Mini Project #1: Digital Audio Effects

Ms. Alice Liu and Prof. Brian Evans

Assigned on Sunday, September 13, 2026
Due on Monday, September 21, 2026, by 11:59 pm via Gradescope submission

Late submission is subject to a penalty of two points per minute late.

Reading: McClellan, Schafer and Yoder, *Signal Processing First*, 2003, Ch. 3. Errata.
Companion Web site with demos and other supplemental information:
http://dspfirst.gatech.edu/
The Web site contains solutions to selected homework problems from *DSP First*.

E-mail address for Ms. Liu (TA) is aliceliu98@utexas.edu. Please post questions on the class Ed Discussion site, which can be answered by anyone in the class. You can post anonymously.

Time Slot	***Monday***	***Tuesday***	***Wednesday***	***Thursday***	***Friday***
11:00 am		Evans (ECJ 1.214)		Evans (ECJ 1.214)	
11:30 am		Evans (ECJ 1.214)		Evans (ECJ 1.214)	
12:00 pm		Evans (ECJ 1.214)		Evans (ECJ 1.214)	
2:00 pm		Evans (EER 6.882 and Zoom)			Liu (EER 6.702 and Zoom)
2:30 pm		Evans (EER 6.882 and Zoom)			Liu (EER 6.702 and Zoom)
3:00 pm		Evans (EER 6.882 and Zoom)	Liu (EER 5.704 and Zoom)	Liu (EER 5.704 and Zoom)	Liu (EER 6.702 and Zoom)
3:30 pm		Evans (EER 6.882 and Zoom)	Liu (EER 5.704 and Zoom)	Liu (EER 5.704 and Zoom)	Liu (EER 6.702 and Zoom)
4:00 pm			Liu (EER 5.704 and Zoom)	Liu (EER 5.704 and Zoom)	
4:30 pm			Liu (EER 5.704 and Zoom)	Liu (EER 5.704 and Zoom)	

Please work in a group of three and create one report together. Each person in a team would submit the same report on Gradescope. Be sure the report represents the independent work of the authors. Submit all code and plots. See Appendix B for additional guidelines.

Per the course syllabus, no Generative AI tools may be used for this mini-project.

1.0 Introduction and Background

In this mini-project, you'll create digital audio effects based on modulation and delay, and evaluate the differences in the original and altered signals through playback and visualizations.

1.1 Sound Waves

When a musician plays, the vibrations of the musical instrument produce sound waves (pressure waves) that humans hear. Just as a musician can alter sound mechanically by bending a guitar string or using a damper pedal on a piano, sound can also be manipulated electronically, either through an amplifier or digitally via a mixing board or computer.

1.2 Musical Notes

When a guitar string is plucked, the string oscillates at a musical note frequency which is inversely proportional to the string length. The pluck also causes the string to oscillate at harmonics of the musical note frequency due to the standing waves that result from both ends of the string being anchored. The oscillating guitar string vibrates the guitar body, producing the sound. A piano works in a similar fashion– pressing a piano key causes a hammer to hit a string anchored at both ends. Over time, the sound will die out as the vibrations die out.

1.3 Modeling Musical Notes

We can model the oscillation in the time domain at the frequency of the musical note f_0 as

$$x(t) = A_1 \cos(2\,\pi\, f_1\, t\, +\, \theta_1)$$

where amplitude A_1, frequency $f_1 = f_0$, and phase θ_1 are constants. We can extend this model to capture the oscillation for N harmonic frequencies where the kth harmonic frequency is $f_k = k\, f_0$:

$$x(t) = \sum_{k=1}^{N} A_k \cos(2\,\pi\, f_k\, t\, +\, \theta_k)$$

We can augment this model to represent the sound dying out with time by multiplying by a decaying function vs. time such as a decaying exponential

$$x(t) = e^{-\frac{t}{\tau}} \sum_{k=1}^{N} A_k \cos(2\,\pi\, f_k\, t\, +\, \theta_k)$$

where τ is a constant in units of seconds (e.g. 0.2 s). If we keep only one frequency,

$$x(t) = A_1\, e^{-\frac{t}{\tau}} \cos(2\,\pi\, f_1\, t\, +\, \theta_1)$$

1.4 *How a Generated Sound is Heard* – See Appendix A.

2.0 Modulation

We modulate our voices when we speak to change volume or pitch. We might whisper to convey private information, or increase the pitch (frequency) of our voice to express excitement. When speaking, we keep people's interest by changing both volume and pitch. When holding a note, a singer might vary the pitch slightly to make the note more interesting to listen to.

In this section, you'll explore two common modulation effects in audio. Vibrato is a change in frequency over time and tremolo is a change in amplitude over time. You'll create each effect by processing an audio signal $x(t)$ to produce another audio signal $y(t)$.

2.1 Frequency Modulation – Vibrato

In musical terms, vibrato is a periodic varying of the pitch that is heard, and can be applied to many instruments such as violins, flutes, and voice.

For simplicity, consider an audio signal that has a single frequency:

$$x(t) = cos(2\pi f_0 t)$$

When applying vibrato to *x*(*t*), the effect is a result of modulating or changing the frequency of a cosine by varying the phase vs. time:

$$y(t) = cos(2\pi f_0 t + \theta(t))$$

where

$$\theta(t) = \frac{\Delta f}{f_\tau} cos(2\pi f_\tau t)$$

(a) Write the Matlab code to

1. Generate a pure tone $x(t)$ lasting 5 seconds at frequency f_0
2. Generate periodic phase-shift $\theta(t)$ by using variables `deltaf` for Δf and `ftau` for f_τ
3. Apply the periodic phase-shift to the original pure tone to produce the vibrato effect $y(t)$
4. Play back $x(t)$ and $y(t)$ as audio
5. Plot the spectrograms zoomed into a small frequency range around f_0

For the sampling rate in the Matlab code, use 48000 Hz which is a standard audio rate. For frequency f_0, please choose a frequency of a note on the Western scale in the fourth octave.

(b) For analysis, please discuss the following:

1. How does Δf affect the vibrato we hear?
2. How does f_τ affect the vibrato we hear?
3. Compare and contrast the time and frequency representations of the original pure tone versus the pure tone with vibrato. What differences are there, if any? Explain.

2.2 Amplitude Modulation - Tremolo

The tremolo audio effect is a result of modulating the amplitude of a signal. In musical terms, we consider specifically the tremolo that refers to a variation in volume (not to be confused with tremolo that refers to oscillation of the note).

For the tremolo effect, a pulse-shaping signal $p(t)$ is applied to the amplitude of $x(t)$

$$y(t) = p(t)x(t)$$

Students can generate any pulse-shaping signal for amplitude modulation, and we provide an example sinusoidal pulse-shaping signal below:

$$p(t) = Acos(2\pi f_\tau t)$$

In this section, you'll apply the tremolo effect on a pure tone and a mystery signal.

(a) Write Matlab code to

1. Analyze the `mysterySignal_tremolo` to determine its frequency components
2. Generate a pure tone, x, for the strongest frequency component in step 1
3. Generate a pulse-shaping signal, p, for amplitude modulation
4. Apply the pulse-shape to the pure tone signal to produce the tremolo effect
5. Apply the pulse-shape to `mysterySignal_tremolo` to produce the tremolo effect
6. Play the audio signals generated in steps 4 and 5
7. Plot the spectrograms for signals in steps 4 and 5 zoomed into a small frequency range around the strongest frequency component identified in step 2

For the sampling rate in the above code, use the sampling rate for mysterySignal_tremolo .

(b) For analysis, please discuss the following:

1. Give the fundamental frequency and the next four harmonic frequencies for the `mysterySignal_tremolo`.
2. Provide the equation used to generate the pure tone in part (a)2.
3. What value of A did you choose for the pulse-shaping signal? How does the value of A affect the audio quality of the signal?
4. After applying the pulse-shaping signal in part (a)4 what is the resulting fundamental period? Is this what you expect? Explain.
5. Explain similarities and differences between the amplitude modulation of the pure tone in part (a)4 vs. the amplitude modulation of the `mysterySignal_tremolo` in part (a)5?

3.0 Time Shift – Echo

Our goal in this section is to examine the echo audio effect that relies on time-shifting and dampening a signal. An echo audio effect is when an initial signal is heard multiple times with decreasing volume over time.

Students will analyze and reproduce an echoing mystery signal as best as they can. We begin by walking students through how to create an echoing effect on a pure tone signal through applying time-shifts and amplitude damping. Then, we task students with using the techniques learned in order to recreate the echoing mystery signal to the best of their ability.

3.1 Analysis of Echoing the Mystery Signal

We now analyze the echoing mystery signal. Students are recommended to play the audio signal and examine the time and frequency representations of the audio signal.

(a) Write MATLAB code to…

1. Play `mysterySignal_echo` as an audio file
2. Plot spectrogram of `mysterySignal_echo`to determine its frequency components

(b) For discussion and analysis, please address the following:

1. What frequency components are present in `mysterySignal_echo`?
2. How many 'echoes' can you identify in `mysterySignal_echo`l?

3.2 Generating an Echoing Pure Tone: Part 1 - Time-Shifts

We consider generating echoes of a pure tone. This approach will have drawbacks because the pure tone does not decay in amplitude, and we'll address the drawbacks in Section 3.3.

For simplicity, we consider a single echo. This means there's a direct path and a secondary delayed path, similar to what you had seen in homework problem 1.4:

$$y_1(t) = x(t) + x(t - t_1)$$

This section will ask you to time-shift the pure tone signal and combine it with the initial signal to produce the first attempt at generating an echo signal.

(a) Write MATLAB code to…

1. Generate a pure tone, x, based on the frequency component with the highest magnitude
2. Determine a time-shift based on your analysis of the `mysterySignal_echo`
3. Apply the time-shift (samples) to the pure tone
4. Combine the initial pure tone with the time-shifted pure tone
5. Playback the combined signal $y_1(t)$

For the sampling rate in above Matlab code, use the sampling rate of `mysterySignal_echo`.

(b) For discussion and analysis, please address the following:

1. Provide the explicit equation used for generating the pure tone.
2. What time-shift t_1 did you choose and why?
3. Does it sound like what you expect from an echo? Why or why not?

3.3 Generating an Echoing Pure Tone: Part 2 – Signal Damping

Students may have noticed that there was audio distortion in *Section 3.2* when combining the original and time-shifted signals together. Additionally, students may have noticed that simply combining time-shifted versions of the initial signal does not produce an "echo" effect. Rather, we simply hear the same tone sustained twice.

In this section, we will resolve both concerns of audio distortion and sustained signals by applying signal damping with a decaying exponential function shown below:

$$p(t) = e^{-kt}$$

(a) Students should write MATLAB code to…

1. Generate an exponentially decaying pulse-shaping signal
2. Apply the pulse-shaping signal to the initial and time-shifted pure tones and combine
3. Playback the signal

(b) For discussion and analysis, please address the following:

a) What value of k did you use to remove the audio distortion present in the signal?
b) Compare and contrast your generated echoing signal with the original `mysterySignal_echo`.
c) What change(s) would you implement to make your recreated signal better match `mysterySignal_echo`? Explain.

4.0 Project Deliverables

The mini-project consists of 2 main deliverables: a written report and code submission.

4.1 Written Report (PDF format)

Please write a self-contained narrative report. The audience is someone who has taken the equivalent of this class. The report should provide references to the textbook and other sources as needed. Please refer to the hints in Appendix B, which apply to homework assignments and mini-project reports, as well as the following additional guidelines for the mini-project:

1. Introduction – Students should introduce the purpose of this mini-project and provide an overview of what their report will discuss.
2. Modulation
    a. Vibrato Discussion & Analysis
    b. Tremolo Discussion & Analysis
3. Time Shifts
    a. Echo Discussion & Analysis
4. Conclusion – Students should discuss any insights or learnings as a result of completing the mini-project.
5. References
6. Usage of additional MATLAB Toolboxes (if any)

The written report will be graded on responses to the analysis questions, clarity, and formatting.

4.2 MATLAB Code (.zip compressed folder)

Students must submit a .zip compressed folder that contains all their code used to complete this mini-project. The code organization should be clear, such that the TA does not need to guess and check which file to run to reproduce results for a particular analysis. Ensure that your code runs without errors before submission! If you have questions or concerns, please feel free to join TA office hours for clarification.

References

1. J. Marr. Interview with T. Murphey. “Johnny Marr on how he created the sound for "How Soon Is Now?" without effects pedals.” Westword.com, 17 April 2013.
2. D. Q. Naiman. “SoundWaves.” Mathematics of Music, Johns Hopkins University Whiting School of Engineering.
3. Native Instruments. “What is an LFO? How to use low frequency oscillators in music production.” Native-instruments.com, 17 July 2023,.
4. J. Owens. “Pitch or Volume? The Difference Between Tremolo and Vibrato.” Fender, 6 February 2023,.
5. M. Rice. “Chapter 3 - Single-carrier modulation.” *Academic Press Library in Mobile and Wireless Communications*, Academic Press, 2016, pp. 73-119.
6. B. L. Roberts. “Exponential Decay.” PY231 Spring 2025 - The Physics in Music”, Boston University.

Appendix A: How a Generated Sound in Heard – It's Complicated

There are at least three factors to how a sound is heard from an audio perception view:

1. **Playback system**. Playback systems on laptops and tablets attenuate frequencies below 200 Hz and above 15 kHz. Frequency response plot.
2. **Listening environment**. Could be noisy. Tones and other signals might also be playing. There are reflections off of walls, floors and ceilings. All this can be mitigated to some extent by listening over earbuds or headphones.
3. **Human auditory system**. A healthy human auditory system for someone 17-19 years of age can generally hear frequencies from 20 Hz to 20 kHz. The human auditory system typically has a stronger response to mid-range frequencies than low or higher frequencies; e.g., the response to 880 Hz is generally higher than the response to 440 Hz. This assumes that the human auditory system has not suffered any damage or infection to reduce its sensing capabilities. Here are several models of frequency responses for human hearing. Here measurements of frequency responses for human hearing from 500 Hz to 6000 Hz due to age.

How significant is an increase of 10 dB in sound level? It will sound twice as loud to the human auditory system:

Understanding Sound Intensity: How Many Decibels Does It Take To Double Volume?

Likewise, a decrease of 10 dB in the sound level is perceived by the human auditory system as half as loud.

When listening to sound, there are additional factors such as audio cognition, interplay with what a human might be seeing, one's emotional state, and ability to focus on the sound.

Appendix B: Homework and Mini-Project Guidelines

Here are some things you should follow for all assignments.

Amount of work to show:

1. An explanation should be given for every single answer. Answers written without explanation will lose two-thirds of the points allotted for that part.
2. Only "standard" formulas (like Euler's formula, trigonometric formulas, etc.) can be used without a reference. If you're using something non-standard, then please put a reference to the formula number in the book, or whatever source you got it from. Just using the final result of a similar problem done in the class, and omitting the intermediate steps, is not okay. You have to show your work.
3. There shouldn't be big jumps in logic from one step to the next.
4. For everything, expect to show at least one intermediate step between the first line and the answer. Even if it seems unnecessary to you, please err on the side of caution. Things that seem obvious to you when you're writing the solution are not quite so obvious for someone reading it.
5. If you're in any doubt about how much work to show, please ask the instructor or the teaching assistant.

MATLAB source code guidelines:

1. Put a comment before the solution of each part, telling the question number of the solution.
2. If you're using complicated logic, leave a comment telling what that block of code is supposed to do.
3. Use variable names that related to their meaning/use.
4. Avoid using two different variables for the same thing.
5. Try to avoid using "magic numbers" in the code. If you're using a number, write a comment telling me how you derived it.
6. Make sure that your code will compile & run in a clean workspace; i.e., one without any variables present. Use a clear all; at least once before submitting it.
7. No marks will be deducted based on the efficiency of the code unless the problem asks you to write efficient code.

Technical points:

1. Merge all the files together into one PDF file.
2. Please adjust the contrast, exposure etc., to get a good scan quality so that the TA can easily read what you write. Take extra care to get a good scan for parts written in pencil.
3. For the MATLAB code you write for an assignment, please copy the code into Word or include a screenshot showing the code. Do not submit handwritten code.

Other things:

1. All plots must have axis labels, with units.
2. Final answers must be boxed, or underlined or otherwise differentiated from the rest of the solution.

3. All final answers must have units, if they exist.
4. Read the questions carefully.
5. Try to answer all parts of a question together. If the solution to some parts of a question is written elsewhere, then leave a note telling the reader where to find it.

Organization of a mini-project report:

Please write a self-contained narrative report. The audience is someone who has taken the equivalent of this class. The report should provide references to the textbook and other sources as needed. Please refer to the hints above, which apply to homework assignments and mini-project reports, as well as the following additional guidelines for the mini-project.

Here are example mini-project #1 reports written by the instructors:

- "FM Synthesis for Musical Instruments" (2018)
- "Sinusoidal Speech Synthesis" (2021)
- "Music Synthesis" (2023)

Please see the homework hints page for specific guidelines for this project.