# ECE280 - Lab 3: Digital Audio Effects

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

# Contents

# 1 Introduction

In the experiment/ lab exercise, we explored simple digital audio effects, specifically reverb and echo. We implemented algorithmic functions to implement these using the audio system toolbox of Simulink. We implemented speech in real time as well as recorded speech. Lastly, we explored the effect of flanging on sound. Overall, the goals reached included, implementing simple input /output using Simulink and manipulating system parameters to explore these outputs on audio outputs

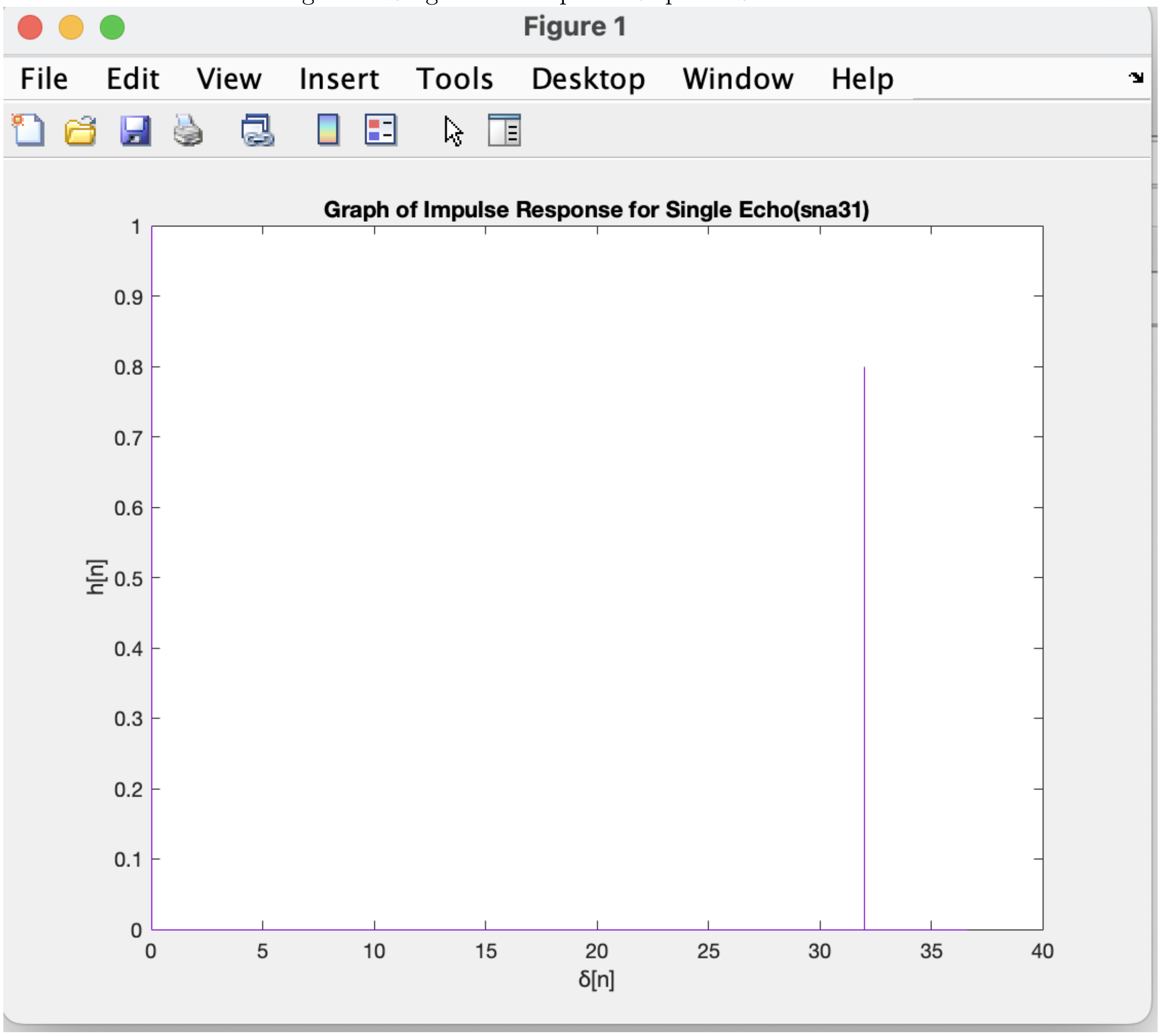
# 2 Background

- **Single Echo**: Echoes are delayed attenuations of an original sound that can be interpreted as reflections of sound intertwined with the original sound itself.
- **Reverberation**: A reverberation is the culmination of echos as the number of echos approaches infinity. t occurs when sound waves reflect off surfaces like walls, floors, and ceilings, creating a sense of depth and space
- **Flanging**: Flanging is an audio effect that creates a swirling, sweeping sound by mixing two identical audio signals together, with one of the signals delayed by a small, variable amount. This delay is modulated over time, creating a series of peaks and troughs in the frequency spectrum. .

# 3 Results and Discussion

1. Make observations about what you heard in each case.
2. Describe what changing the coefficients in the difference equations did to the observed sound.
3. **Single Echo Effect** Results and Discussion.
    - Describe how the SimuLink model is related to the difference equation. The Simulink model represents the equation by creating a model that sums an original input and an amplified form of the delayed original signal. Thus the model is a physical interpretation of the difference equation
    - What do you hear when a signal is processed using the single echo system? When the model is implemented using a single echo, there is a slight delay and the sound is repeated again. Thus the sound is repeated for a single running of the program before the original sound "dies" out . Thus one hears the original sound and an echo of itself delayed by D samples
    - What effect does changing the parameter, $a$, have on the output signal? : The echo becomes louder and more pronounced. If a is too high; it can dominate the original sound for a significantly huge amplitude, making the original sound hard to discern. For a decreased a, the echo gets quieter and quieter and blends even more with the original sound almost seamlessly. For a =0, there is no echo
    - What effect does changing the parameter, $D$, have on the output signal? Changing D manipulates how soon the echo is heard. For a smaller Delay, the echo arrives sooner after the original sound and creates a more cohesive sound that may sound like a phase shift than a distinct echo. For a large value of D, the echo arrives much later, creating a larger separation between the original sound and its echo but this could also make the sound less cohesive and make the echo prone to misinterpretation as a different original sound
    - Why does the output sound this way? The combination of the original sound and its amplified delayed version creates the echo effect. The model produces what is heard by summing an original sound with an amplified form of its delayed version. The amplification is implemented using a gain whilst a delay is implemented using a time/sampling delay.
    - Provide an analytic expression and a sketch for the impulse response, $h[n]$, of the single echo system. ' $y[n]= delta[n] + adelta[n-D]$'

Figure 1: Single Echo Impulse Response Sketch




- How can you interpret $h[n]$ in terms of your observations? The impulse response h[n] scales the impulse and delays this signal by D. The impulse response clearly has an amplitude of 8 and has been delayed to exactly 32 seconds after input of impulse at t =0
- Assuming a sampling rate of $8000Hz$, how would you choose the delay, $D$, to achieve a time delay of 0.25 seconds? 2000 such that $2000/8000 = 0.25$ .
- Is the single echo effect system memoryless? The system has memory
- Is the single echo effect system casual? The system is causal
- Is the single echo effect system stable? The system is stable
- Is the single echo effect system time invariant? The system is time invariant
- Is the single echo effect system linear? The system is linear

4. **Reverberation** Results and Discussion.

   - Describe how the SimuLink model is related to the difference equation. The Simulink is a physical representation of the difference equation . Thus it feeds back the input into the system by first delaying the original output and amplifying it by a scale.
   - What do you hear when a signal is processed using the simple reverberation system? When the system is implemented, a series of echos that appear to play right after each other is heard up until a cohesive sound of echos is heard due to the stop time being set to "inf"

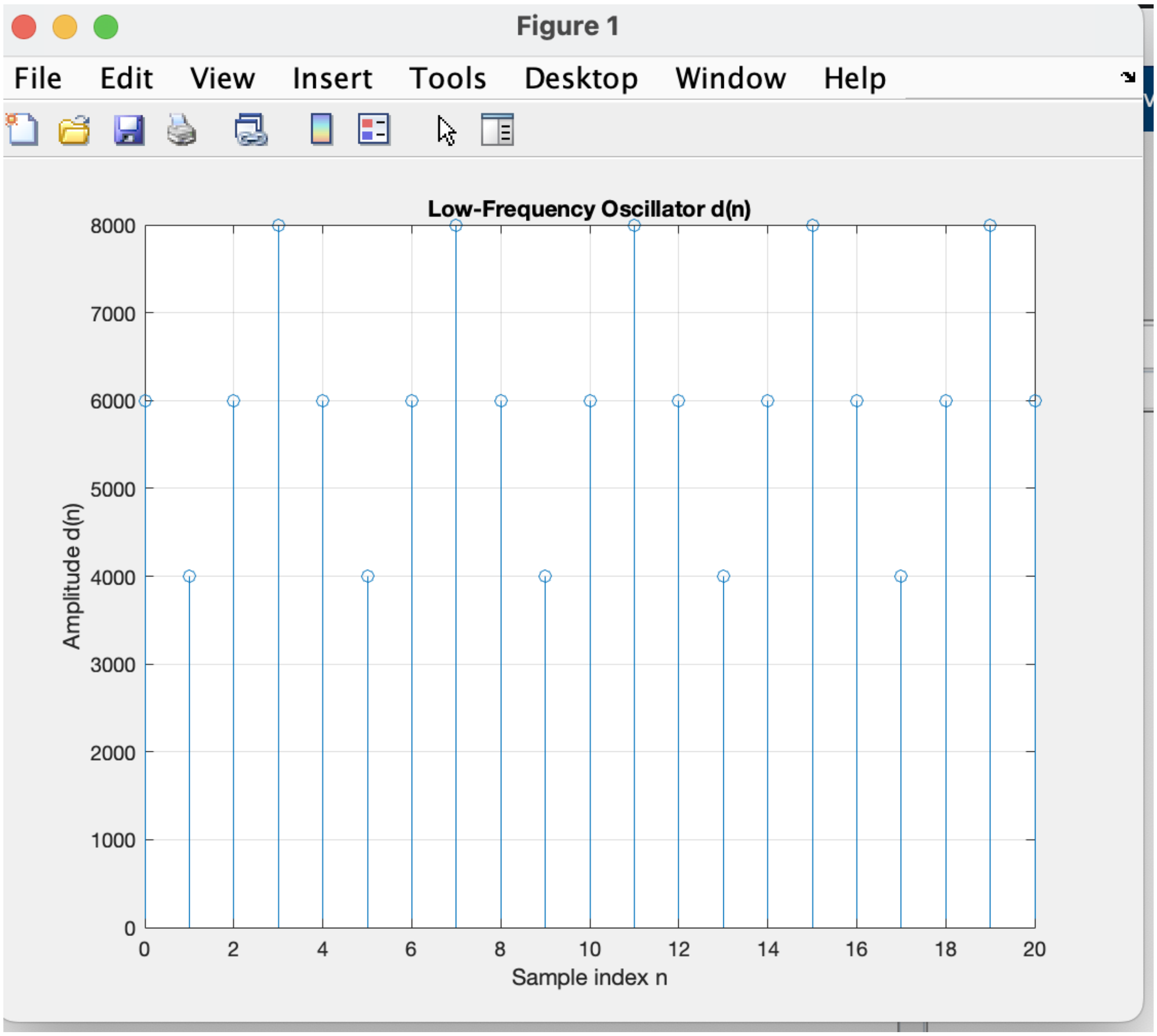
- What effect does changing the parameter $a$ have? Changing the value of "a" manipulates the loudness of the sounds produced in addition to the original sounds, A higher a means louder resulting sounds whilst smaller D implies lower sounding reverbs
- What effect does changing the parameter $D$ have? D controls the sampling spacing between the original sound and reverb. D represents the delay in reverb
- Compare the simple reverberation filter with the single echo filter with the same delay and gain. How do they differ? Whilst the simple reverb feeds back an amplified and delayed version of the output into the system(adding it to the input) to produce an output, the simple echo feeds back a delayed and amplified version of the input added to the original input to create a final sound
- Compare the simple and allpass reverberators. How are they similar? How are they different? Why do you think the allpass reverberator sounds more natural?

  The allpass reverberators sound more natural due to the inclusion of a delayed output to the overall response of the system. They are similar in that the allpass reverberator is just a simple reverberator with a delayed input accounted for . Thus in both cases, the system feeds back an amplified and delayed version of the output into the system(adding it to the input) to produce an output. The allpass reverberator sounds more natural because it emulates the response of a real life system - output as well as input is delayed in reverb

5. **Flanging**:

- What do you hear when a signal is processed using the flanging system? Flanging doesn't introduce a significantly observable change to the system as from before. The sound subtly appears to have a moving sensation due to the moving delays introduced . With song.wav, it produced a disturbing recurring noise.
- What effect does changing the gain, delay, and flange frequency have? The gain modulates the loudness of the moving / delayed sample whilst D changes the magnitude of the static portion of the delay . The flange frequency determines how soon the delay of signals occurs
- Plot the low-frequency oscillator, $d(n)$, for these parameters: $D = 4000$ samples, $f_s = 8000$ Hz, $f_d = 2000$ Hz.

Figure 2: Plot of low-frequency oscillator




- What is the maximum delay when $D = 4000$ samples and $f_s = 8000$? What is the minimum delay? From the graph the maximum is 8000Hz, and the minimum is 2000Hz

# 4 Conclusions

In this lab exercise, we investigated simple digital audio effects, focusing on reverb, echo, and flanging. Utilizing the audio system toolbox in Simulink, we implemented algorithmic functions to create these effects, processing both real-time and recorded speech inputs. Through this experimentation, we gained hands-on experience with implementing input/output systems in Simulink and manipulating various system parameters.

We observed how changes in gain, delay, and modulation frequency affect the audio output, particularly with the flanging effect. Overall, the experiment successfully met its goals, enhancing our understanding of digital audio processing and the practical application of audio effects in real-time scenarios.