

# A3: Real-time Activity Sensing

Machine Learning and Sensing | Spring 2025

# Assignment Overview

## Goal:

- Deploy a real-time inference system
- Compare ML vs DL in real-time

## Submit:

- Code (.ipynb or .py)
- requirements.txt
- Two demo videos (ML + DL model real-time inference)

# A3.1– Real-Time Acoustic Inference (20 pts)

**Objective:** Deploy real-time system using best model from A1, and DL model from A2 (or state-of-the-art DL model).

## Steps:

- Mic audio → sliding window → inference → live display
- Record 2-3 minutes for each video including the 5 main activities (randomly).
- Display:
  - Waveform (real-time)
  - Predictions + confidence (ML + DL)
  - Inference time (ms)

## Reference:

- Mic input: PyAudio, python-sounddevice
- GUI: Tkinter, PyQt
- Ubiquitous:  
[https://github.com/FIGLAB/ubicoustics/blob/master/example\\_liveprediction\\_simple.py](https://github.com/FIGLAB/ubicoustics/blob/master/example_liveprediction_simple.py)  
[https://github.com/FIGLAB/ubicoustics/blob/master/example\\_liveprediction\\_detail.py](https://github.com/FIGLAB/ubicoustics/blob/master/example_liveprediction_detail.py)

# A3.1– Real-Time Acoustic Inference (20 pts)

## Other Models:

1. [Wav2Vec 2.0](#)
2. [AST \(Audio Spectrogram Transformer\)](#)
3. [AudioMAE](#)

## Pre-trained models

Model	Finetuning split	Dataset	Model
Wav2Vec 2.0 Base	No finetuning	<a href="#">Librispeech</a>	<a href="#">download</a>
Wav2Vec 2.0 Base	10 minutes	<a href="#">Librispeech</a>	<a href="#">download</a>
Wav2Vec 2.0 Base	100 hours	<a href="#">Librispeech</a>	<a href="#">download</a>
Wav2Vec 2.0 Base	960 hours	<a href="#">Librispeech</a>	<a href="#">download</a>
Wav2Vec 2.0 Large	No finetuning	<a href="#">Librispeech</a>	<a href="#">download</a>
Wav2Vec 2.0 Large	10 minutes	<a href="#">Librispeech</a>	<a href="#">download</a>
Wav2Vec 2.0 Large	100 hours	<a href="#">Librispeech</a>	<a href="#">download</a>
Wav2Vec 2.0 Large	960 hours	<a href="#">Librispeech</a>	<a href="#">download</a>
Wav2Vec 2.0 Large (LV-60)*	No finetuning	<a href="#">Libri-Light</a>	<a href="#">download</a>

# Write-up Summary

## A3.1 Report:

- Describe the pipeline: audio capture → processing → inference → display.
- How did you handle buffering and overlapping windows?
- What were the typical inference times for each model?
- Was one model noticeably faster or more stable?
- What kind of interface did you build? How did you visualize predictions and confidence?
- How well did the system respond to different environments (quiet, noisy, echo, etc.)?

# Submission Checklist

**Code:** Notebook or script + requirements.txt

**video\_ml.mp4:** Real-time demo with ML model (A1/A2)

**video\_dl.mp4:** Real-time demo with DL model (A3.1)