MUS424:
Signal Processing Techniques for Digital Audio Effects `Handout #13`
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Homework #5: Minimum and Linear Phase, Critical Band Smoothing, and Warped Prony [30 points]
Due Date: May 16, 2019

Minimum Phase, Linear Phase, Critical Band Smoothing, Warped Filter design

Submission Instructions

Submit via Canvas. Create **a single compressed file** (`.zip`, `.tar` or `.tar.gz`) containing all your submitted files. Upload the compressed file to the homework submission dropbox. Name the file using the following convention:

`<suid>_hw<number>.zip`

where `<suid>` is your Stanford username and `<number>` is the homework number. For example, for Homework #1 my own submission would be named `cavdir_hw1.zip`.

For coding problems (either C++ or Matlab code), submit all the files necessary to compile/run your code, including instructions on how to do it. In case of theory problems, submit the solutions in **PDF format only**. LaTeX or other equation editors are preferred, but scans are also accepted. In case of scanned handwriting, make sure the scan is legible. Illegible homework will not be graded.

The files `Jen57Off.wav` and `Jen57On.wav` contains impulse responses of a Jensen guitar cabinet `ir`, measured with a Shure SM57 microphone placed off-xis and on-axis. The impulse responses are sampled at 44.1 kHz, `fs`. The files directory also contains a roughly 5-second-long distorted guitar snippet also sampled at 44.1 kHz. You will use these throughout this homework assignment.

Problem 1. [16 Points] Spectral Delay Filter

The spectral delay filter (SDF) can be implemented as a cascade of low-order allpass filters.

1(a). [6 Points] Theory The transfer function of the first order allpass filter is

$$A(z) = \frac{\rho + z^{-1}}{1 + \rho z^{-1}}$$

Derive the group delay of the allpass filter.

$$\gamma_\rho(\omega) = -\frac{d}{d\omega}\measuredangle A(e^{j\omega})$$

Turn in your derivation steps.

$$Hint: \log(A(e^{j\omega})) = \log(\mid A(e^{j\omega}) \mid) + j\measuredangle A(e^{j\omega})$$

1(b). [**2 Points**] Plot the group delay for ρ values of [-0.75, -0.2, 0.5] in the frequency range of [0, fs/2].

1(c). [**4 Points**] Implemented the spectral delay filter as N cascaded first-order allpass filters as shown in Figure 1(c).. Form the impulse response for N = 1000 taps, fs = 8000 Hz, and ρ values above for $G(\omega) = A(e^{j\omega})^N$.

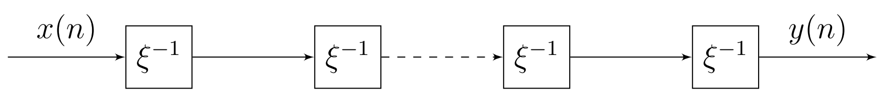

Figure 1: Spectral Delay Filter

Turn in the impulse response associated with the cascade. Plot spectrograms for each of the ρ values above.

1(d). [**4 Points**] Plot spectrograms of the impulse responses of the system shown in Figure 2 for each of the three ρ values from part b.

Hint: It can either be implemented in the time domain with sample by sample processing, or in the frequency domain by evaluating the transfer function and applying the inverse DFT.

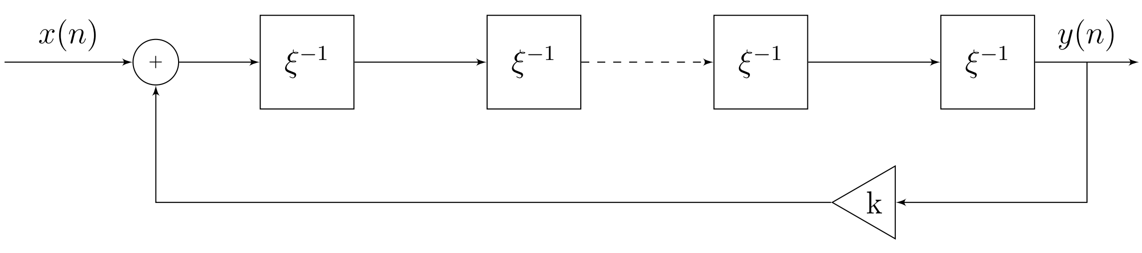

Figure 2: Feedback spectral delay filter

Problem 2. [14 Points] Prony's Method

In this problem, you will fit an IIR filter to the FIR filter manipulated in the previous problems. Use a windowed, minimum-phase impulse response, based on a 2.0 ERB-smoothing of the measured impulse response.

2(a). **[2 Points]** Use Prony's method to fit 6th-order and 12th-order IIR filters to the smoothed minimum-phase impulse response. Using a logarithmic frequency axis, plot the dB transfer function magnitude associated with the measured impulse response, 6th-order model, and 12th-order model on the same axis. Turn in your code and plots, and note any discrepancies between the measured and modeled transfer function magnitudes. Do either of the models capture the important psychoacoustic features of the filter?

2(b). **[2 Points]** Now, follow the same procedure but perform a frequency pre-warping before modeling the impulse using Prony's method.

For the pre-warping, use the allpass transformation

$$\zeta^{-1}(z) = \frac{\rho + z^{-1}}{1 + \rho z^{-1}} \tag{1}$$

defined by the warping parameter $\rho \in (-1, 1)$. You will have to decide the ρ which gives large bandwidth to important spectral features in order to improve Prony's method fit. Plot the dB magnitude spectra for both the original and pre-warped filter on the same plot. Also plot the associated impulse responses for both the original and pre-warped filter on another plot with an offset. Turn in your code and plots.

2(c). **[2 Points]** Use the MATLAB function `prony` to form the 6th-order and 12th-order models of the pre-warped impulse responses. Plot the pre-warped and modeled pre-warped transfer functions on the same plot. Also plot the associated impulse responses on another plot with an offset. Compare the 6th-order and 12th-order warped models with their pre-warped counterparts above.

2(d). **[4 Points]** Now perform a post-warping to undo the pre-warping on the poles and zeros of the modeled IIR filter. MATLAB function `tf2zp` can be helpful. For the post-warping use the same transformation in eq. 1 with $-\rho$. Plot the dB magnitude spectra for both the original and post-warped modeled filter on the same plot. Also plot the associated impulse responses for both the original and post-warped modeled on the same plot with an offset.

2(e). **[2 Points]** Using `zplane`, plot the modelled and post-warped pole-zero diagrams. Comment on the results.

2(f). **[2 Points]** Convolve the provided guitar clip with the given FIR filter, Prony's method-modeled IIR filter, and warped Prony's method-modeled IIR filter. Listen to the results and comment on how successful the various techniques were for modeling the given FIR filter. Turn in your code and audio files along with your discussion on these techniques.

2(g). **[0 points; for fun only, but tell us if you do it...]** Express your warped Prony's method IIR filter as a cascade of biquads, and incorporate the guitar cabinet model to your distortion project from last week. Add controls for a user to mix the amount of filtered/unfiltered signal appears at the output.