

DESC 9115 2012 LAB REPORT 1 - DELAY

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Introduction

In this lab report the chosen topic researched is Delay. It is chosen due to the close correlation with the first initial technology review being reverberation. This code was not implemented from a book or from an online source but rather from an engineer who guided me through in a way to understand the basic function of a delay utilizing Matlab code. The code that has been implemented and can easily be tweaked in such a way that combs filtering can also be generated.

Process

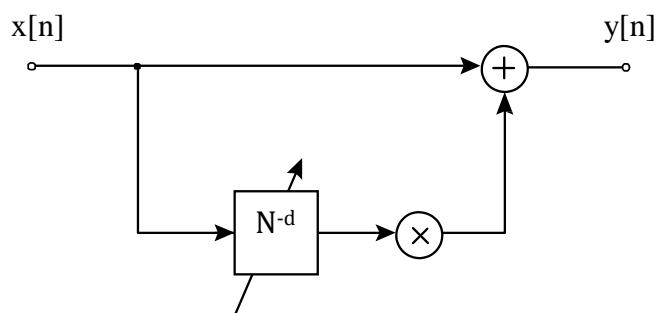
An arrangement that simulates one single delay is called a comb filter. This input signal will be delayed by a given time delay. The spectral shape can be acquired by summing an input signal which is $y[n]$ with the delayed signal or the same signal which is $x[n]$. The formula is as follows;

$$y[n] = x[n] + g * x[n - d]$$

g is the factor of gain for the filter which in this case coded as + 1.

In order to properly use this function, we need to understand how to use it by processing a digital audio file. First we need to know what desired fundamental frequency or signal that we are to use by calling the desired wav file and in this case it's an impulse response, which was given in a tutorial by Luis. This wav file is then fed back into Matlab, which it now has been given a value of "y" which is an array of the wav data.

The next step is to calculate or input the amount of delay time to be given to the wav file in value of samples per second and for this example it has been given the value of 0.01 second. I have also given it another parameter, which is gain. This is added to the second wav file with a gain of 0.1 dB.



The next step is to create an array to combine the delayed waveform and combined waveform. First the wave file must be called, this is defined in WavFN = (.wav). This is then collapsed into an array with defined values for sample rate and number of bits. We then define the delay time for the wav file, in this case 0.02, next is the delay time sample, which is the delay time multiply the frequency sample and gain will also be added.

Next process is to create an array that will hold both the combined and delayed waveform with delay time sample. The second wave file, y2 will now be filled with delayed time samples along with the amount of added gain. At this point, it will combine both the delayed waveform with the original waveform and this will in turn process all the waveform into one singular output wave. I have also included in the script to plot 3 graphs with each with its title, Original, Delayed and Combine, which is on the following page.

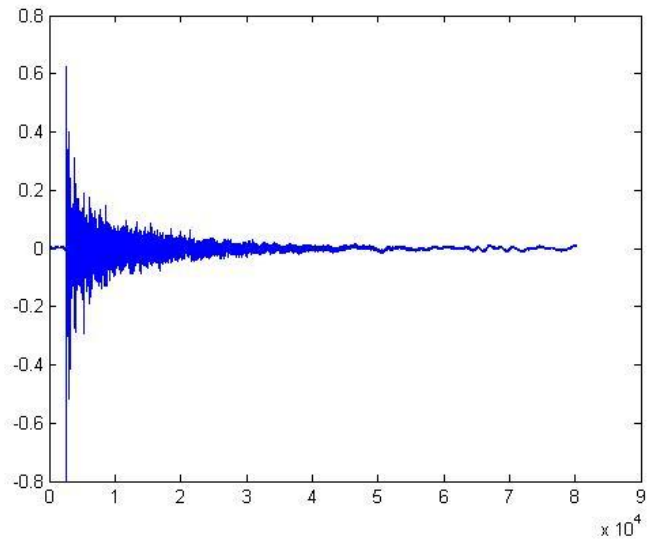
Conclusion

By applying the equation within Matlab, we are able to alter the original sound wave so that we may produce an output fitted to the specifications outlined by the user. In doing so, we're able to construct an audio file that either duplicate or manipulate the sound entirely. This application has relevance to sound engineers and designers wanting to manipulate existing sound without the cost of creating new audio rather from scratch.

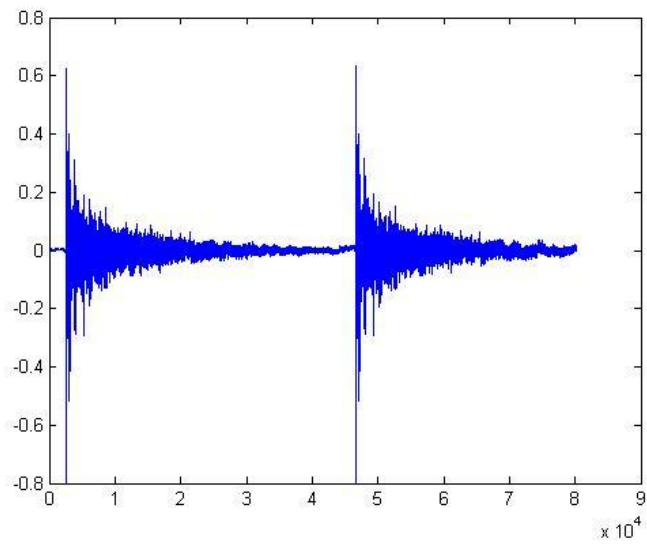
Function of Script

```
outwave = Delay2('ImpulseResp_A_Mono.wav',1,.5)
```

Original Impulse Response



Combined Impulse Response



Delayed Impulse Response

