ELEN 4810 Homework 2

Due **Wednesday, November 11**. Please submit your responses via CourseWorks. Please combine your responses to the analytical and computational problems in a single pdf file.

For the analytical section of the assignment, submit a single PDF of name `yourunihere hw02.pdf` - e.g., "`abc1234 hw02.pdf`". For the coding problems, please put your code in a single script file `yourunihere hw02.m` - e.g., "`abc1234 hw02.m`". Also, use the Publish option on MATLAB to create a PDF that will publish your script as well as its figure outputs. Please submit this as `yourunihere hw02_m.pdf` - e.g., "`abc1234 hw02_m.pdf`"

Thanks!

Analytical Questions

Please complete problems 2.34, 2.62, 2.74 in Oppenheim and Schafer (3rd Edition). Justify your answers!

Computational Questions

1 Introduction

For the computational assignment this week we will apply some (discrete) time-domain signal processing techniques on real audio data. We will implement both linear and non-linear systems as MATLAB functions.

For this purpose, you are provided the `.wav` file `excerpt.wav` for experimentation.

Preparation. Earphones are recommended for the assignment. The following commands will be useful:

- `[x, fs] = audioread('excerpt.wav')` grabs the audio signal `x` and sampling frequency `fs` (in Hz) from `excerpt.wav`.
- `sound(x,fs)` and `clear sound` starts and stops playback of the excerpt respectively.

Since `x` is a *stereo* signal, it contains two column vectors representing sequences from the left and right channels. Listen to the audio, which column represents the left channel? The right channel?

We will assume that the continuous time domain is expressed in units of **seconds**. Recall that for an arbitrary continuous time signal $a(t)$ and a sampling period T, its discrete-time representation is given by

$$a[n] = a(nT), \quad \forall n \in \mathbb{Z} \ . \tag{1}$$

- What is the sampling period T corresponding to the discrete-time signal `x`?
- What is the sample length of `x`? The duration of `excerpt.wav`? Are the answers consistent?

2 Problems

Note: Each problem involves writing a function requiring (far) less than 30 lines of code. Furthermore, no submission is required for the observation questions.

2.1 Channel Delay

In a file named `audio_delay.m` write the function

`[ y ] = audio_delay(x, fs, mtpair).`

This function takes an audio sequence and delays its left and right channel signals by a specified duration. Here `x` and `fs` is as described above; the argument `mtpair = [lmt rmt]` is an array with `lmt` and `rmt` as the left and right delays in *milliseconds*.

The resulting output `y` should be a single 2 column array, representing a stereo signal where the left and right channel signals are delayed by `lmt` and `rmt` ms respectively. Letting $\vec{x}(t) = (x_L(t), x_R(t))$ represent the left-right input signal pair in continuous-time domain. The desired output $\vec{y}(t) = (y_L(t), y_R(t))$ is expressed as

$$\begin{aligned} y_L(t) &= D_{\mathtt{lmt}\cdot 10^3}(x_L)(t) \quad \text{for the left channel,} \\ y_R(t) &= D_{\mathtt{rmt}\cdot 10^3}(x_R)(t) \quad \text{for the right channel.} \end{aligned} \tag{2}$$

For implementation we seek a discrete-time representation via (1), i.e. for $\vec{x}[n] = (x_L[n], x_L[n])$ we want

$$\begin{aligned} y_L[n] &= D_{g(\mathtt{lmt})}(x_L)[n] \quad \text{for the left channel,} \\ y_R[n] &= D_{g(\mathtt{rmt})}(x_R)[n] \quad \text{for the right channel.} \end{aligned} \tag{3}$$

So that applying `sound(y,fs)` to $\vec{y}[n]$ plays the stereo output corresponding to (2). It is left for you to figure out the correct conversion g for converting ms-time to sample delay. You should also determine the correct length of y.

Note: Please use the `floor()` function to assign the sample delay to the nearest smaller integer.

Observations. Process `x` using this function with channel delays `1000` and `500` ms to the left and right channels respectively. What do you hear compared to the original music sequence?

2.2 Panning

In audio processing, we often use systems that are not time-invariant. Recall that in Homework 1 we have seen the system $\mathcal{T}(x)[n] = \sin(\pi n)\, x[n]$ – we will apply a similar function here. Write the following function in file named `audio_pan.m`,

`[ y ] = audio_pan(x, fs, f).`

The function takes as an argument the *panning frequency* `f`. The function should output a sequence pair $\vec{y}[n] = (y_L[n], y_R[n])$ so that `sound(y,fs)` plays a stereo signal corresponding to

$$y_L(t) = x_L(t) \cdot |\sin(2\pi f t)| \qquad y_R(t) = x_R(t) \cdot |\cos(2\pi f t)| \ . \tag{4}$$

Again you have to adopt a discrete-time representation of the system (4) by considering (1). As this is not an LTI system, you cannot use the convolution operator to implement this function.

Observations. Try to assign `f = 5`, what does the output sound like? What happens if you set `f = 50`?

2.3 Low Frequency Oscillation

The last question is a little tricky. We will apply a low frequency oscillation (LFO) to the signal, which is effectively a 'delay operator with oscillating delay time'.

Please write the following function in a file named `audio_LFO.m`,

`[ y ] = audio_LFO(x, fs, f, mt),`

where `f` is the *oscillation frequency* and `mt` is the *maximum delay time* in ms. The function should output an array `y` of length `length(x)` so that `sound(y,fs)` plays a stereo signal corresponding to

$$\vec{y}(t) = \begin{cases} \vec{x}\left(t - \left(\mathtt{mt} \cdot 10^{-3}\right) |\sin(2\pi \mathtt{f} t)|\right) & t > \mathtt{mt} \cdot 10^{-3} \\ \vec{x}(t) & t \leq \mathtt{mt} \cdot 10^{-3} \end{cases} . \tag{5}$$

The same delay operator is applied to both channels. Use (1) to derive $\vec{y}[n]$ in the form

$$\vec{y}[n] = \begin{cases} \vec{x}\left[\, n - \lfloor p(\mathtt{mt}) \mid \sin(2\pi \cdot \mathtt{f} \cdot q(n)) \mid \rfloor \,\right] & n > p(\mathtt{mt}) \\ \vec{x}[n] & n \leq p(\mathtt{mt}) \end{cases} . \tag{6}$$

Is this system time invariant? Linear?

Again it is left for you to determine the correct conversions: p to go from ms-time to samples and q to go from samples to time. Note that where the sample delay is concerned, the quantities $p(\mathtt{mt}) \cdot |\sin(2\pi \cdot \mathtt{f} \cdot q(t))|$ and $p(\mathtt{mt})$ need to be integers. Use the `floor()` function to accomplish this.

Observations. Try `f = 1` and `mt = 5`, what do you hear? Try any input you wish and compare the results.

3 Remarks and Submission

Now that we have implemented some systems for processing audio signals, try applying these functions to audio signals of your choice: just remember to use stereo signals!

You may also want to try writing the output back into `.wav` or some other audio format; take a look at the function `audiowrite()`. Notice that the input arguments (in addition to the filename) is simply the sequence pair and the sampling frequency.

Place the completed functions `audio_delay.m`, `audio_pan.m` and `audio_LFO.m` into a single `.zip` file with your UNI and homework number as the filename, i.e. in the form of `'ma3810_hw2.zip'`. *Do **not** upload any `.wav` files!* Please upload and submit to Courseworks by **1:10pm on Wednesday, November 11th**.