

# Digital Audio Workshop 2.0

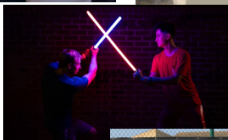
## IEEE UCF Skills Series

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February 12, 2024

# About Me



1. 3rd year Electrical Engineering Major, Signals & Communications Track
2. Systems Engineering CWEP at Lockheed Martin (GNC team) - almost 2 years
3. 2023-2024 IEEE UCF Vice President
4. Undergraduate Research in UCF ECE's NWSL, & CFRSL
5. Free Time: Church, Reading, Music Production, Tennis/Pickleball/Basketball, Spicy Food

# Overview

DSP theory

yt-dlp via command prompt

MATLAB

LMMS Demo

# Audio Fundamentals

*Longitudinal* sound waves are a form of energy transfer through a medium, usually the air. However, in this workshop, we will focus on *transverse* waves, which we are used to seeing in our classes. Audio is a mathematical phenomenon, and by extension, an electrical one as well.

# Signal Characterization

**Analog** - in a finite range of amplitude, has an infinite possibility of numerical values

**Digital** - in a finite range of amplitude, has a finite possibility of numerical values

**Continuous** - in a finite duration of time, has an infinite possibility of numerical values

**Discrete** - in a finite duration of time, has a finite possibility of numerical values

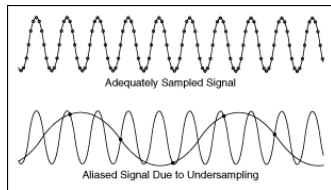
# ADC, DAC

Analog-to-Digital Conversion (ADC) changes a continuous analog signal into a discrete digital. This method is called *sampling*.

Digital-to-Analog Conversion (DAC) is the reverse process. This method is *sample-and-hold*.

## A Rule to Follow in DSP

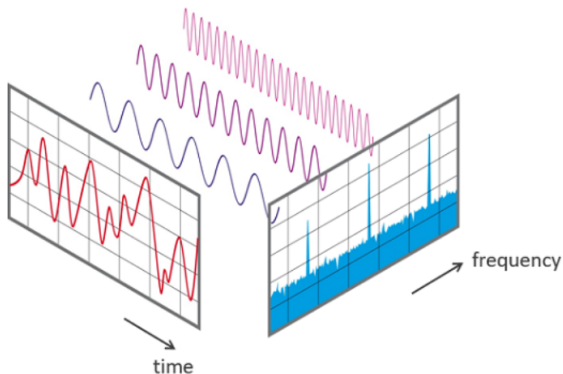
To prevent aliasing, a periodic signal should be sampled at greater than 2x the frequency of its highest component.



### Theorem (Nyquist Sampling Theory)

$$f_s \geq 2 * f_0$$

# The Frequency Domain



Question: Is there such a thing as negative frequency?



# Fourier Analysis

1878: Joseph Fourier publishes  
"The Analytical Theory of Heat."  
This is adapted to the discipline  
of electrical signals and called the  
*Fourier Series*.

Further adaptations:  
FT/FFT, DFT/DTFT

SECTION II.	
FIRST EXAMPLE OF THE USE OF TRIGONOMETRIC SERIES IN THE THEORY OF HEAT.	
ART.	PAGE
171—178. Investigation of the coefficients in the equation	
$1 = a \cos x + b \cos 3x + c \cos 5x + d \cos 7x + \text{etc.}$	
From which we conclude	
$a_k = \frac{1}{2k-1} \frac{4}{\pi} (-1)^{k+1},$	
or	$\frac{\pi}{4} = \cos x - \frac{1}{3} \cos 3x + \frac{1}{5} \cos 5x - \frac{1}{7} \cos 7x + \text{etc.} \quad . \quad . \quad . \quad 137$

# Tools

yt-dlp is an open-source program that allows for quick extraction of audio from YouTube videos. Download here: [▶ GitHub](#) (requires ffmpeg and Python 3.8 or later)

ffmpeg is "fast forward moving picture experts group." It is used by yt-dlp to extract and process audio. Download here: [▶ GitHub](#)

Installation Tutorial: [▶ YouTube](#)

Dependencies: [▶ MediaFire](#)

## yt-dlp command

```
yt-dlp --ignore-errors --format bestaudio  
--extract-audio --audio-format mp3 --audio-quality 160k  
--output "%(title)s.%(ext)s" "INSERT-URL-HERE"
```

For a playlist, insert - -yes-playlist before the URL. ffmpeg files must be in the PWD or added to your PATH.

# MATLAB functions

1. `audioread()` - why are there 2 columns of data?
2. `audiowrite()`
3. `sound()`
4. `fft()`
5. `fftshift()`
6. `abs()`

The units for the `audioread()` data are normalized. We aren't given the specific energy of the wave, only the relative amplitude. Really this tells us the pressure readings from whatever transducer recorded the signal.

# MATLAB scripts

generator.m produces a sum-of-sines with a uniformly distributed noise layer and outputs to playback so we can listen.

denoise.m is where we design a filter to recover a portion of the signal. While some noise is removed, we do not perform noise cancellation.

# Low Frequency Oscillator (LFO)

Using the digital audio workstation LMMS (Linux Multi-Media Studio), we can simulate a digitally controlled, multi-input, single output system. In this demo, the dynamic system is a lowpass filter and the controller is an LFO. The inputs are the control signals and the output is the synthesized sound.

Is this a linear time-invariant system? [▶ Article](#)

## Relevant Coursework

- EEL3123C Linear Circuits II
- EEL3552C Signal Analysis & Analog Communication
- EEL4750 DSP Fundamentals
- EEL4515C Digital Communications
- EEL4140C Analog Filter Design
- EEL5513 DSP Applications
- EEE5555 Surface Acoustic Waves
- EEL5630 Digital Control Systems