text{ m}^2}{450\text{ m}^2} = 0.2291 < 0.3$ (Sabine ok)

$$\alpha_\text{m} = \frac{\sum_i \alpha_i S_i}{S} = \frac{\alpha_\text{wf} \cdot S_\text{wf} + \alpha_\text{c} \cdot (S_\text{c} - S_\text{abs}) + \alpha_\text{abs} \cdot S_\text{abs}}{S}$$

$$\alpha_\text{abs} = \frac{\alpha_\text{m} \cdot S - \alpha_\text{wf} \cdot S_\text{wf} - \alpha_\text{c} \cdot (S_\text{c} - S_\text{abs})}{S_\text{abs}}$$

$$= \frac{0.2291 \cdot 450\text{ m}^2 - 0.05 \cdot 300\text{ m}^2 - 0.0659 \cdot (150\text{ m}^2 - 100\text{ m}^2)}{100\text{ m}^2}$$

$$= 0.8480$$

## Problem 2c

Sound pressure level in the reverberant field due to a source with sound power $L_W$

$$L_p = L_W + 10\log\left(\frac{4}{A}\right)$$

Assuming the same source in both cases:

$$\Delta L_p = L_{p,2} - L_{p,1} = L_W + 10\log\left(\frac{4}{A_2}\right) - L_W - 10\log\left(\frac{4}{A_1}\right) = 10\log\left(\frac{A_1}{A_2}\right) =$$

$$10\log\left(\frac{A}{A_d}\right) = 10\log\left(\frac{24.88\text{ m}^2}{103.08\text{ m}^2}\right) \approx -6.2\text{ dB}$$

## Problem 3 a)

Mode I: $q_x = 1$, $q_y = 1$, $q_z = 0$
Mode II: $q_x = 1$, $q_y = 2$, $q_z = 0$

$$f_{1,1,0} = \frac{c}{2}\sqrt{\left(\frac{q_x}{l_x}\right)^2 + \left(\frac{q_y}{l_y}\right)^2 + \left(\frac{q_z}{l_z}\right)^2} = \frac{343}{2}\sqrt{\left(\frac{1}{3}\right)^2 + \left(\frac{1}{5}\right)^2} = 66.7 \text{ Hz}$$

$$f_{1,2,0} = \frac{343}{2}\sqrt{\left(\frac{1}{3}\right)^2 + \left(\frac{2}{5}\right)^2} = 89.3 \text{ Hz}$$

## Problem 3b)

$$|p_{\text{corner}}| = |\hat{p}|$$

$$|p_{\text{pos}}(1.6, 2.5, z)| = \left|\hat{p}\cos\left(\frac{1\pi}{3}1.6\right) cos\left(\frac{2\pi}{5}2.5\right)\right| = 0.1045|\hat{p}|$$

$$\Delta L_p = 20\log\left(\frac{|p_{\text{pos}}(1.6,2.5,z)|}{|p_{\text{corner}}|}\right) = 20\log\left(\frac{0.1045|\hat{p}|}{|\hat{p}|}\right) = -19.6 \text{ dB}$$

## Problem 3c)

Resonance frequency of resonator panel:

$$f_0 = \frac{c}{2\pi}\sqrt{\frac{S}{V\left(l_h + \frac{\pi}{2}b\right)}}$$

S= hole area, V= volume “belonging to” each hole, x= distance between holes, d= distance panel to wall => $V = x^2 d$

$$\left(\frac{2\pi f_0}{c}\right)^2 = \frac{S}{V\left(l_h + \frac{\pi}{2}b\right)} \rightarrow V = \frac{\left(\frac{a}{2}\right)^2\pi}{\left(l_h + \frac{\pi a}{4}\right)}\cdot\left(\frac{c}{2\pi f_0}\right)^2 \rightarrow d = \frac{\left(\frac{a}{2}\right)^2\pi}{\left(l_h + \frac{\pi a}{4}\right)}\cdot\left(\frac{c}{x\cdot 2\pi f_0}\right)^2$$

Smaller hole => lower resonance frequency, hence
$a = a_1$ and $f_0 = 66.7$ Hz => $d_1 = 11.9$ cm
($a = a_2$ and $f_0 = 89.3$ Hz => $d_2 = 11.9$ cm)

Albert should mount the perforated panel 11.9 cm away from wall. (Losses in the resulting resonating system can be achieved by placing mineral wool in the air gap behind the panel.)

## Problem 4

a) Measured in dB, the first microphone has a sensitivity of

$$L_{mic1} = 20\log_{10}\left(\frac{0.015}{\sqrt{0.6}}\right) = -34 \text{ dBu (@ 94 dBSPL)}$$

With 50 dB of gain we obtain the signal level +6 dBu. With 94 dBSPL at the microphone, the output level after gain would be L = - 34 dBu + 50 dBu = 16 dBu. So in this case, the actual SPL at the microphone was 94-(16-6) dBSPL = 84 dBSPL.
With the new microphone, the gain setting was 45 dB. Assuming this microphone has the sensitivity $L_{mic2}$ at 94 dBSPL, we know that $L_{mic2}$ + 45 dBu - 10 dBu = 15 dBu, leading to $L_{mic2}$ = 10 dBu + 15 dBu – 45 dBu = -20 dBu at 94 dBSPL, which corresponds to 76 mV/Pa.

b) The current line level is at 15 dBu, so an increase by 9 dB is the maximum. The maximum SPL would then be 84 dBSPL + 9 dBSPL = 93 dBSPL.

## Problem 5

a)

Check Schröder frequency: $f_{min} = 2000\sqrt{\frac{T}{V}} = 2000\sqrt{\frac{0.5}{60}} = 183$ Hz $\rightarrow OK$

Check Sabine: $\overline{\alpha} = \frac{55V}{cS_{office}T} = \frac{55\cdot 60}{343\cdot 94\cdot 0.5} = 0.2 < 0.3 \rightarrow OK$

$$Lp_{office} = Lp_{factory} - R_{tot} - 10\log\left(\frac{A_{office}}{S_T}\right)$$

$$A_{office} = \frac{55V_{office}}{cT_{office}} = 19.4 \text{ m}^2\text{S}$$

$$S_T = S_{window} + S_{wall,roof} = 2(1.5\cdot 4 + 1.5\cdot 5) + [2(1.5\cdot 4 + 1.5\cdot 5) + 4\cdot 5] = \\ = 27 + 47 = 74\, m^2$$

$$R_{tot} = 10log\left(\frac{\sum_i S_i}{\sum_i S_i 10^{-R_i/10}}\right)$$

| f | 500 | 1k | [Hz] |
|---|---|---|---|
| $R_{tot}$ | 30.7 | 36.7 | [dB] |
| $Lp_{office}$ | 50.2 | 43.2 | [dB] re 20µPa |
| A-weighting | -3 | 0 | [dB] |
| $Lp_{,A,office}$ | 47.2 | 43.2 | [dBA] re 20µPa |
| | | | |

$$Lp_{tot,A} = 10\log\left(10^{4.72} + 10^{4.32}\right) = 48.7\, dBA$$

b)

$$Lp_{computer} = Lw_{computer} + 10log\left(\frac{1}{4\pi r^2} + \frac{4}{A_{office}}\right)$$

Uncorrelated combination of $L_{p,computer}$ and $L_{p,office}$ (from a)), see row * in table below

| f | 500 | 1k | [Hz] |
|---|---|---|---|
| $Lp_{computer}$ | 45.5 | 48.5 | [dB] re 20µPa |
| *Combined | 51.5 | 49.6 | [dB] re 20µPa |
| A-weighting | -3 | 0 | [dB] |
| $Lp_{,A,combined}$ | 48.5 | 49.6 | [dBA] re 20µPa |

$$Lp_{combined,tot,A} = 10\log\left(10^{4.85} + 10^{4.96}\right) = 52.1\,dBA$$

## Problem 6

a)

A pressure gradient microphone has an output proportional to $\cos\theta$. $\theta_L$, the angle to the source from the direction of the Left microphone, is $\theta_L = 45 - 30 = 15°$. The level decrease due to the fact that the source is off-center for the Left microphone becomes

$$\Delta L_{\text{MIC,L}} = 20\log(\cos\theta_L) \approx -0.3\text{dB}.$$

For the Right microphone, $\theta_R = 45 + 30 = 75°$. The level decrease becomes

$$\Delta L_{\text{MIC,R}} = 20\log(\cos\theta_R) \approx -11.7\text{dB}$$

The difference in channel level becomes $\Delta L_{\text{u}} = \Delta L_{\text{MIC,L}} - \Delta L_{\text{MIC,R}} = 11.4\text{dB}$.

b)

The microphone is now placed in a diffuse sound field and we need to calculate the room absorption. Sabine's formula gives us the absorption area as

$$A = \frac{0.16V}{T_{60}} = \frac{0.16 \times 11880}{5.3} \approx 359\text{m}^2\text{Sabine}$$

The source is a point source and hence $DF_{\text{SRC}} = 1$. The figure-8 microphone has $DI = 4.8$ for $\theta = 0$, but in this case the source is 15° off center, so the output level is 0.3 dB lower and the directivity in the source direction becomes $4.8 - 0.3 = 4.5$dB. The directional factor is calculated according to the given formula as $DF_{\text{MIC}} = 10^{(DI-0.3)/10} \approx 2.8$.

For the Left microphone we then obtain

$$L_{\text{MIC,L}} = G + 10\log\left[\frac{2.8}{4\pi 2^2} + \frac{4}{359}\right] = G - 11.7\text{dBu}$$

For the Right microphone, the effective $DI$ is – in similarity with a) – $4.8 - 11.7 = -6.9$dB resulting in $DF_{\text{MIC}} \approx 0.2$ and the output level becomes

$$L_{\text{MIC,R}} = G + 10\log\left[\frac{0.2}{4\pi 2^2} + \frac{4}{359}\right] = G - 18.2\text{dBu}$$

The level difference between the channels is then $\Delta L_{\text{u}} = 18.2 - 11.7 = 6.5$dB, which is clearly smaller than for the free field recording.