Digital Signal Filtering Audio

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I. Introduction

In this computer assignment we were given a beep tone, two audio clips, and their corresponding time vectors. Using known digital filters such as the FIR and IIR filters we made various sound effects. These sound effects consisted of echoing the original signal by delaying and rescaling. By changing the values of the coefficients of the transfer function, these sound effects were created. After adding the echo with each filter, we created a surround sound stereo output which plays the filtered signal on the right speaker and the original signal on the left.

II. Methods

The method to complete this computer assignment started with plotting the beep with the corresponding time vector and playing the sound. Using the same commands of playing and plotting the original beep signal we applied them to the FIR and IIR filtered beep signals. They are identical until the values of "a" and "d" are changed which are used for the coefficients of the transfer function of each filter. After changing the values, we looked and the plot and sound to compare to the original. When changing "d" it delayed the filtered signal by d seconds creating an echo. When changing "a" it rescaled the original signal's amplitude by a which created the second tone to be less loud. The difference between the filters is that for the IIR filter it uses a copy of the output of the input signal rather than the input. We then examined when "a" was greater than one and this created a lower sounding signal which got louder after d seconds. Using these techniques, we applied them to the two audio clips which created an echo. We also applied the original signal in the left speaker and the filtered signal in the right speaker to give a surround sound type of effect.

III. Results

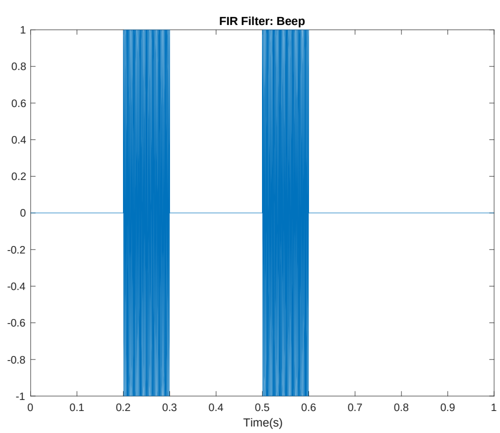

Figure 1: FIR Filter - Delayed by .3s - d=.3, a=1

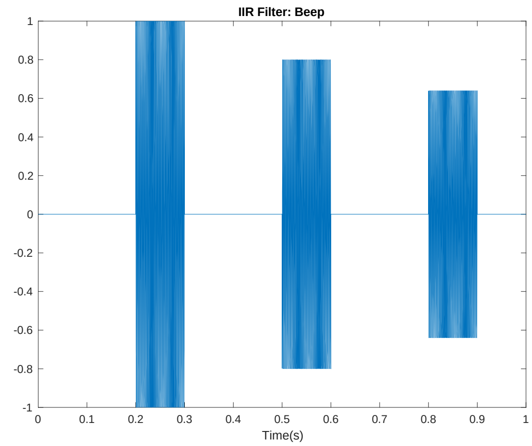

Figure 2: IIR Filter - Rescaled by .8 - d=.3,a=.8

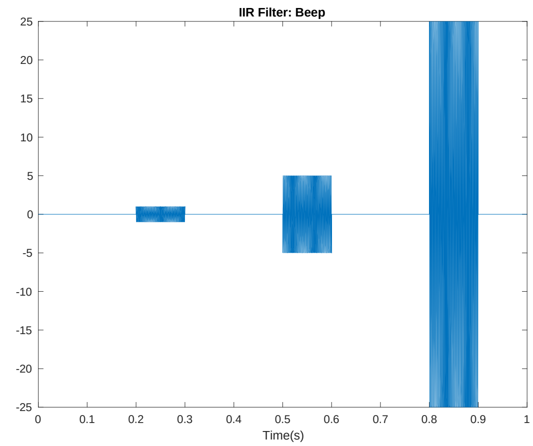

Figure 3: IIR Filter - a > 1 - d = .3

We then used the same technique on the audio clips in order to create our sound effects. When changing the values for the audio clips the sound was laid over the original signal, but if "d" was not chosen correctly it would sound like a mess and the closer it was to the value one sounded best. When changing the sound to stereo the value sounded best closest to the value zero. They responded oppositely.

IV. Discussion

This computer assignment helped understand digital filters and their coefficient effects with audio. In other applications, needing to delay of scale a signal may be important to complete a task and a simple change of a coefficient may do the trick. We also attempted to play audio1 from the left speaker and the delayed audio2 from the right speaker which worked because of their equal sampling rate. There is also a creation of an mp4 file of the stereo audio by using more MATLAB tools.