YIFENG (YVONNE) YUAN

Audio DSP | Audio Machine Learning

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Audio and computer music researcher experienced in developing audio plugins, training audio models, and building interactive sound systems. Works with Python, PyTorch, C++/JUCE, and Max/MSP.

Also a sound designer and film composer, with film work presented at the Venice Biennale and Busan International Film Festival.

EDUCATION

Ph.D. Music (Computer Music), UC Santa Barbara - Expected Fall 2026
M.S. Media Arts and Technology, UC Santa Barbara - Summer 2026
M.A. Music Composition, UCLA - 2019-2021 | B.A. Musicology & Sociology, UCLA - 2014-2018
Related Coursework: Digital Audio Programming, Signal Analysis and Processing

RESEARCH EXPERIENCE

Graduate Researcher - UC Santa Barbara, Department of Music Oct 2023–Present

SPEKANA - Real-Time Audio Analysis Plugin | C++ / JUCE / VST3

Accepted for online poster presentation at Audio Developer Conference (ADC), November 2026.

- Developed an audio analysis plugin that translates spectral features into MIDI notes and CC messages for creative and generative music workflows.
- Implemented spectral peak detection in C++ using JUCE's FFT module, with a 2048-point FFT, Hann window, and 512-sample hop; added configurable MIDI mappings, linear range scaling, smoothing, and musical scale options.
- Evaluated pitch output and peak-detection behavior using DAW-generated sine tones, multitone signals, and white noise; added envelope-relative thresholds to suppress weak peak candidates.

Peripheral Voice - Neural Vocal Synthesis | RAVE / Colab / Max/MSP

- Recorded, edited, and segmented approximately 150 minutes of experimental vocal audio; trained RAVE models in Google Colab for nonverbal vocal synthesis.
- Identified excessive silence in the initial dataset, cleaned the recordings, and retrained; compared checkpoints through qualitative listening and integrated a trained model into Max/MSP.

Live Vocal Analysis System | Python / wav2vec 2.0 / Max/MSP

Talk presented at Co-Creativity in Music, Sound, and AI, UC Riverside, June 2026

- Integrated a pretrained wav2vec 2.0-based model to estimate arousal, valence, and dominance from live mono 16 kHz microphone input in Python.
- Used 1-second rolling windows and a 250 ms hop; routed estimates over OSC to control vocal distortion, reverb/notch filtering, and delay in Max/MSP.
- Tuned parameter-change thresholds and applied continuous smoothing to stabilize live effect control.

Field Poetry - Environment-Responsive Sound Box | Arduino Uno / C++ / Max/MSP

- Wrote Arduino code for sensor acquisition and smoothing; sent light, temperature, humidity, and ambient sound amplitude over USB serial to control generative music in Max/MSP.

RELEVANT EXPERIENCE

Audio Engineer - UC Santa Barbara | 2023-2025

- Configured speaker routing for playback systems with up to eight channels, along with performer-monitor routing; supported microphone placement, audio interface/mixer setup, and soundchecks.

Teaching Assistant | UC Santa Barbara (2022-2026); UCLA (2019-2021)

Courses: Musicianship series, Music Theory series, Jazz History (Academic Writing)

- Independently planned and taught two weekly sessions at UCLA; developed critical-listening exercises, rhythmic/melodic/harmonic dictation assignments, and midterm and final assessments.
- Led Jazz History discussions using comparative listening of historical and contemporary recordings; guided students in music analysis and academic writing.

TECHNICAL SKILLS

Programming and Prototyping: C++, Python, MATLAB, JUCE/VST3, Max/MSP, gen~, OSC, Git, Arduino
Machine Learning: PyTorch, RAVE, wav2vec 2.0, audio data preparation, model training, evaluation
Audio DSP: FFT/STFT, windowing, spectral descriptors, peak detection, filtering, smoothing