2020 ASAS Homework 2: Convolution and Linear Time-invariant Filtering

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1. **Averaging and "differencing"** (Remarks: Yes, here we are going to use "difference" as a verb). Please first find a piece of music or any sound and read it into MATLAB. Denote the signal as $x[n]$.

(a) Check the sampling rate, the number of channels, and the number of bits per sample.

(b) The simplest low-pass filter is to average over consecutive samples; that is, let

$$y[n] = \frac{1}{p}\sum_{m=1}^{p} x[n-m+1],$$

where p is the order of the filter, and $y[n]$ is the output of the filter. Implement this filter and listen to the output. Increase p from 1 to 10 to see if you can hear the difference.

(c) Check your result in (b) against the following results. They should essentially be identical.

```
y_conv = conv(x, 1/p*ones(p,1));
```

(d) In fact, the array 1/p*ones(p,1) can be regarded as the impulse response of this **FIR** (finite impulse-response) filter. Denote h = `1/p*ones(p,1)` and use `freqz(h)` to plot the frequency response of the filter.

(e) Now, define $y_d = x[n] - x[n-1]$. Of course this can be done with a "for" loop, but alternatively, please use `conv()` to do the job.

(f) Set `x` = 0.1*randn(A_CERTAIN_LENGTH, 1) so it is an instance of Gaussian white noise with mean zero and standard deviation $\sigma = 0.1$. Listen and compare the result before and after averaging/differencing. Does it feel more unpleasant before or after averaging/differencing? Describe how you feel about it and explain, perhaps, the reason why.

Remark: When you listening to a signal stored in a vector, make sure that its range is between ± 1 to avoid clipping effect. Or, alternatively, use `soundsc()` to avoid clipping. Always remember to specify the sampling rate.

2. **Infinite impulse response (IIR)**. Take any audio signal $x[n]$ and for any $n > 1$ implement the following by a for loop,

$$y[n] = \alpha y[n-1] + x[n],$$

Where $0 < |\alpha| < 1$ is a constant. You can assume $y[1] = 0$.

(a) Check your result against `y = filter([1], [1 -alpha], x)`.

(b) Create an instance of Gaussian white noise for about 1 second long. Listen and compare the result before and after filtering.

(c) Create a periodic impulse train:

$$x[n] = \begin{cases} 0.5, & \text{if } n = 80m \\ 0, & \text{otherwise.} \end{cases}$$

Where m is an integer. In other words, $x[n]$ is non-zero at every 80th sample. Listen and compare the result before and after filtering.

(d) Discuss how to restore $x[n]$ from $y[n]$. (Hint: if α is known, it is quite easy. However, if α is unknown, you may need to make assumptions.)

Preview of the next homework: We will study how to model speech as a glottal source signal being filtered by the vocal tract next time.