

Beatbox to Analog Drum Translation Using a Convolutional Neural Network

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## **Presentation Roadmap**

- Problem Statement & Purpose
- Mel Spectrograms
- Dataset & Data Preparation
- CNN Model Architecture & Results
- Sample Triggering
- Future Considerations

# Project Statement & Purpose

## **Project Statement**

Utilize a Convolutional Neural Network (CNN) to translate a beatboxing recording into a corresponding audio track, matching the sound and timing of a performance using preloaded drum samples.



### **Project Purpose**

- Facilitate quick form musical idea creation in a way that can be natural, intuitive, and inspiring for musicians
- Former NMF based solution relied heavily on analytically chosen parameters Train a CNN that will result in a more efficient, accurate, and robust
  - translation



Mel Spectrograms

#### **CNN Input: Mel Spectrograms**

Traditional Spectrograms

- Result of taking Fourier Transform of audio signal
- Visual representation of the frequency content of the audio signal
- Allows us to treat the audio information like any other image recognition task



Hi-hat Closed

Kick

#### **CNN Input: Mel Spectrograms**

- Represents the frequency information more closely to how humans actually perceive sound
- Focusing on the most important aspects
- More meaningful representations
- More visually distinct





Hi-hat Closed



## **Dataset & Data Preparation**

#### Dataset

- Amateur Vocal Percussion (AVP) dataset [1]
- Contains 28 Participants with the following .wav files:
  - High Hat Open
  - High Hat Closed
  - Kick
  - Snare
  - Improvisation
- Each .wav file 10-12 seconds long, with about 30 percussive sounds
- Each has accompanying .csv file which contains the onset time, instrument, and phenome of each sound

[1] A. Delgado, S. K. T. McDonald, N. Xu, and M. Sandler, "A new dataset for amateur vocal percussion analysis," *Proceedings of the 14th International Audio Mostly Conference: A Journey in Sound*, 2019. DOI: 10.1145/3356590.3356844

#### **Data Extraction**

- Each of the sounds extracted from the .wav files using the corresponding onset
- sound[i]= audio[ (onset[i]-1000) : onset[i+1] ]
  - Where onset is the sound location in samples, and audio is the .wav file
  - Buffer the beginning of the sample by 1000 to prevent smearing in the mel spectrogram
- Each sound is then fixed to a length of 11025 samples



#### **Data Augmentation**

- To increase robustness of the network, the dataset is augmented with samples that have added noise, pitch shift, time shift, and gain change
- This doubles the amount of data from 4873 samples to 9746 samples

```
def apply_augmentation(samples: np.array):
qaussian noise = AddGaussianNoise(
     min amplitude=0.001.
     max amplitude= 0.015,
     p=0.5
time_shift = Shift(
     min fraction=-0.2,
     max fraction=0.2,
     rollover=False,
     fade=True,
     p=0.5
pitch_shift = PitchShift(
     min semitones=-0.5,
    max semitones=0.5,
     p=0.25
gain = Gain(p=0.5)
augmenter = Compose(
     [time shift, gain, pitch shift, gaussian noise])
return augmenter(samples=samples,
                  sample rate=44100)
```

#### **Data Preparation**

- For each file:
  - 1. Extract each hit using the onset locations in the .csv
  - 2. Apply augmentation
  - 3. Get mel spectrogram of the augmented audio
  - 4. Add corresponding label, spectrogram, and audio to the dataset
- After data extraction and preparation, the mel spectrograms are saved into a DataLoader to avoid redundant preparation
- Training, validation and testing is a 3:1:1 split

# CNN Model Architecture & Results

### **CNN Architecture and Training**

- Input size of (N, 3, 256, 256)
- Peak training accuracy of ~97%
- Validation accuracy ~94%

Layer (type:depth-idx)	Input Shape	Output Shape
Network	[1, 3, 256, 256]	[1, 4]
-Sequential: 1-1	[1, 3, 256, 256]	[1, 4]
└─Conv2d: 2-1	[1, 3, 256, 256]	[1, 32, 252, 252]
└─MaxPool2d: 2-2	[1, 32, 252, 252]	[1, 32, 126, 126]
└─ReLU: 2-3	[1, 32, 126, 126]	[1, 32, 126, 126]
└─Conv2d: 2-4	[1, 32, 126, 126]	[1, 64, 122, 122]
Dropout2d: 2-5	[1, 64, 122, 122]	[1, 64, 122, 122]
MaxPool2d: 2-6	[1, 64, 122, 122]	[1, 64, 61, 61]
ReLU: 2-7	[1, 64, 61, 61]	[1, 64, 61, 61]
└─Flatten: 2-8	[1, 64, 61, 61]	[1, 238144]
Linear: 2-9	[1, 238144]	[1, 200]
	[1, 200]	[1, 200]
L Dropout2d: 2-11	[1, 200]	[1, 200]
Linear: 2-12	[1, 200]	[1, 4]
Total params: 47,683,500 Trainable params: 47,683,500 Non-trainable params: 0 Total mult-adds (M): 965.08		
Forward/backward pass size (MB): 23.88		
Params size (MB): 190.73		
Estimated Tatal Size (MD), 215 40		

# Sample Triggering

### **Input Onset Detection & Separation**



- Separate individual drum sounds using Librosa onset\_detection function
  - Can encode detected onsets in time, frames, or samples
- Predict label of individual sound using CNN
- Based on label and detected onset, trigger sample

## **Future Considerations**

## **Future Considerations**

- Accuracy still improving at 10 epochs
  - Shows promise for expanded training
    - More epochs
    - More training data (more diverse, more augmentations)
- Product consideration continued learning with user feedback



Questions?