ECE280 - Lab 3: Digital Audio Effects

Jerry Worthy (jdw102)

March 4, 2022

I have adhered to the Duke Community Standard in completing this assignment.

Contents

1 Introduction

The objective of this lab is to become more comfortable using simulink to simulate different signals and systems. This lab will allow students to design and use simple audio systems as well as manipulate those signals using different audio system toolbox resources.After the lab, the students will draw meaningful conclusions about how the behavior of the signal is related to the design of their systems and the different properties of these systems.

2 Background

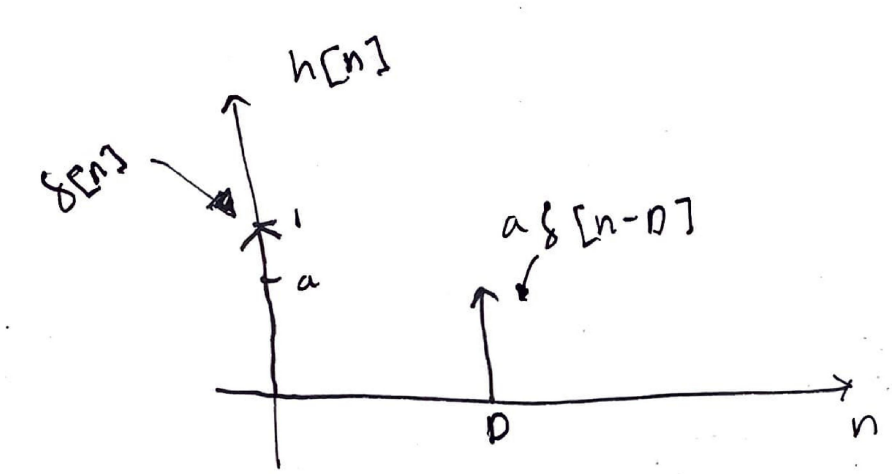
- **Single Echo**: The single echo uses a delay and a gain to repeat back the original sound wave after a certain amount of time at a fraction of the original magnitude. This is done by branching off the original signal, delaying and scaling it, and then adding it back to the original.
- **Reverberation**: The reverberation filter works similarly to the single echo, but instead of modulating the input and adding to itself once, it actually modulates the output of the adder and adds it back to the original signal. This results in an infinite number of echos with decaying, or growing, volume over time. It uses a delay and gain to achieve this effect and loops through the adder recursively.
- **Flanging**: Flanging works similarly to the single echo effect but with a varying delay in the echo. This delay varies according to a sinusoidal function and the modulated signal is mixed back with the original signal to create the effect. It does this using a sine signal producer and a variable fractional delay to modulate the branched original signal.

3 Results and Discussion

1. Make observations about what you heard in each case. For the first exercise, there was only one, normal, echo as the output was simply the same as the audio input, thus everything sounded normal. The other observations are made in the subsections below.
2. Describe what changing the coefficients in the difference equations did to the observed sound. Overall, changing the coefficients followed roughly the same pattern, with the gain, a, changing the extent to which the sound was modulated, and the delay, D, changing the time between echos. However, these topics are expanded upon further in the subsections below for each specific exercise, including flanging.
3. **Single Echo Effect** Results and Discussion.
    - Describe how the SimuLink model is related to the difference equation. The difference equation is described as a sum of the original signal and a delayed signal with a scaled magnitude. This is done by branching the signal into a delay and a gain and then re adding it to the original signal using an adder before leading it the output. In this case, the offset of D is done with the delay and the scaling of a is done with the gain.
    - What do you hear when a signal is processed using the single echo system? The single echo system repeats the input audio back out of the speaker with a scaled volume. In Simulink this leads to the original signal being repeated back with no delay at first and then the delayed signal.
    - What effect does changing the parameter, a, have on the output signal? This changes the volume of the delayed output signal.
    - What effect does changing the parameter, D, have on the output signal? This changes how long the signal is delayed.
    - Why does the output sound this way? After giving the input by speaking into the microphone, the signal is split into two branches. The first branch goes directly to the output, which is why an initial echo is first heard. The second branch is delayed and then passes through a gain which changes its volume, which is why the second echo comes after and has a different volume.

- Provide an analytic expression and a sketch for the impulse response, $h[n]$, of the single echo system. $h[n] = \delta[n] \ + a\delta[n - D]$

Figure 1: Single Echo Impulse Response Sketch



- How can you interpret $h[n]$ in terms of your observations? A single impulse will produce the impulse itself and a delayed scaled impulse.
- Assuming a sampling rate of $8000Hz$, how would you choose the delay, D, to achieve a time delay of 0.25 seconds? A delay of 0.25 seconds would be a quarter of the sampling rate. Since the sampling rate is 8000 Hz, this means we need a delay of 2000 frames.
- Is the single echo effect system memoryless? No, it has a delay, which means it relies on the past.
- Is the single echo effect system casual? Yes, it does not rely on future values of the signal.
- Is the single echo effect system stable? Yes, for every bounded input, the output will also be bounded since it solely depends on the transformed input.
- Is the single echo effect system time invariant? Yes, a shift in time in the input corresponds to a shift in time in the output.
- Is the single echo effect system linear? Yes, linear combinations of input signals correspond to linear combinations of the corresponding output signals.

4. **Reverberation** Results and Discussion.

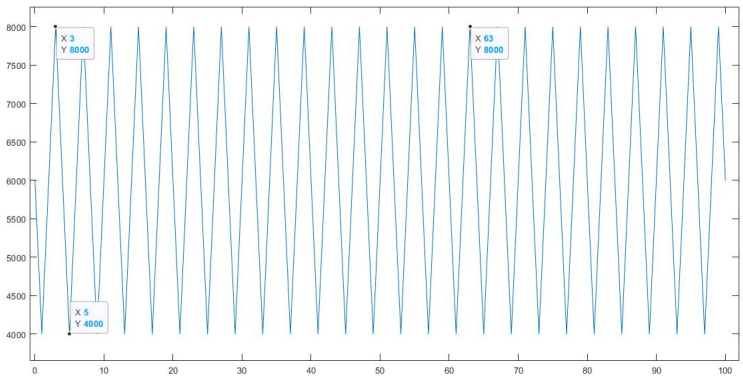
- Describe how the SimuLink model is related to the difference equation. Instead of branching before the adder, this model branches after, to mimic the recursive behavior of the difference equation. The adder adds its own output to itself, after first going through a delay and a gain, to mimic the a and D parameters in the difference equation.
- What do you hear when a signal is processed using the simple reverberation system? There is an infinite number of echos of the original noise that continue to decrease in volume until silence.
- What effect does changing the parameter a have? Changing a changes the rate at which the echo volume increases or decrease. If it is less than 1 it will decay and if it is greater than 1 it will increase over time.
- What effect does changing the parameter D have? Changing D changes the time between each echo, it increases or decreases the delay.

- Compare the simple reverberation filter with the single echo filter with the same delay and gain. How do they differ? The reverberation filter continuously produces an infinite number of echos for the audio input that each decay over time until almost silent, while the single echo filter only repeats the audio input once before cutting it off. This means the reverberation filter in a sense stacks more signals on top of each other, which makes it sound more interesting but also more jumbled, while the single echo is quick and clean with only one echo.
- Compare the simple and allpass reverberators. How are they similar? How are they different? Why do you think the allpass reverberator sounds more natural? Both have the recurring echo of the audio signal with the decaying (or growing) volume, however the allpass reverberator seems to keep the individual echos more clear and separate from each other with a faster decay of volume, which makes it easier to comprehend the noise. The allpass reverbator may sound more natural because it mimics better the natural diffusion of sound waves produced in an open area, as there is a more sharp drop off of volume of each echo. It seems to do this by also delaying the signal with a gain of -1, which would cancel out the echo after a certain number of loops.

5. **Flanging**:

- What do you hear when a signal is processed using the flanging system? The flanging sytem seems to produce a sort of delayed echo with a changing delay time. For example, it will repeat the audio signal back very quickly at varying rates, to create a sort of subtle warble sound effect depending on the frequency at which the signal is delayed. The
- What effect does changing the gain, delay, and flange frequency have? Changing the gain changes how obvious the flange effect is by changing its volume, the higher the gain, the more distorted the original sound by the effect. Changing the delay changes the speed at which the overall modulated audio is outputted just as it did with the echo effect. Changing the flange frequency changes how much the modulated delay is changing. Higher frequencies produce more warble-like and distorted sounds while lower frequencies produce more subtle distortions. This is because rapidly changing the delay of the sound wave causes more fluctuation in the form of the combined sound wave.
- Plot the low-frequency oscillator, $d(n)$, for these parameters: $D = 4000$ samples, $f_s = 8000$ Hz, $f_d = 2000$ Hz.

Figure 2: Low-Frequency Oscillator



- What is the maximum delay when $D = 4000$ samples and $f_s = 8000$? What is the minimum delay? The maximum delay is 8000 and the minimum delay is 4000.

4 Conclusions

This lab was very effective in demonstrating how equations, such as the difference equation, can be translated into a simulated model using different functions and blocks. There was a lot of analytical value in using audio

signals to demonstrate this, as students could qualitatively observe how the signal was being transformed by the system by simply listening to the result. There was a lot of overlap as between what the lab asked of the students and what was taught in lecture. Such as the importance of visualizing transformed systems through block diagrams as well as analyzing the resulting systems' properties; determining whether they were memory-less, causal, linear, time-invariant, and stable.