

Lab 1 Pre-lab : Modifying Signals

In this lab, you will work through a series of exercises to introduce you to working with audio signals and explore the impact of different amplitude and time operations on signals. This is a two-week lab.

Lab 1 Turn-in Checklist

- Pre-lab (upload to canvas before lab)
- Lab 1 Jupyter notebook with code for the first 4 exercises assignment in separate cells. Each assignment cell should contain markdown cells (same as lab overview cells) for the responses to lab report questions. Include your lab members' names at the top of the notebook.
- 1 individual Jupyter notebook with code + markdown cells for the last exercise

The rest of this document are the pre-lab exercises to be solved and answered before coming to the lab

Note: The pre-lab and last exercise should be done **individually**, and all other assignments should be completed in groups of 3-4 people.

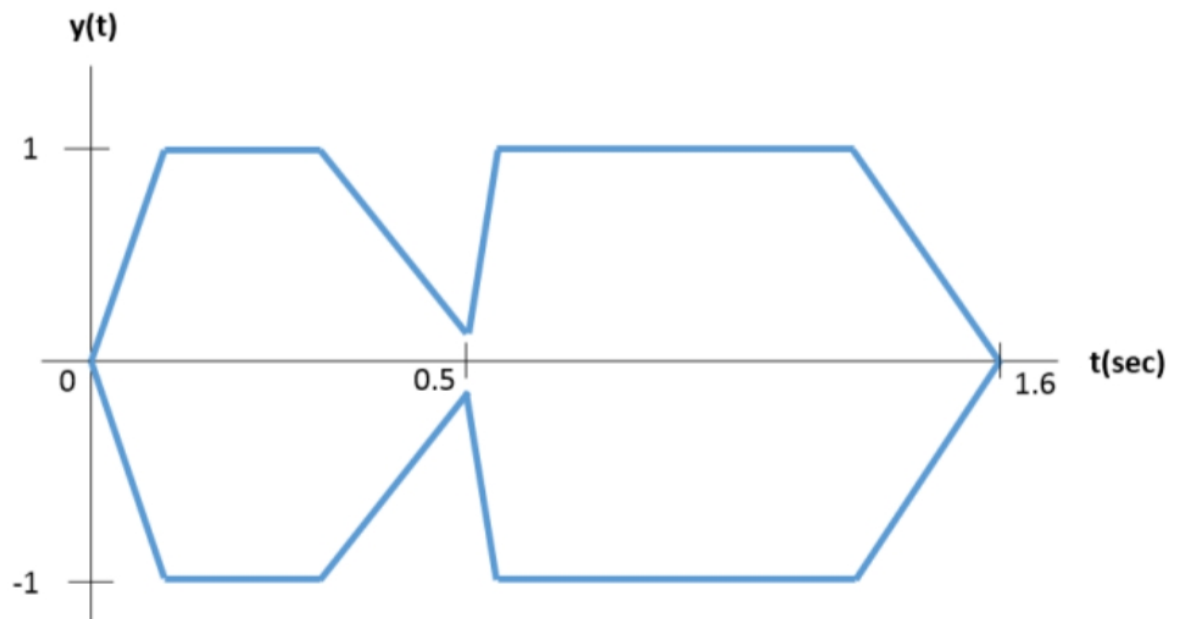
Complete the following exercises and submit answers as PDF (File>Save and Export As... > PDF)

1. An audio signal $y(t)$ is sampled with $f_s=32$ kHz. What sampling period does that correspond to? If you mistakenly play the signal with $f_s=16$ kHz, will it be shorter or longer than the original? How else will it sound different?

The 32 kHz sampling frequency corresponds to a period of 31.25 microseconds. If you play a signal with half of the sampling rate, it will be twice as long as the original signal. If you play it with twice the sampling rate, it will be half the length of the original signal.

Respectively, it will sound slower for $\frac{f_s}{2}$ and faster for $2f_s$.

2.The envelope of an audio signal $y(t)$ (the train sound that you will work with in this lab) is approximated below:



a. Sketch the signal $y(2t)$. Label the height and the time points corresponding to $t = 0$, $t = 0.5$, $t = 1.6$ after the time scaling. If you played $y(2t)$, would it sound like $y(t)$ has higher frequencies or lower frequencies?

Playing $y(2t)$ would sound like the signal had higher frequencies.

b. Repeat (a) for the signal $y(0.5t)$

It would sound like it had lower frequencies.

c. Sketch the signal $y(t-1)$ and $y(t+0.5)$, again labeling critical time points.

The first transformation would cause the signal to sound delayed by 1 second, with all other qualities the same. The second transformation would cause it to sound prematurely, 0.5 seconds before the initial sounding time, still withstanding the other qualities of the signal.

3. When the signal is digitized, you need to implement the time shift in terms of the number of samples: $y[n-n_1]$ and $y[n+n_2]$. Find n_1 and n_2 (corresponding to $t_1=1$ and $t_2=0.5$, respectively) for the case when $f_s=32\text{kHz}$.

$$n_n = t_n * f_s$$

Therefore, $n_1 = 32,000$ samples, and $n_2 = 16,000$ samples.

4. On a computer, we may have the constraint of keeping the time window fixed. Assuming the time window is constrained to be $[0,3]$ sec, which of the time transformations in part 1 will require you to throw away some of the transformed signal? If you were to implement $y(t)=x(2(t+1.5))$ with a fixed time window, would it be better to scale first or shift first, or does it not matter?

The transformations $y(0.5t)$ and $y(t + 0.5)$ would require us to throw away a certain amount of information away. In any case, for $y(t) = x(2(t + 1.5))$ the same amount of information will

be lost. If we scale first, then the signal's time length is divided by two, then we would shift it left by 1.5 seconds, putting 1.5 seconds of the signal outside of the window. on the other case, if we first shift the signal left by 3 seconds, and then scale by $1/2$, we still lose 1.5 seconds of information, so it does not matter.

In []:

4) INVERTIBILITY

— A SYSTEM IS INVERTIBLE IF IT MEETS THIS CRITERION

$$T\{x(t)\} = y(t) \Rightarrow \exists T_1 \text{ s.t. } T_1\{y(t)\} = x(t) \Rightarrow \forall y$$

— A SYSTEM IS INVERTIBLE IF EACH INPUT RESULTS IN A UNIQUE OUTPUT & VICE VERSA

$$T\{x(t)\} = |x(t)| \text{ NOT INVERTIBLE}$$

PRELAB 1 SKETCH

