

Audio Tagging Challenge

Machine Learning II

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Introduction

Facial and audio recognition have become a large part of our lives. We use our voices to control smart devices, such as home appliances, and sometimes even the home. We get music recommendations at digital storefronts based on what our individual preferences are, while almost every social media platform automatically recognizes faces in photos. We are very curious to find out how those algorithms work, and after a semester of studying deep learning, see if we can develop a mechanism of our own.

So far what we have been working on is image tagging. However, in order to apply what we learned on a broader level, as well as deepen our understanding on the algorithm, we chose to challenge ourselves with a project about audio tagging. The greatest difference between the audio tagging and image tagging is that we have an extra dimension of time to consider. Therefore, we designed a network with LSTMs to train the network, and classify the audio clips with as high accuracy as possible.

Description of the Project

Dataset

The “Medley” dataset we use consists of 21571 3-second audio clips, each with a single label, distributed among 8 classes: clarinet, distorted electric guitar, female singer, flute, piano, tenor saxophone, trumpet, and violin. Each clip on the csv file which the dataset provides has already been prelabeled as training, validation, and test. The training set size is 5841, the validation set size is 3494, and the test set size is 12236.

Deep Learning Network and Training Algorithm

We designed our network with LSTMs since the data we extracted from the dataset is a time series data. We designed the network with Pytorch framework, and coded the model with nn module. Two layers of LSTMs were put into our model, with one layer of dropout 0.1 to prevent overfitting. One extra linear layer was used to get the output we need. We used cross entropy as our loss function because it combined softmax and NLL loss functions, and we hoped to get a more complete result from it. Adam, Adadelata, and SGD were all tried as optimizers, and we hoped to find the one with the highest accuracy.

The data we used was two fold. Since we needed to extract the data from the audio clip files, we first considered the extraction in the sense of time domain using wav file. To make the prediction more accurate, we also considered extracting other features in frequency domain. By using librosa package, we were able to extract the Mel frequency cepstral coefficients (MFCCs), spectral centroid and spectral contrast data from each clip. Theoretically, this data should train the network much better since the frequency domain captures the differences of the classes more distinctly.

To understand the concept of MFCCs, we need to have a basic knowledge of Fourier Transform. Simply put, a Fourier Transform extracts sin and cos waves with different features (i.e. amplitude, frequency, etc) from a composed wave, and expresses it in a frequency domain:

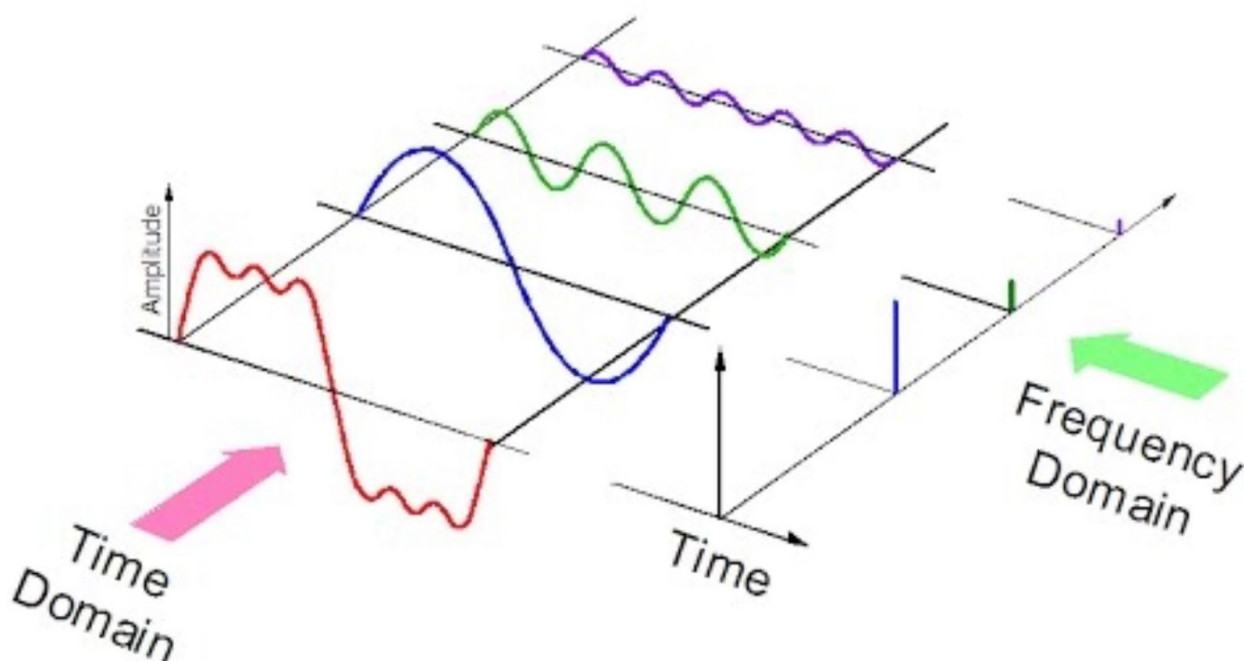


Figure 1: Fourier Transformation

It is used to achieve MFCCs:

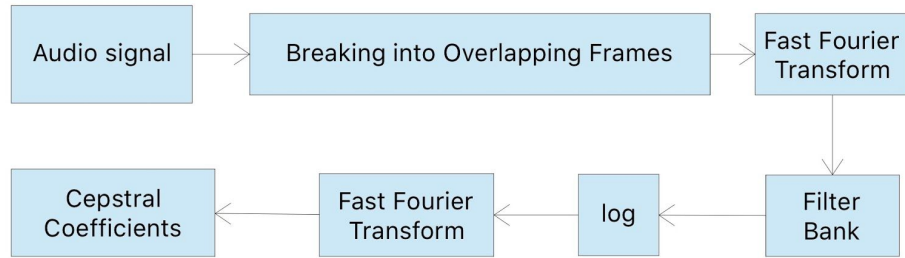


Figure 2: MFCC archiving

For the sake of simplicity, we can view the overlapping frames and filter bank as filters. We also utilize spectral centroid, which describes the “brightness” of the sound, and the spectral contrast, which measure the “level difference between the peak and valleys in the spectrum”(Nogueira et al., Optimization of a Spectral Contrast Enhancement Algorithm for Cochlear Implants Based on a Vowel Identification Model). Therefore, we use MFCCs, spectral centroid and spectral contrast as the data in frequency domain for our network to analyze.

Experimental Setup

Data for Training and Testing:

To train the tagging algorithm, we split the data into training, validation, and test subsets. In order to format and load the data to the tagging algorithm, we created two functions to load and split the audio files. We then created a MedleyDataset class to transform the data into a format that Dataloader and LSTM layer can process.

Once we loaded and transformed the data, we ran the training subset through the tagging algorithm. At first the algorithm had a small accuracy rating, so we re-ran the first data transformation through librosa to extract MFCC, spectral centroid and spectral contrast for better results.

We judged the performance of our network by the overall accuracy and accuracy of each class of the prediction on validation set, and the time used to achieve that.

We used mini batches to train our network, since we found out from exam 1 that it is a compromise between time and accuracy. As our training set size was 5841, we experimented around batch sizes of 30, 50, and 100. 100 seems to have a better result when we are doing the analysis in time domain. Hence, we use the same batch size on the analysis in frequency domain.

Results

Time Domain

When we first ran the algorithm, the algorithm began with a loss of 1.5 and ended with a loss around 0.01. After running the model multiple times, the loss (or performance index) decreased significantly. Below is a graph of loss over training time:

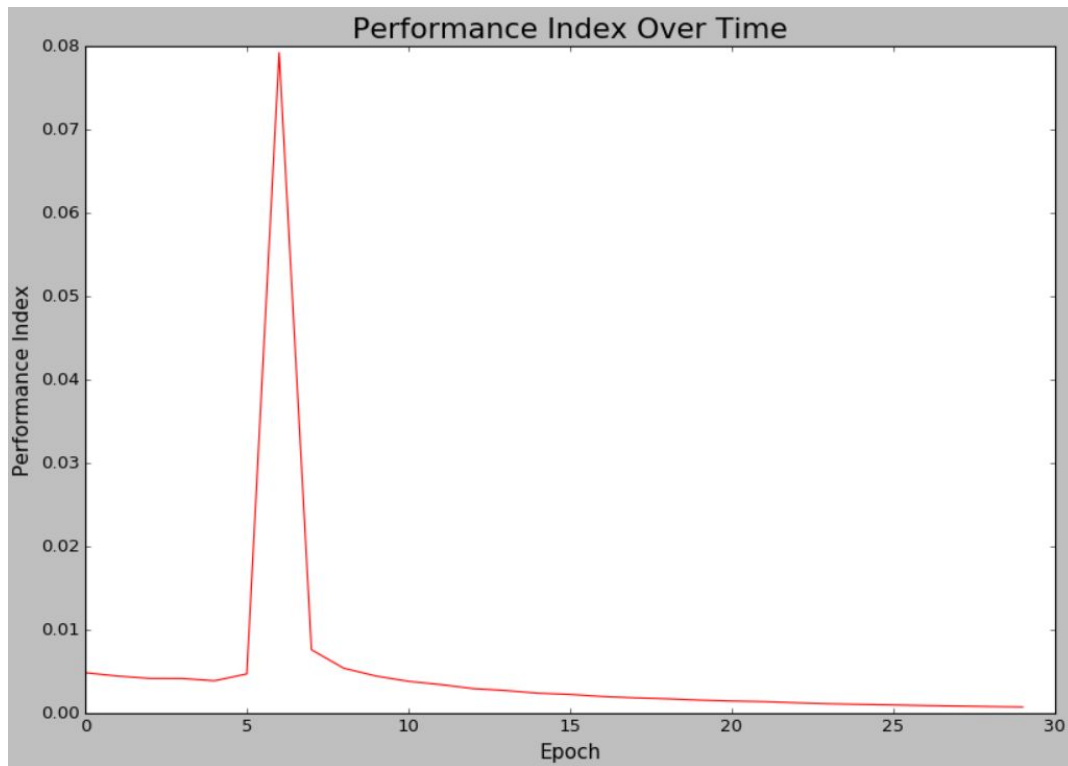


Figure 3: Loss over time

Based on the above graph, the loss started around 0.005 and ended near 0.00, or no error. However, there's a spike in loss value between the fifth and eighth epoch, with loss almost 20 times that of previous epochs.

However, based on the accuracy function, the algorithm has an accuracy rating of 24%. Further, the confusion matrix is filled with 0s:

```
[0. 0. 0. 0. 0. 0. 0. 0.]
[0. 0. 0. 0. 0. 0. 0. 0.]
[0. 0. 0. 0. 0. 0. 0. 0.]
[0. 0. 0. 0. 0. 0. 0. 0.]
[0. 0. 0. 0. 0. 0. 0. 0.]
[0. 0. 0. 0. 0. 0. 0. 0.]
[0. 0. 0. 0. 0. 0. 0. 0.]
[0. 0. 0. 0. 0. 0. 0. 0.]
```

Figure 4: Confusion matrix

Given the low loss value during training, it seems odd that the algorithm would have a low accuracy rate and a confusion matrix filled with 0s. There is either an issue in our algorithm or an issue in our accuracy code, or possibly an issue with both.

Frequency Domain

When we were analyzing the time domain, we got extremely low loss (almost 0s) every epoch, but the overall accuracy hovered around 30%. Therefore, we believe there are three problems:

1. The train set is relative small
2. The information we extract is not very useful
3. The program might be over or under fitting.

We tried to adjust the dropout rate, learning rate and the number of neurons to prevent over or under fitting, and improved the second problem with the analysis in frequency domain. We also tried to redivide the size of train set to make it bigger, but it doesn't seem to have a significant impact on the time domain data.

Hence we started the analysis in the frequency domain.

The analysis in frequency domain gets much better results. It partially confirms that the reason why we have such low accuracy in time domain analysis is because we didn't extract enough useful data for the network to train.

While training the network, the loss starts from 1.39, and decreased to 0.19 over 50 epochs and batch number 100. The overall accuracy is 78% . We acquired the loss of training time image, which seems to have a clear trend of decreasing:

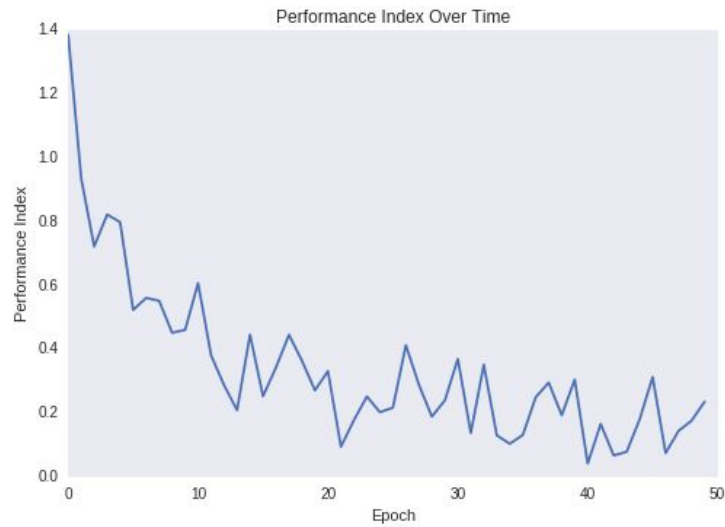


Figure 5-1. Epoch 50, Batch Size 100, Learning Rate 0.0001, Dropout 0.1, Adam, LSTM Layer 2

The confusion matrix (as rows represent the prediction, while columns represent the actual) is as follows:

Accuracy of the network on the 3494 validation audio clips: 78 %
 Accuracy of clarinet : 48 %
 Accuracy of distorted electric guitar : 83 %
 Accuracy of female singer : 75 %
 Accuracy of flute : 30 %
 Accuracy of piano : 89 %
 Accuracy of tenor saxophone : 24 %
 Accuracy of trumpet : 84 %
 Accuracy of violin : 86 %

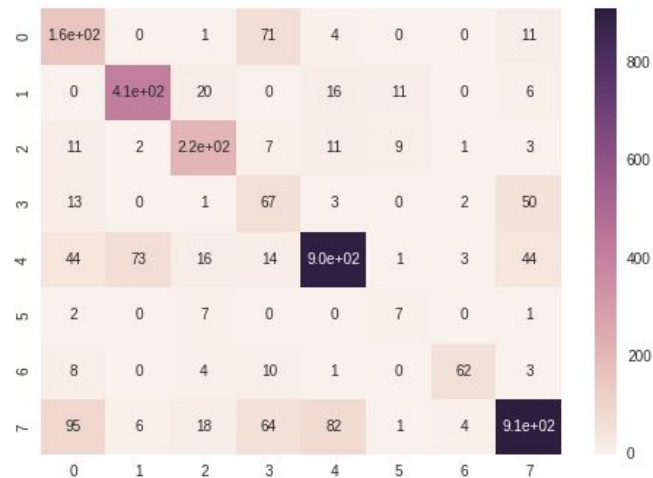


Figure 5-2. Epoch 50, Batch Size 100, Learning Rate 0.0001, Dropout 0.1, Adam, LSTM Layer 2

We can see that the confusion matrix confirms the lower category accuracy on flute, and saxophone. The network seem to be easily confuse flute with clarinet, and saxophone with guitar

or female singer. This could result from the fact that the frequency of those pairs are indeed quite close, or it could also come from the fact that the number of clips on those categories are much smaller than, say, the piano and violin clips. For better results, other data are possibly needed. To get the optimal training result from my network, we are using this set of training parameters as my base of comparison (epoch: 50, batch size: 100, learning : 0.0001, drop out: 0.1, LSTMs layer: 2), and then adjusted the different parameters to see how the code performs.

Batchsize:

Using the same epochs, we adjusted the batch size to 50, and get the following result:

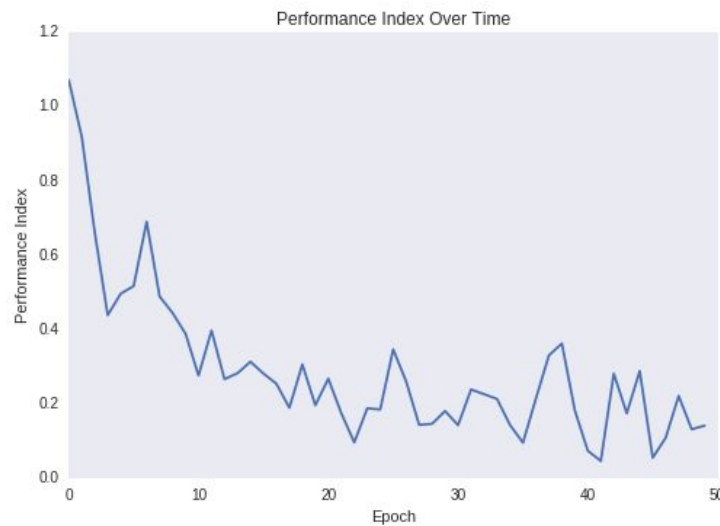


Figure 6-1. Epoch 50, Batch Size 50, Learning Rate 0.0001, Dropout 0.1, Adam, LSTM Layer 2

Accuracy of the network on the 3494 validation audio clips: 80 %
 Accuracy of clarinet : 52 %
 Accuracy of distorted electric guitar : 84 %
 Accuracy of female singer : 73 %
 Accuracy of flute : 33 %
 Accuracy of piano : 88 %
 Accuracy of tenor saxophone : 24 %
 Accuracy of trumpet : 84 %
 Accuracy of violin : 90 %

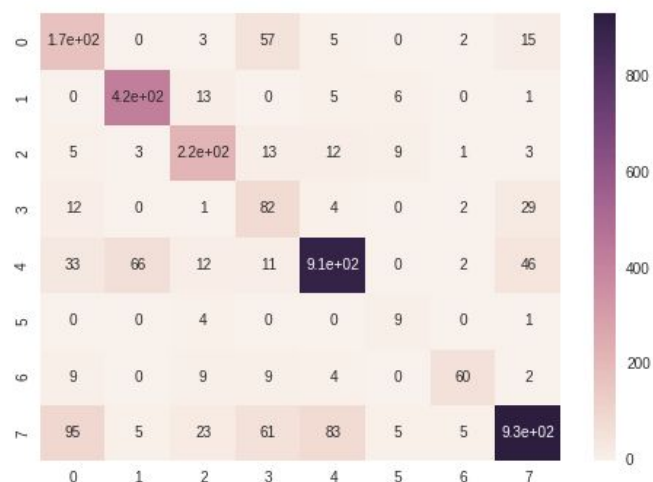


Figure 6-2. Epoch 50, Batch Size 50, Learning Rate 0.0001, Dropout 0.1, Adam, LSTM Layer 2

Then we adjusted the batch size to 20 and get:

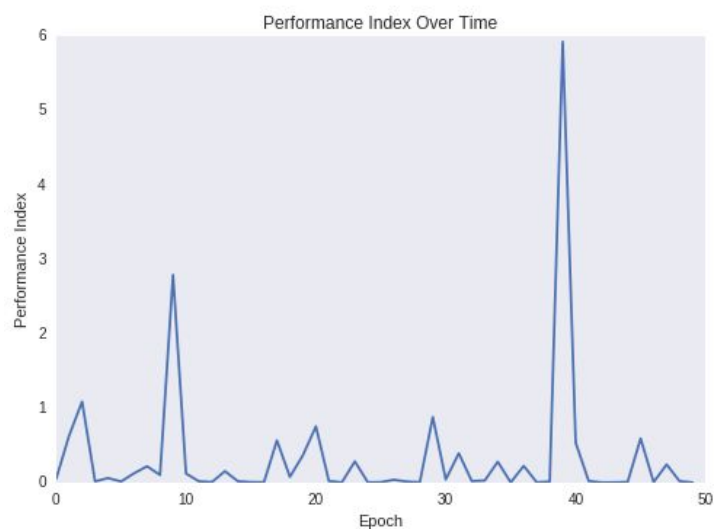


Figure 7-1. Epoch 50, Batch Size 20, Learning Rate 0.0001, Dropout 0.1, Adam, LSTM Layer 2

Accuracy of the network on the 3494 validation audio clips: 78 %
 Accuracy of clarinet : 44 %
 Accuracy of distorted electric guitar : 71 %
 Accuracy of female singer : 75 %
 Accuracy of flute : 29 %
 Accuracy of piano : 89 %
 Accuracy of tenor saxophone : 17 %
 Accuracy of trumpet : 77 %
 Accuracy of violin : 92 %

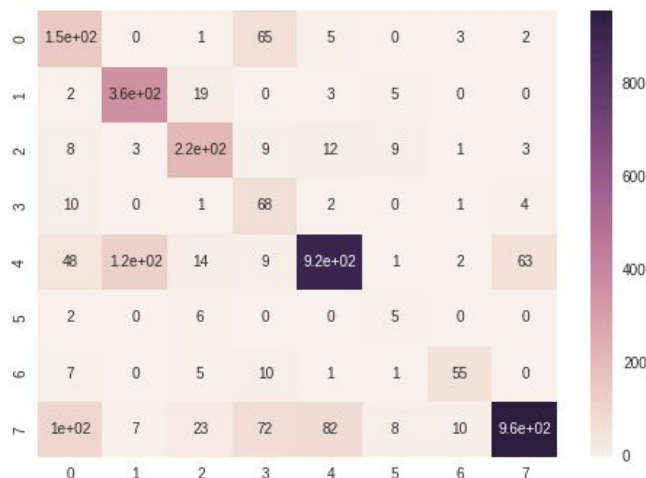


Figure 7-2. Epoch 50, Batch Size 20, Learning Rate 0.0001, Dropout 0.1, Adam, LSTM Layer 2

The performance index behaves very strangely when batch size is 20, it seems to be constantly very low, however, there are a few pikes occurring, the biggest one happens around epoch 10 and 40. It could be due to the fact that the batch size is too low, and therefore it affects the training of the network, as the calculation of the gradient is closer to the individual file when batch size is smaller. The performance index of batch size 50 is quite similar to 100, with a downward trend. It seems that the accuracy peaks in the middle, where the batch size equals 50, although the difference is not that significant. It could be due to the fact that while using mini-batch, we are compromising between stochastic and gradient descent, and there would be an optimal point in the middle. We suppose 50 is closer to that optimal point.

Learning rate:

We are now changing the batch size back to 100, and trying to adjust the learning rate from 0.001. The training loss changes from 0.88 to 0.40 over 50 epochs. We can see the loss hovers over that number, this suggests that the learning rate is too high, as a result, every step the weight and bias getting updated, they change way too much.

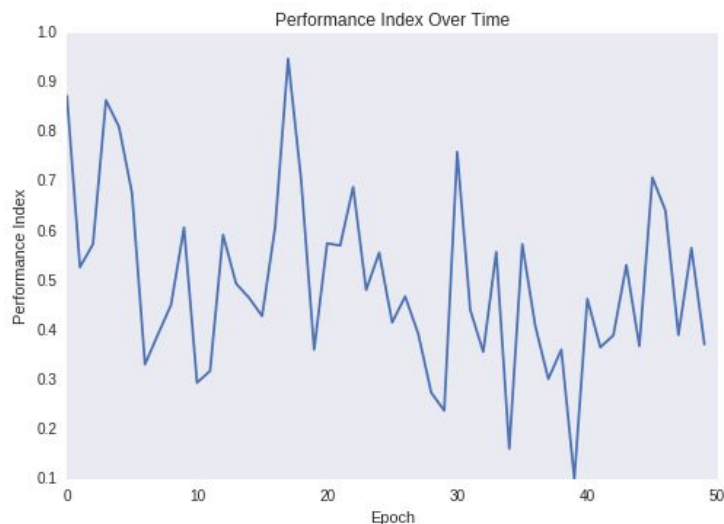


Figure 8-1. Epoch 50, Batch Size 100, Learning Rate 0.001, Dropout 0.1, Adam, LSTM Layer 2

Accuracy of the network on the 3494 validation audio clips: 75 %
 Accuracy of clarinet : 35 %
 Accuracy of distorted electric guitar : 79 %
 Accuracy of female singer : 77 %
 Accuracy of flute : 14 %
 Accuracy of piano : 86 %
 Accuracy of tenor saxophone : 17 %
 Accuracy of trumpet : 69 %
 Accuracy of violin : 93 %

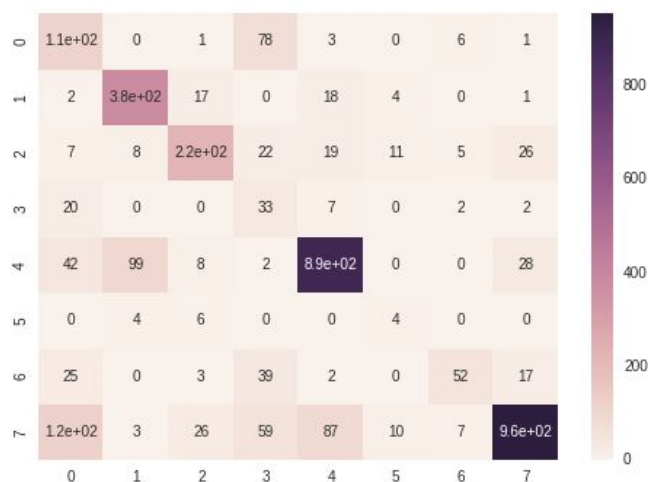


Figure 8-2. Epoch 50, Batch Size 100, Learning Rate 0.001, Dropout 0.1, Adam, LSTM Layer 2

The overall accuracy is nicely at 75%, but it could be random. Now we adjust the learning rate to 0.00001. The following result suggests the overall accuracy is at 67%, while it's actually a little bit worse than the result we acquire from the learning rate 0.001, and certainly not as good as the learning rate 0.0001. The performance index seem to enter a rapid oscillation after epoch 10 compare to the result of 0.001.

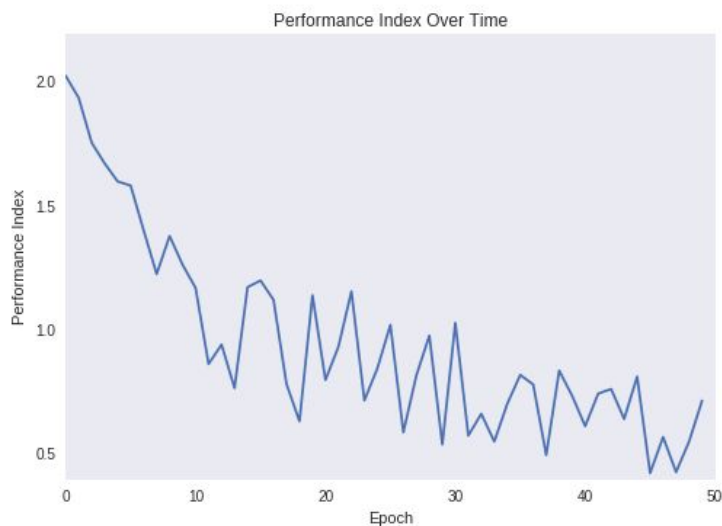


Figure 9-2. Epoch 50, Batch Size 100, Learning Rate 0.00001, Dropout 0.1, Adam, LSTM Layer 2

Accuracy of the network on the 3494 validation audio clips: 67 %
 Accuracy of clarinet : 11 %
 Accuracy of distorted electric guitar : 58 %
 Accuracy of female singer : 45 %
 Accuracy of flute : 0 %
 Accuracy of piano : 88 %
 Accuracy of tenor saxophone : 0 %
 Accuracy of trumpet : 13 %
 Accuracy of violin : 95 %

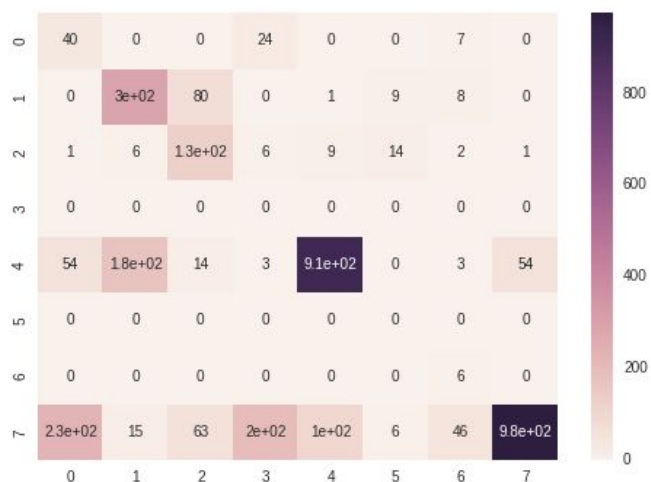


Figure 9-2. Epoch 50, Batch Size 100, Learning Rate 0.00001, Dropout 0.1, Adam, LSTM Layer 2

We can see that the learning too fast or too slow have an impact on our result; learning too fast, then there is a problem with over fitting, the loss hover around a range, and has trouble to decrease; learning too slow, it doesn't reach the result as fast.

Dropout:

Up to this point we have been using the dropout rate 0.1, and now we change it to 0.05. The result is as follows:



Figure 10-1. Epoch 50, Batch Size 100, Learning Rate 0.0001, Dropout 0.05, Adam, LSTM Layer 2

Accuracy of the network on the 3494 validation audio clips: 75 %
 Accuracy of clarinet : 39 %
 Accuracy of distorted electric guitar : 76 %
 Accuracy of female singer : 61 %
 Accuracy of flute : 27 %
 Accuracy of piano : 85 %
 Accuracy of tenor saxophone : 17 %
 Accuracy of trumpet : 76 %
 Accuracy of violin : 94 %

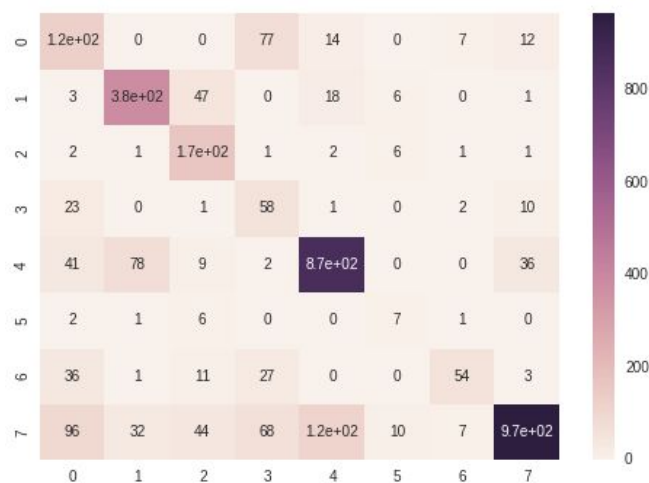


Figure 10-2. Epoch 50, Batch Size 100, Learning Rate 0.0001, Dropout 0.05, Adam, LSTM Layer 2

We can see from the confusion matrix that the code seems to mistaken the pair of flute and clarinet, saxophone and guitar/female singer, also it seems that the network confuses clarinet with piano sometimes. This agrees with the lack of significant trend in performance index. But the overall accuracy is quite good, at 75%. Then we have the results from changing the dropout rate to 0.5:

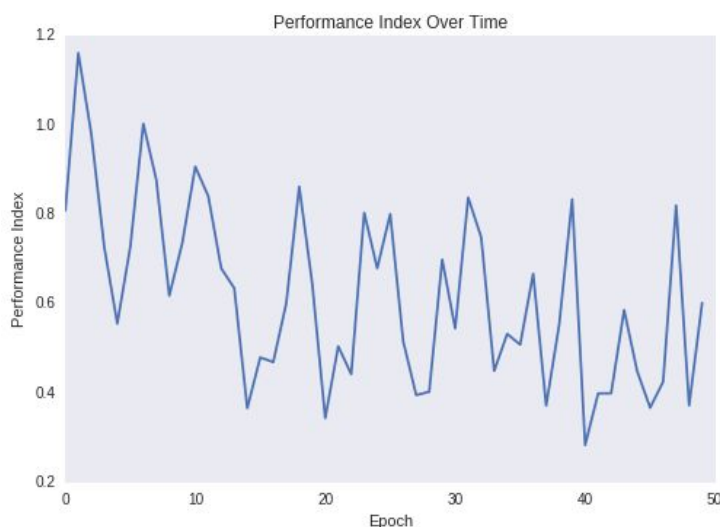


Figure 11-1. Epoch 50, Batch Size 100, Learning Rate 0.0001, Dropout 0.5, Adam, LSTM Layer 2

Accuracy of the network on the 3494 validation audio clips: 70 %
 Accuracy of clarinet : 23 %
 Accuracy of distorted electric guitar : 68 %
 Accuracy of female singer : 50 %
 Accuracy of flute : 15 %
 Accuracy of piano : 86 %
 Accuracy of tenor saxophone : 3 %
 Accuracy of trumpet : 59 %
 Accuracy of violin : 88 %

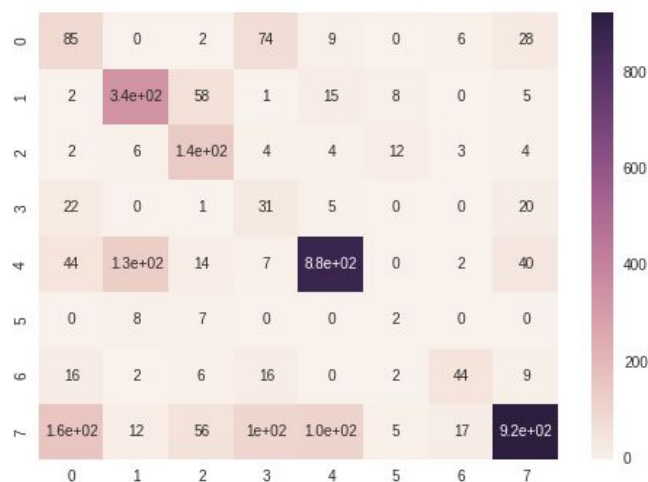


Figure 11-2. Epoch 50, Batch Size 100, Learning Rate 0.0001, Dropout 0.5, Adam, LSTM Layer 2

We see a slight decrease in overall accuracy, at 70%, but the network behaves badly not only on flute and saxophone, but also several other classes. The only 2 classes having a good result is piano and violin. Both of the dropout rate works worse than our dropout at 0.1. And if we take a look at the performance index images, we can see that there is no significant trend of loss throughout time. This might due to over or underfitting.

Optimizer:

We are currently using Adam as my optimizer. Switching to SGD gives the result:

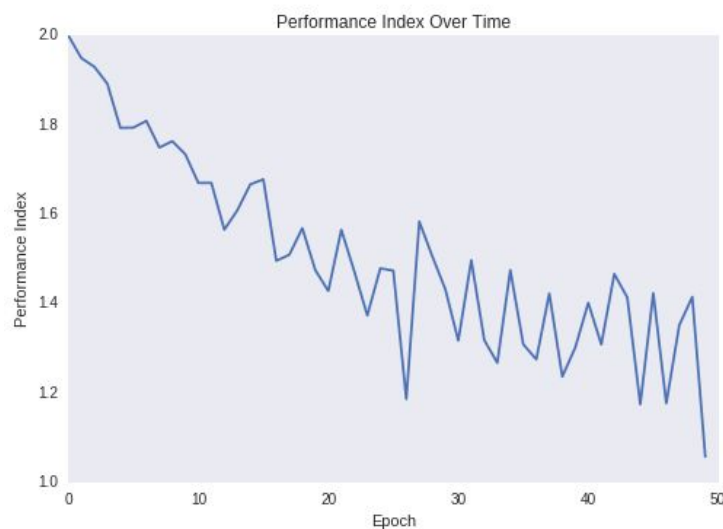


Figure 12-1. Epoch 50, Batch Size 100, Learning Rate 0.0001, Dropout 0.1, SGD, LSTM Layer 2

Accuracy of the network on the 3494 validation audio clips: 52 %
 Accuracy of clarinet : 0 %
 Accuracy of distorted electric guitar : 0 %
 Accuracy of female singer : 0 %
 Accuracy of flute : 0 %
 Accuracy of piano : 87 %
 Accuracy of tenor saxophone : 0 %
 Accuracy of trumpet : 0 %
 Accuracy of violin : 90 %

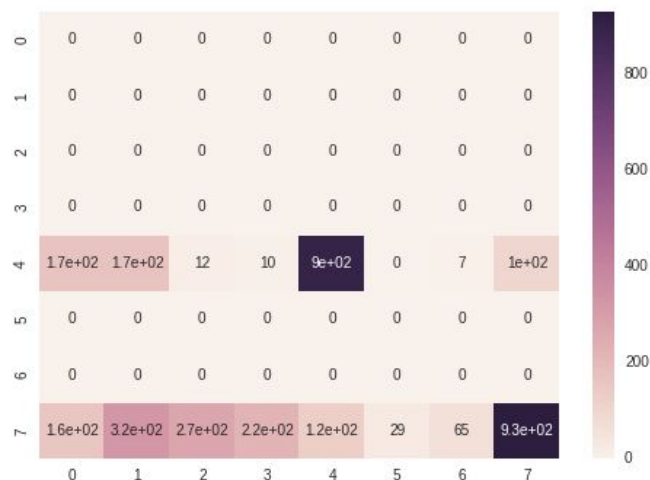


Figure 12-2. Epoch 50, Batch Size 100, Learning Rate 0.0001, Dropout 0.1, SGD, LSTM Layer 2

The overall accuracy is about 52%, and the confusion matrix suggests that the category of piano and violin are most likely predicted. The prediction result seems bad, however there is a clear trend on the loss over time, it just vibrates at the end of the epochs. Adadelata optimizer with Rhode = 0.8 and eps = 1e-6 give:

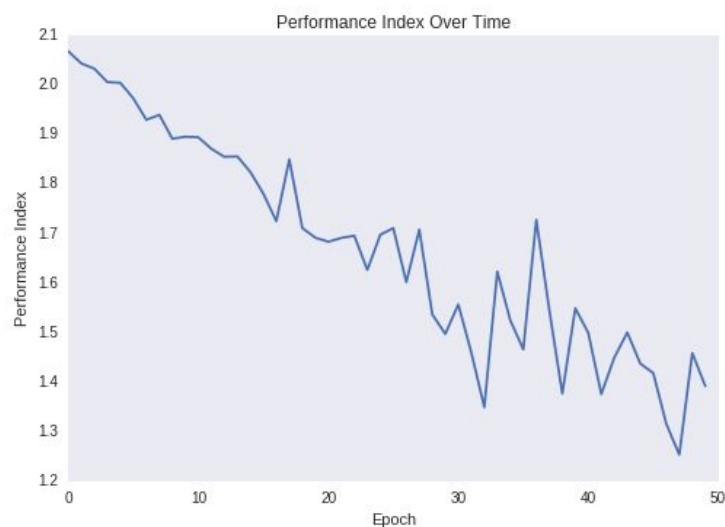


Figure 13-1. Epoch 50, Batch Size 100, Learning Rate 0.0001, Dropout 0.1, Adadelata, LSTM Layer 2

Accuracy of the network on the 3494 validation audio clips: 49 %
 Accuracy of clarinet : 0 %
 Accuracy of distorted electric guitar : 0 %
 Accuracy of female singer : 0 %
 Accuracy of flute : 0 %
 Accuracy of piano : 93 %
 Accuracy of tenor saxophone : 0 %
 Accuracy of trumpet : 0 %
 Accuracy of violin : 75 %

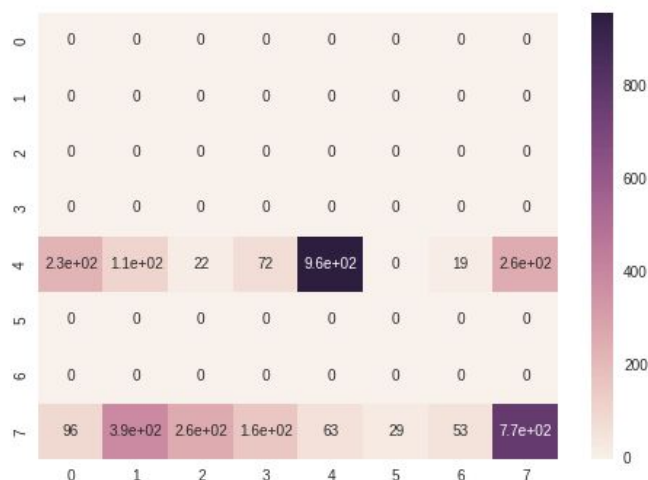


Figure 13-2. Epoch 50, Batch Size 100, Learning Rate 0.0001, Dropout 0.1, Adadelata, LSTM Layer 2

It produces 49% of overall accuracy, which is quite close to SGD optimizer. Still the network seems to think everything is either piano or violin, although we have a clear trend of loss over time. We can see from the results that Adam outperforms SGD and Adadelata.

Extra Layer of LSTMs:

Now we are adding another layer of LSTMs in the nn module. The result is as follows:

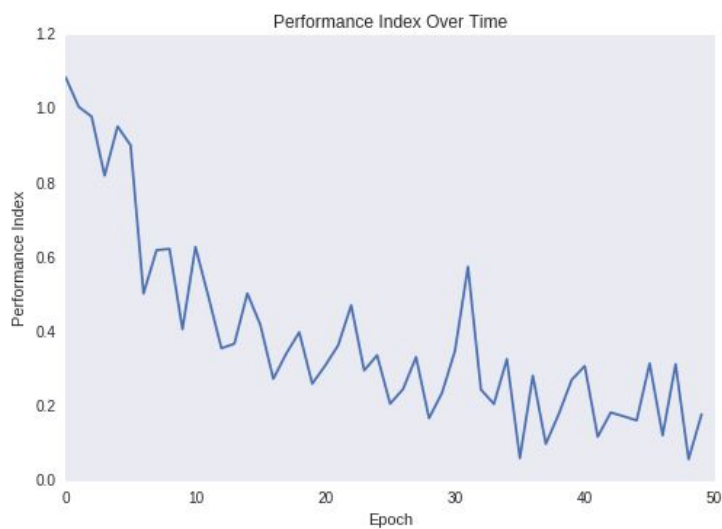


Figure 14-1. Epoch 50, Batch Size 100, Learning Rate 0.0001, Dropout 0.1, Adam, LSTM Layer 2+1

Accuracy of the network on the 3494 validation audio clips: 79 %
 Accuracy of clarinet : 45 %
 Accuracy of distorted electric guitar : 92 %
 Accuracy of female singer : 74 %
 Accuracy of flute : 30 %
 Accuracy of piano : 88 %
 Accuracy of tenor saxophone : 55 %
 Accuracy of trumpet : 79 %
 Accuracy of violin : 91 %

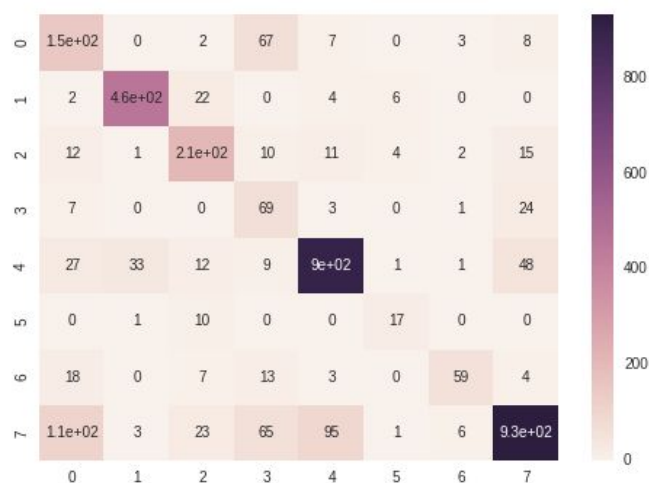


Figure 14-2. Epoch 50, Batch Size 100, Learning Rate 0.0001, Dropout 0.1, Adam, LSTM Layer 2+1

From the result we can see that the extra layer of LSTMs has an overall accuracy of 79%. The performance index oscillates similarly to our comparison parameters. The addition of the LSTMs layer does not change the result much. We now use the previously tested optimal training parameters: batch size = 50, training rate = 0.0001, dropout = 0.1, optimizer = Adam, to predict the test set, and get the result as follows:

Accuracy of the network on the 3494 validation audio clips: 53 %
 Accuracy of clarinet : 29 %
 Accuracy of distorted electric guitar : 83 %
 Accuracy of female singer : 69 %
 Accuracy of flute : 6 %
 Accuracy of piano : 98 %
 Accuracy of tenor saxophone : 6 %
 Accuracy of trumpet : 50 %
 Accuracy of violin : 56 %

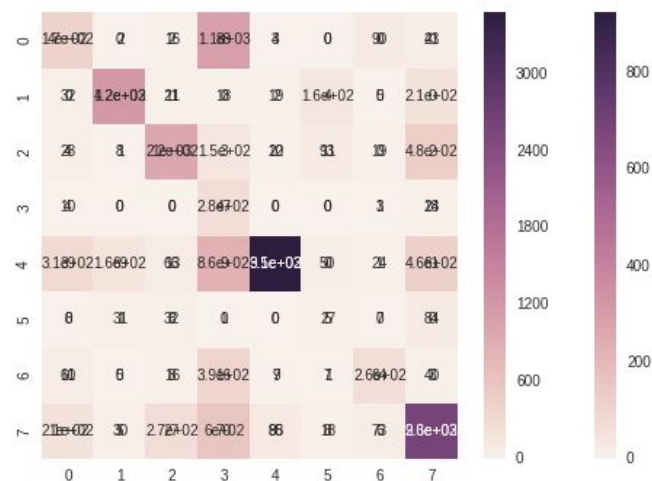


Figure 15. Epoch 50, Batch Size 50, Learning Rate 0.0001, Dropout 0.1, Adam, LSTM Layer 2; Test Set

We can see that the overall accuracy for the test set is 53%, not as high as the validation set. Again, from the confusion matrix (noted that it's in the scientific mode) we know the major problem is that the network has trouble distinguishing the flute and clarinet pair, and saxophone and female singer pair.

Summary and Conclusions

To better understand time-series data and modules, we used Medley-solos audio data and pytorch LSTM layers. To prepare the data, we transformed the audio files into numpy arrays and loaded the arrays as a torch dataset. After extracting, transforming, and loading the data, we set up the parameters for training the time domain and frequency domain. We then adjusted parameters, including size of training dataset and learning rate, to increase accuracy rate.

From the result, we can see that the change of the training parameters changes the performance of the network drastically. We need to find the optimal set of parameters to prevent issues like over or underfitting, learning too fast or slow, and find compromise between accuracy and running time.

We didn't get great results on the test set. This is majorly due to the fact that the network keeps confuse flute with clarinet, and saxophone with female singer or guitar. Since this behavior appears in all of the results of validation set as well. We think a better set of useful data extraction, or a more balance dividing of clips among the classes is needed.

We could also extract more features using librosa, for instance, using librosa.chrome could be helpful to make the prediction more accurate. Using a bigger sized train data on frequency domain analysis is also a way to train the network better.

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Appendix

Time domain analysis:

Main(time).py:

```

1  import numpy as np
2  import pandas as pd
3  from scipy.io import wavfile
4  from torch.autograd import Variable
5  import time
6  import IPython.display as ipd
7  import matplotlib.pyplot as plt
8  import os, glob
9
10 import torch
11 import torch.nn as nn
12 from torch.utils.data import Dataset, DataLoader
13 import torch.nn.functional as F
14 import torch.optim as optim
15 import torchvision.transforms as transforms
16 from sklearn.metrics import confusion_matrix
17
18 #%%
19 # Initial data sizes
20 input_size = 131072      #Number of inputs (splited because of the LSTM model)
21 hidden_size1 = 500      #Number of neurons
22 hidden_size2 = 30      #Number of neurons
23 num_classes = 8        #8 classes/labels
24 num_epochs = 30        #Number of epochs
25 batch_size = 100       #Number of audio clips to ran through 1 iteration
26 learning_rate = 0.001
27 confusion_m = np.zeros((8,8))
28
29 #%%
30 #Classes
31
32 classes = ['clarinet', 'distorted electric guitar', 'female singer', 'flute',
33           'piano', 'tenor saxophone', 'trumpet', 'violin']
34
35 #%%

```

```

36 #Spit data into test, train, validation sets
37
38 print("Loading CSV...")
39 Medley = pd.read_csv("Medley-solos-DB_metadata.csv")
40
41 train = Medley.iloc[12236:18077]
42 train.index = range(len(train.index))
43
44 validation = Medley.iloc[18077:]
45 validation.index = range(len(validation.index))
46
47 test = Medley.iloc[0:12236]
48 test.index = range(len(test.index))
49
50 print("Number of audios=", Medley.shape[0], " Number of classes=", len(Medley.instrument.unique()))
51 print()
52 print(Medley.instrument.unique())
53
54
55 print("Train set size: ", len(train))
56 print("Validation set size: ", len(validation))
57 print("Test set size: ", len(test))
58
59 ###
60 #Load audio file linked to the uuid
61 def full_name(file):
62     correspding_row = Medley.loc[Medley['uuid4'] == file].iloc[0]
63     subset = str(correspding_row.loc['subset'])
64     instrument_id = str(correspding_row.loc['instrument_id'])
65     parts = ['Medley-solos-DB_', str(subset), '-', str(instrument_id), '_', file, '.wav.wav']
66     s = ''
67     file_name = s.join(parts)
68     return file_name
69
70 def load_file(file):
71     file_name = full_name(file)
72     path = '/home/ubuntu/Final-Project-Group1/Medley-solos-DB/'
73     parts = [path, file_name]

```



```

74     s = ''
75     link = s.join(parts)
76     return link
77
78     #%%
79
80     #%%
81     #Process Dataset
82
83     class MedleyDataset(Dataset):
84         def __init__(self, dataset_csv, transform=None):
85             self.dataset_frame = dataset_csv
86             self.transform = transform
87
88         def __len__(self):
89             return len(self.dataset_frame)
90
91         def __getitem__(self, index):
92             uuid4 = self.dataset_frame.iloc[index, 4]
93             instrument_list = self.dataset_frame.iloc[index, 2]
94             instrument_id = int(instrument_list)
95             link = load_file(uuid4)
96
97             fs, audio = wavfile.read(link)
98             audio = audio.astype('float')
99
100             sample = {'audio': audio, 'label': instrument_id}
101
102             if self.transform:
103                 sample = self.transform(sample)
104
105             return sample
106
107         #%%
108         transform = transforms.Compose([transforms.ToTensor(),
109                                         transforms.Normalize((0.5,), (0.5,))])
110         #%%
111
112     audio_dataset_train = MedleyDataset(dataset_csv= train)
113     audio_dataset_validation = MedleyDataset(dataset_csv= validation)

```

```

113 audio_dataset_test = MedleyDataset(dataset_csv= test)
114
115 #%%
116 #LSTM Model
117
118 class Net(nn.Module):
119     def __init__(self, input_size, hidden_size1, hidden_size2, num_classes):
120         super(Net, self).__init__()
121         self.LSTM1 = nn.LSTM(input_size, hidden_size1, batch_first = True, num_layers = 2, dropout = 0.1).cuda()
122         self.LSTM2 = nn.LSTM(hidden_size1, hidden_size2, batch_first = True).cuda()
123         self.lstm2tag = nn.Linear(hidden_size2, num_classes).cuda()
124
125     def forward(self, x):
126         #print(x.shape)
127         s = x.shape[0]
128         #print(s)
129         x = x.reshape(s, 1, input_size)
130         out, states = self.LSTM1(x).cuda()
131         out, states = self.LSTM2(out).cuda()
132         #print("Afterlstm")
133         #print(out.shape)
134         out = out[:,0,:]
135         out = self.lstm2tag(out)
136         return out
137
138 #%%
139 #Train LSTM
140
141 model = Net(input_size, hidden_size1, hidden_size2, num_classes)
142
143 #%%
144 criterion = nn.CrossEntropyLoss()
145
146 #optimizer = torch.optim.Adadelta(model.parameters(), rho = 0.8, eps = 1e-6, lr=learning_rate)
147 #optimizer = torch.optim.SGD(params = model.parameters(), lr=learning_rate, momentum=0, dampening=0, weight_decay=0, nesterov=F
148 optimizer = torch.optim.Adam(params = model.parameters(), lr=learning_rate, betas=(0.9, 0.999), eps=1e-08, weight_decay=0, amsg

```

```

150 train_loader = torch.utils.data.DataLoader(dataset = audio_dataset_train, batch_size = batch_size, shuffle = True)
151 validation_loader = torch.utils.data.DataLoader(dataset = audio_dataset_validation, batch_size = batch_size, shuffle = False)
152 test_loader = torch.utils.data.DataLoader(dataset = audio_dataset_test, batch_size = batch_size, shuffle = False)
153
154 ###
155 start_time = time.time()
156
157 device = torch.device("cuda:0" if torch.cuda.is_available() else "cpu")
158
159 epochs = np.array([])
160 loss_index = np.array([])
161
162 for epoch in range(num_epochs):
163     for i, data in enumerate(test_loader):
164
165         model.zero_grad()
166
167         audios = data['audio']
168         labels = data['label']
169
170         audios = audios.type(torch.FloatTensor)
171         audios = Variable(audios.cuda())
172
173         output = model(audios)
174
175         labels = labels.type(torch.LongTensor)
176         labels = Variable(labels.cuda())
177
178         loss = criterion(output, labels).cuda()
179         loss.backward()
180         optimizer.step()
181         optimizer.zero_grad()
182
183         if (i + 1) % 10 == 0:
184             print('Epoch [%d/%d], Step [%d/%d], Loss: %.4f'
185                   % (epoch + 1, num_epochs, i + 1, len(test) // batch_size, loss.data[0]))
186
187     epochs = np.append(epochs, epoch)

```

```

188     loss_index = np.append(loss_index, loss.item())
189
190     end_time = time.time()
191     elapsed_time = end_time - start_time
192     print("Elapsed time:", elapsed_time)
193
194     ###
195
196     plt.figure(figsize = (12,8))
197     plt.plot(epochs, loss_index, 'r')
198     plt.xlabel("Epoch", fontsize = 14)
199     plt.ylabel("Performance Index", fontsize = 14)
200     plt.title("Performance Index Over Time", fontsize = 20)
201     plt.show()
202
203     ###
204     #Test accuracy on validation set
205
206     correct = 0
207     total = 0
208
209     for i, data in enumerate(validation_loader):
210         audios = data['audio']
211         labels = data['label']
212
213         audios = audios.type(torch.FloatTensor)
214         audios = Variable(audios.cuda())
215
216         outputs = model(audios)
217
218         labels = labels.type(torch.LongTensor)
219         labels = Variable(labels.cuda())
220
221         _, predicted = torch.max(outputs.data, 1)
222
223         total += labels.size(0)
224         #correct += (predicted == labels).sum()
225         for j in range(len(predicted)):

```

```

226         if predicted[j] == labels[j]: correct += 1
227     # print(total)
228     # print(correct)
229
230
231     #Confusion matrix
232     results = confusion_matrix(labels, predicted)
233     confusion_m = confusion_m + results
234
235     print(confusion_m)
236     print('Accuracy of the network on the 3494 validation audio clips: %d %%' % (100 * correct / total))
237
238     ###
239     #Test accuracy of each class on Validation set
240
241     class_correct = list(0. for i in range(8))
242     class_total = list(0. for i in range(8))
243
244     for data in test_loader:
245         audios = data['audio']
246         labels = data['label']
247
248         audios = audios.type(torch.FloatTensor)
249         audios = Variable(audios.cuda())
250
251         outputs = model(audios)
252
253         labels = labels.type(torch.LongTensor)
254         labels = Variable(labels.cuda())
255
256         _, predicted = torch.max(outputs.data, 1)
257
258         labels = labels.cpu().numpy()
259         c = (predicted.cpu().numpy() == labels)
260         s = labels.shape[0]
261         for i in range(s):
262             label = labels[i]
263             class_correct[label] += c[i]
264             class_total[label] += 1
265
266             class_correct[label] += c[i]
267             class_total[label] += 1
268
269     ###
270
271     for i in range(8):
272         if (class_total[i] == 0):
273             print('Accuracy of %5s : %2d %%' % (classes[i], 0))
274         else:
275             print('Accuracy of %5s : %2d %%' % (classes[i], 100 * class_correct[i] / class_total[i]))
276
277     ###
278     torch.save(model.state_dict(), 'model_time.pkl')

```

Frequency domain analysis:

music_feature_loader:

```

1  #This file extracts the music feature (mfcc, spectral centroid, spectral contrast) using librosa, and save it as.npy numpy
2  #-----
3  #Load audio file linked to the uuid
4  def full_name(file):
5      correspding_row = Medley.loc[Medley['uuid4'] == file].iloc[0]
6      subset = str(correspding_row.loc['subset'])
7      instrument_id = str(correspding_row.loc['instrument_id'])
8      parts = ['Medley-solos-DB_', str(subset), '-', str(instrument_id), '_', file, '.wav.wav']
9      s = ''
10     file_name = s.join(parts)
11     return file_name
12
13 def find_file(file):
14     file_name = full_name(file)
15     path = '/home/ubuntu/Machine-Learning/Medley-solos-DB/'
16     parts = [path, file_name]
17     s = ''
18     link = s.join(parts)
19     return link
20
21 #-----
22 #Generating music data ndarray in frequency domain using librosa
23
24 def get_music_features(dataset):
25     timeseries_length = 3
26     audio = np.zeros((len(dataset), timeseries_length, 21), dtype=np.float64)
27
28     for i in range(len(dataset)):
29         row = dataset.loc[i]
30         uuid4_name = str(row.loc['uuid4'])
31         link = find_file(uuid4_name)
32
33         y, sr = librosa.load(link)
34
35         mfcc = librosa.feature.mfcc(y=y, sr=sr, n_mfcc=13)
36         spectral_center = librosa.feature.spectral_centroid(y=y, sr=sr)
37         spectral_contrast = librosa.feature.spectral_contrast(y=y, sr=sr)
38
39         audio[i, :, 0:13] = mfcc.T[0:timeseries_length, :]
40         audio[i, :, 13:14] = spectral_center.T[0:timeseries_length, :]
41         audio[i, :, 14:21] = spectral_contrast.T[0:timeseries_length, :]
42
43
44         if ((i + 1) % 100 == 0):
45             print("Extracted features audio clip %i of %i." % (i + 1, len(dataset)))
46
47     return audio
48
49 train_audio_data = get_music_features(train)
50 validation_audio_data = get_music_features(validation)
51 test_audio_data = get_music_features(test)
52
53 np.save('train_audio_data.npy', train_audio_data)
54 np.save('validation_audio_data.npy', validation_audio_data)
55 np.save('test_audio_data.npy', test_audio_data)

```

Main(Frequency):

```

1  #The commented part is the same as the music_feature_loader
2  #Need to run music_feature_loader first to generate the 3 .npy file, to run this code.
3
4
5  import numpy as np
6  import pandas as pd
7  from torch.autograd import Variable
8  import time
9  import librosa
10
11  import torch
12  import torch.nn as nn
13  from torch.utils.data import Dataset, DataLoader
14
15  import matplotlib
16  import matplotlib.pyplot as plt
17  import seaborn as sns; sns.set()
18
19  device = torch.device("cuda:0" if torch.cuda.is_available() else "cpu")
20
21  # -----
22  # Initial data sizes
23  input_size = 63 # Number of inputs (splited because of the LSTM model) using Librosa
24  hidden_size1 = 500 # Number of neurons
25  hidden_size2 = 30 # Number of neurons
26  num_classes = 8 # 8 classes/labels
27  num_epochs = 300 # Number of epochs
28  batch_size = 100 # Number of audio clips to ran through 1 iteration
29  learning_rate = 0.001
30  confusion_m = np.zeros((8, 8))
31  # -----
32  # Classes
33
34  classes = ['clarinet', 'distorted electric guitar', 'female singer', 'flute', 'piano', 'tenor saxophone', 'trumpet',
35            'violin']
36
37  # -----
38  # Spit data into test, train, validation sets
39
40  print("Loading CSV...")

```

```

41 Medley = pd.read_csv("Medley-solos-DB_metadata.csv")
42
43 train = Medley.iloc[12236:18077]
44 train.index = range(len(train.index))
45
46 validation = Medley.iloc[18077:]
47 validation.index = range(len(validation.index))
48
49 test = Medley.iloc[0:12236]
50 test.index = range(len(test.index))
51
52 print("Number of audios=", Medley.shape[0], " Number of classes=", len(Medley.instrument.unique()))
53 print(Medley.instrument.unique())
54
55 print("Train set size: ")
56 print(len(train))
57 print("Validation set size: ")
58 print(len(validation))
59 print("Test set size: ")
60 print(len(test))
61
62 """
63 #-----
64 #Load audio file linked to the uuid
65 def full_name(file):
66     correspding_row = Medley.loc[Medley['uuid4'] == file].iloc[0]
67     subset = str(correspding_row.loc['subset'])
68     instrument_id = str(correspding_row.loc['instrument_id'])
69     parts = ['Medley-solos-DB_', str(subset), '-', str(instrument_id), '_', file, '.wav.wav']
70     s = ''
71     file_name = s.join(parts)
72     return file_name
73
74 def find_file(file):
75     file_name = full_name(file)
76     path = '/home/ubuntu/Machine-Learning/Medley-solos-DB/'
77     parts = [path, file_name]
78     s = ''
79     link = s.join(parts)
80     return link

```



```

81
82 #-----
83 #Generating music data ndarray in frequency domain using librosa
84
85 def get_music_features(dataset):
86     timeseries_length = 3
87     audio = np.zeros((len(dataset), timeseries_length, 21), dtype=np.float64)
88
89     for i in range(len(dataset)):
90         row = dataset.loc[i]
91         uuid4_name = str(row.loc['uuid4'])
92         link = find_file(uuid4_name)
93
94         y, sr = librosa.load(link)
95
96         mfcc = librosa.feature.mfcc(y=y, sr=sr, n_mfcc=13)
97         spectral_center = librosa.feature.spectral_centroid(y=y, sr=sr)
98         spectral_contrast = librosa.feature.spectral_contrast(y=y, sr=sr)
99
100        audio[i, :, 0:13] = mfcc.T[0:timeseries_length, :]
101        audio[i, :, 13:14] = spectral_center.T[0:timeseries_length, :]
102        audio[i, :, 14:21] = spectral_contrast.T[0:timeseries_length, :]
103
104
105        if ((i + 1) % 100 == 0):
106            print("Extracted features audio clip %i of %i." % (i + 1, len(dataset)))
107
108    return audio
109
110 train_audio_data = get_music_features(train)
111 validation_audio_data = get_music_features(validation)
112 test_audio_data = get_music_features(test)
113
114 np.save('train_audio_data.npy', train_audio_data)
115 np.save('validation_audio_data.npy', validation_audio_data)
116 np.save('test_audio_data.npy', test_audio_data)
117 """
118 # -----
119 # load npy music feature matrix data
120 train_audio_data = np.load('train_audio_data.npy')

```

```

121 validation_audio_data = np.load('validation_audio_data.npy')
122 test_audio_data = np.load('test_audio_data.npy')
123
124
125 # -----
126 # Process Dataset
127
128 class MedleyDataset(Dataset):
129     def __init__(self, dataset_csv, transform=None):
130         self.dataset_frame = dataset_csv
131         self.transform = transform
132
133     def __len__(self):
134         return len(self.dataset_frame)
135
136     def __getitem__(self, index):
137         uuid4 = self.dataset_frame.iloc[index, 4]
138         instrument_list = self.dataset_frame.iloc[index, 2]
139         instrument_id = int(instrument_list)
140         set = self.dataset_frame
141         if (set.shape == train.shape):
142             feature_matrix = train_audio_data
143         if (set.shape == validation.shape):
144             feature_matrix = validation_audio_data
145         if (set.shape == test.shape):
146             feature_matrix = test_audio_data
147
148         audio = feature_matrix[index, :, :]
149
150         audio = audio.astype('float')
151
152         sample = {'audio': audio, 'label': instrument_id}
153
154         if self.transform:
155             sample = self.transform(sample)
156
157         return sample
158
159
160 audio_dataset_train = MedleyDataset(dataset_csv=train)

```

```

161 audio_dataset_validation = MedleyDataset(dataset_csv=validation)
162 audio_dataset_test = MedleyDataset(dataset_csv=test)
163
164
165 # -----
166 # LSTM Model
167
168 class Net(nn.Module):
169     def __init__(self, input_size, hidden_size1, hidden_size2, num_classes):
170         super(Net, self).__init__()
171         self.LSTM1 = nn.LSTM(input_size, hidden_size1, batch_first=True, num_layers=2, dropout=0.1).cuda()
172         #self.LSTM2 = nn.LSTM(hidden_size1, hidden_size2, batch_first = True).cuda()
173         self.lstm2tag = nn.Linear(hidden_size1, num_classes).cuda()
174
175     def forward(self, x):
176         #print(x.shape)
177         s = x.shape[0]
178         x = x.reshape(s, 1, input_size)
179         out, states = self.LSTM1(x)
180         #out, states = self.LSTM2(out)
181         out = out[:, 0, :]
182         out = self.lstm2tag(out)
183         return out
184
185 # -----
186 # Train LSTM
187
188
189 model = Net(input_size, hidden_size1, hidden_size2, num_classes)
190 criterion = nn.CrossEntropyLoss()
191
192
193 #optimizer = torch.optim.Adadelta(model.parameters(), rho = 0.8, eps = 1e-6, lr=learning_rate)
194 #optimizer = torch.optim.SGD(params = model.parameters(), lr=learning_rate, momentum=0, dampening=0, weight_decay=0, nesterov=False)
195 optimizer = torch.optim.Adam(params=model.parameters(), lr=learning_rate, betas=(0.9, 0.999), eps=1e-08, weight_decay=0, amsgrad=False)
196
197 train_loader = torch.utils.data.DataLoader(dataset=audio_dataset_train, batch_size=batch_size, shuffle=True)
198 validation_loader = torch.utils.data.DataLoader(dataset=audio_dataset_validation, batch_size=batch_size, shuffle=True)
199 test_loader = torch.utils.data.DataLoader(dataset = audio_dataset_test, batch_size = batch_size, shuffle = True)
200

```

```

201
202 start_time = time.time()
203
204 epochs = np.array([])
205 loss_index = np.array([])
206
207 for epoch in range(num_epochs):
208     for i, data in enumerate(train_loader):
209
210         model.zero_grad()
211
212         audios = data['audio']
213         labels = data['label']
214
215         audios = audios.type(torch.FloatTensor)
216         audios = Variable(audios.cuda())
217
218         output = model(audios)
219
220         labels = labels.type(torch.LongTensor)
221         labels = Variable(labels.cuda())
222
223         loss = criterion(output, labels).cuda()
224         loss.backward()
225         optimizer.step()
226         optimizer.zero_grad()
227
228
229         if (i + 1) % 10 == 0:
230             print('Epoch [%d/%d], Step [%d/%d], Loss: %.4f'
231                   % (epoch + 1, num_epochs, i + 1, len(train) // batch_size, loss.data[0]))
232
233 epochs = np.append(epochs, epoch)
234 loss_index = np.append(loss_index, loss.item())
235
236 end_time = time.time()
237 elapsed_time = end_time - start_time
238 print("Elapsed time:", elapsed_time)
239
240

```

```

241
242 fig, ax = plt.subplots()
243 ax.plot(epochs, loss_index)
244
245 ax.set(xlabel='Epoch', ylabel='Performance Index', title='Performance Index Over Time')
246 ax.grid()
247 fig.savefig("test.png")
248 plt.show()
249
250
251 # -----
252 # Test accuracy on validation set
253
254 correct = 0
255 total = 0
256
257 for i, data in enumerate(validation_loader):
258
259     audios = data['audio']
260     labels = data['label']
261
262     audios = audios.type(torch.FloatTensor)
263     audios = Variable(audios.cuda())
264
265     outputs = model(audios)
266
267     labels = labels.type(torch.LongTensor)
268     labels = Variable(labels.cuda())
269
270     _, predicted = torch.max(outputs.data, 1)
271
272     total += labels.size(0)
273     correct += (predicted == labels).sum()
274
275
276     leng = predicted.shape[0]
277
278     a = predicted.cpu().detach().numpy()
279     b = labels.cpu().detach().numpy()
280

```

```

281
282     for j in range(leng):
283         k1 = a[j]
284         k2 = b[j]
285         confusion_m[k1,k2] += 1
286
287 #confusion matrix
288 confusion_m.astype(int)
289 ax = sns.heatmap(confusion_m, annot=True)
290 print(confusion_m)
291 print('Accuracy of the network on the 3494 validation audio clips: %d %%' % (100 * correct / total))
292 # -----
293 # Test accuracy of each class on Validation set
294
295 class_correct = list(0. for i in range(8))
296 class_total = list(0. for i in range(8))
297
298 for data in validation_loader:
299     audios = data['audio']
300     labels = data['label']
301
302     audios = audios.type(torch.FloatTensor)
303     audios = Variable(audios.cuda())
304
305     outputs = model(audios)
306
307     labels = labels.type(torch.LongTensor)
308     labels = Variable(labels.cuda())
309
310     _, predicted = torch.max(outputs.data, 1)
311
312     labels = labels.cpu().numpy()
313     c = (predicted.cpu().numpy() == labels)
314     s = labels.shape[0]
315     for i in range(s):
316         label = labels[i]
317         class_correct[label] += c[i]
318         class_total[label] += 1
319
320 # -----

```

```

321
322 for i in range(8):
323     if (class_total[i] == 0):
324         print('Accuracy of %5s : %2d %%' % (classes[i], 0))
325     else:
326         print('Accuracy of %5s : %2d %%' % (classes[i], 100 * class_correct[i] / class_total[i]))
327 # -----
328 torch.save(model.state_dict(), 'model_time.pkl')
329
330 # -----
331 # Test accuracy on test set
332
333 correct = 0
334 total = 0
335
336 for i, data in enumerate(test_loader):
337
338     audios = data['audio']
339     labels = data['label']
340
341     audios = audios.type(torch.FloatTensor)
342     audios = Variable(audios.cuda())
343
344     outputs = model(audios)
345
346     labels = labels.type(torch.LongTensor)
347     labels = Variable(labels.cuda())
348
349     _, predicted = torch.max(outputs.data, 1)
350
351     total += labels.size(0)
352     correct += (predicted == labels).sum()
353
354
355     leng = predicted.shape[0]
356
357     a = predicted.cpu().detach().numpy()
358     b = labels.cpu().detach().numpy()
359
360

```

```

361     for j in range(leng):
362         k1 = a[j]
363         k2 = b[j]
364         confusion_m[k1,k2] += 1
365
366 #confusion matrix
367 confusion_m.astype(int)
368 ax = sns.heatmap(confusion_m, annot=True)
369 print(confusion_m)
370 print('Accuracy of the network on the 3494 validation audio clips: %d %%' % (100 * correct / total))
371 # -----
372 # Test accuracy of each class on test set
373
374 class_correct = list(0. for i in range(8))
375 class_total = list(0. for i in range(8))
376
377 for data in test_loader:
378     audios = data['audio']
379     labels = data['label']
380
381     audios = audios.type(torch.FloatTensor)
382     audios = Variable(audios.cuda())
383
384     outputs = model(audios)
385
386     labels = labels.type(torch.LongTensor)
387     labels = Variable(labels.cuda())
388
389     _, predicted = torch.max(outputs.data, 1)
390
391     labels = labels.cpu().numpy()
392     c = (predicted.cpu().numpy() == labels)
393     s = labels.shape[0]
394     for i in range(s):
395         label = labels[i]
396         class_correct[label] += c[i]
397         class_total[label] += 1
398
399 # -----
400
401 for i in range(8):
402     if (class_total[i] == 0):
403         print('Accuracy of %5s : %2d %%' % (classes[i], 0))
404     else:
405         print('Accuracy of %5s : %2d %%' % (classes[i], 100 * class_correct[i] / class_total[i]))
406 # -----
407 torch.save(model.state_dict(), 'model_time.pkl')

```