Columbia University **EE E4896 Music Signal Processing**
Department of Electrical Engineering **Spring 2005**

Homework 2
– Due April 4, 2005 –

Problem 1

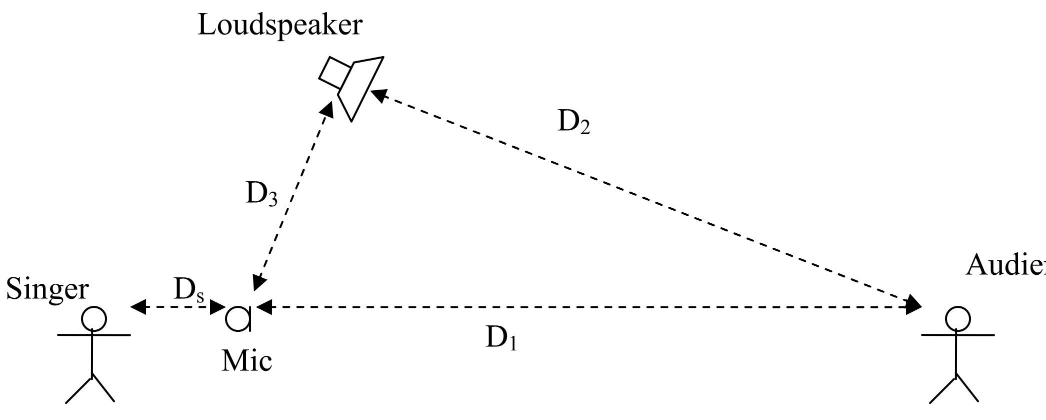
The significant frequency range of a signal extends to f_{max} (Hz). Beyond f_{max} the spectrum attenuates by α dB/octave. We have available an off-the-shelf antialiasing prefilter that has a flat passband to f_{max} and attenuates by β dB/octave beyond that. It is required that within the f_{max} range of interest, the aliased components due to sampling are suppressed by A db. Show that the minimum sampling rate that we should use is given by:

$$f_s = f_{max} + 2^{A/\gamma} f_{max}$$

where $\gamma = \alpha + \beta$.

Problem 2

Here we will explore the issues that arise in the design of basic sound reinforcement systems. The prototypical model is shown in the figure below, where a singer is performing in front of a mic, which is connected to a loudspeaker in order to reach the audience site with sufficiently high SPL.



Assume that the singer is generating 70 db SPL at 1m, and that D_1 = 7 m, D_2 = 6 m, and D_3 = 4 m. Compute the dB SPL of the singer's voice at the audience site. Obviously, we would like the SPL due to the loudspeaker to be significantly higher than that, giving the impression to the audience that it is situated much closer to the performer. What is limiting us (beyond the power limitation of the loudspeaker) is that some of the amplified sound coming out of the loudspeaker will be picked up by the microphone, creating a feedback loop. When the dB SPL (as picked up by the mic) coming from the singer equals the dB SPL coming from the loudspeaker, the loop will have unity gain and can thus become unstable, driving the output of the loudspeaker to its maximum (a familiar phenomenon during installation of sound systems). The difference of dB SPL at the audience with the loudspeaker on and off is called "acoustical gain" of the system.

Assume that both the mic and loudspeaker have omnidirectional patterns (both horizontally and vertically). Derive a general expression for the maximum acoustical gain at the audience site that avoids a positive feedback loop. Assuming that the loudspeaker and audience positions are fixed, how would you move the singer and microphone to further increase the gain? Plug-in the numbers for the different parameters as given above, and compute the maximum SPL at the audience. (Note that in practice you would want to leave at least a 6 dB margin; during performance levels are typically higher than expected.)

Now assume that the mic is the Neumann KM184, that the speaker is the Mackie SRM 450, and that the singer is female. Recompute the maximum SPL and acoustical gain at the audience site so that feedback is avoided. Use a 6 dB margin as well. (Specifications are available at www.mackie.com and www.neumannusa.com.)

Further assume that you have a handheld 1/3 octave band spectrum analyzer at your disposal and a 2-channel 31-band graphic equalizer that can be inserted in the mic-to-loudspeaker path. Explain how you could further increase the gain of the system while still avoiding feedback. (Note: after thinking through this you may want to see how the dbx DriveRack PA works).

Problem 3

Here you will use MATLAB to recreate two demonstrations given in the “Total Recording” CD[1]. Be very careful when auditioning WAV files that you create, especially with headphones, as volume levels could get extremely high if you make errors and can damage your ears.

a) Using MATLAB, create a mono 1 kHz sine wave tone, at an RMS level of -20 dB FS. A few seconds should be sufficient. Now create a second tone at a slightly lower or higher frequency (e.g., 1,002 or 998 Hz), and create a stero file placing the first signal in the left channel and the second in the right. Listen to the result. Now mix the two together into a single mono WAV file (with equal weights). Plot the result in MATLAB and listen to the result.

b) Create a noise waveform (mono) with an RMS level of -20 dB FS (use random numbers drawn from a uniform distribution, appropriately scaled). Now create a stereo version of it in which the right channel has a copy of the left one but attenuated. Produce a single stereo file where the attenuation proceeds in steps of 1 dB down to 10 dB, spaced 5 sec appart. Listen to the result. Plot on a graph your perception of the position of the phantom image with respect to the difference in dB between the two channels[2]. (Note: at this point you may also want to try zero attenuation but phase inversion for the right channel and listen to the result. Where do you hear the phantom image now?)

c) Repeat (b), but here copy the left channel to the right one with a delay that proceeds in steps of 5 ms up 100 ms. Listen to the result. Plot on a graph your perception of the position of the phantom image with respect to the difference in time between the two channels. Compare with (c).

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[1] You will need to submit MATLAB code as well as the relevant result (.wav) files. You can either email ZIP files to the TA or send a URL to your web site, if you have one, so that we retrieve them.

[2] Draw a line segment between the two speakers, and put an ‘x’ where you think the phantom image is located. Write the attenuation value used for that position.