Music Analysis using Computational Methods

A Dissertion

submitted in Partial Fulfilment of the Requirements for the Award of Degree of

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In

PHYSICS

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CERTIFICATE

The undersigned certifies that **Mr. SHASHANK SINHA**, **Roll No. (1810002)**, **Enrolment No. (180222)** is registered for the Five-Year Integrated Master's Program in the Department of Physics.

I hereby recommend that the Dissertation entitled, **MUSIC ANALYSIS USING COMPUTATIONAL METHODS** be accepted as the partial fulfilment of the requirements for evaluation and award of the five-year Integrated M.Sc. Degree in Physics.

Date……………………….

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Thanking You

SHASHANK SINHA

Contents

Abstract

Music is nothing but a sequence of events, whether it is a drum roll or guitar riff or even piano solo, all are basically a sequence of music elements. And the thing our ears like is the correlation between music elements, which comes after each other. For example, a note or chord can create a dissonance, and what follows it is a resolving music element, which resolves that dissonance. And it is this correlation between music elements, which we can use to our advantage in working on music with algorithms as our tools.

Music generation can be formulated as a next music element prediction problem, which would allow us to generate as much music as we wanted by just keep passing the previously generated music element.

For implementation, used GRU (Gated Recurrent Unit) in-place of the vanilla RNN (Recurrent Neural Network), because of its long term dependencies retaining ability, which is much needed in music dataset. Each GRU takes previous layer's output as input and activation, and the output would be the next music element.

Introduction

Machine learning is now being used for many interesting applications in a variety of fields. Music generation is one of the interesting applications of machine learning. As music itself is sequential data, it can be modelled using a sequential machine learning model such as the recurrent neural network.

About the Dataset

- A music element (data-point) is described in terms of following parts :
    1. *Pitch* : The measure of frequency of the sound. It is a theoretical term, but in music, it is specified in terms of notes and chords, along with their octaves.
    2. *Notes* : This represents relatively specified frequencies, which are distributed within an octave. There are 12 different note-names in music, namely C, C# or Db (called as ‘C-sharp’, or ‘D-flat’), D, D#, E, F, F#, G, G#, A, A#, and B.
    3. *Chords* : This represents a group of notes, in which basically frequencies of constituent notes overlap, and a beam of frequencies is what we listen. For example: F-major (F-note + A-note + C-note).
    4. *Octave* : These are specified frequency-ranges, which periodically repeats after spanning a certain frequency range. The higher the octave number, the higher is its frequency range, i.e. 5^{th} octave has higher frequency than that of 4^{th} octave. Taking a note as example, if we try to read music-files as text like “C4”, this denotes C-note of 4^{th} octave.

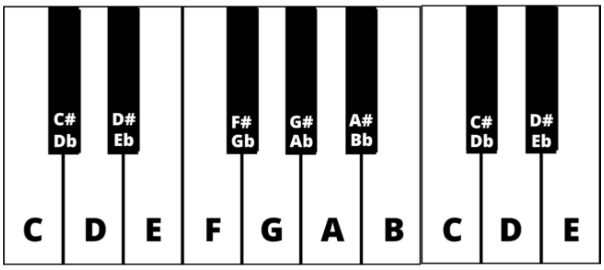


Figure 1: Music Notes (can be better visualized as Piano-keys)

5. *Offset* : It is the instance(timing of occurence) of a music-element. The rhythm of music depends on this, i.e. the timing-gap of 2 music elements.
6. *Instrument* : The instrument on which the music element is played. This alters the timbre of that music element.

- All the audio files are .midi files, collecting only piano music. Chosen such dataset, because it's easy to retrieve data from this kind of music.

Algorithms used (in Neural Network) :

1) ***LSTM Layer***

- *Long Short Term Memory* (usually called LSTMs).
- LSTM is a special kind of *RNN (Recurrent Neural Network)*. In RNN, same weights & activation function are used in each iteration to predict next future value.
- In RNN :

$$a_1 = A[(W_{xa})*(x_1) + (W_{aa})*(a_0) + b_a]$$
$$y_1 = A'[(W_{ay})*(a_1) + b_y]$$

....

$$a_n = A[(W_{xa})*(x_n) + (W_{aa})*(a_{n-1}) + b_a]$$
$$y_n = A'[(W_{ay})*(a_n) + b_y]$$

- In calculation of a_n, gradient corresponding to a_0 will be n-times product of (W_{aa}). So if :

W_{aa} > 1 : *Exploding Gradient problem* will emerge. The resulting gradient will be extremely large, & will negatively effect predictions. This problem is solvable by implementing some limitations on each gradient.

W_{aa} < 1 : *Vanishing Gradient problem* will emerge. The resulting gradient will be almost 0, i.e. no effect of old outputs on new predictions. This is also a negative effect, and solving this is a relatively difficult task.

- For solving Vanishing Gradient problem, a different Neural network unit comes in use, i.e. *Gated Recurrent Unit (GRU)*. In this unit, a potential value for prediction is calculated & a sigmoid update-function is there to decide whether value of a_{i-1} is assigned to a_i , or the calculated potential value. There is also a *Reset Gate*, which decides how much is (a_{i-1}) contributing to the potential value. Following are the equations :

  ➔ Reset Gate :

  $R_{gate} = sigma[(W_r)*(x_i) + (W'_r)*(a_{i-1}) + b_3]$

  ➔ Potential value :

  $a_{potential\_i} = tanh[(W_{aa})*(a_{i-1})*(R_{gate}) + (W_{xa})*(x_i) + b_1]$

  ➔ Sigmoid update-function :

  $u = sigma[(W'_a)*(a_{i-1}) + (W'_x)*(x_i) + b_2]$

  ➔ Decider equation :

  $a_i = (u)*(a_{potential\_i}) + (1-u)*(a_{i-1})$

- In LSTM, the algorithm goes one-step beyond. As there is a Reset Gate in GRU, there are 3-gates in LSTM :- (equation of all gates are similar to that of reset gate, i.e. all are sigma function)

  i.) *Input Gate(in)* -- tells what will be the contribution of potential value (c_{t_cap}) in prediction(c_t) .

  ii.) *Forget Gate(f)* – tells what will be the contribution of past value (c_{t-1}) in prediction(c_t) .

iii.) ***Output Gate(g_{out})*** – calculates activation function for next layer of LSTM.

➔ Prediction :

$c_t = (in)*(c_{t\_cap}) + (f)*(c_{t-1})$

- LSTM is more complex algorithm than RNN & GRU. As they also store more memory for future predictions. They store – 3 gates, many weights, past value(c_{t-1}), and past activation function(a_{t-1}).
- LSTM(units, input_shape, return_sequences)
  Above is the syntax of LSTM layer, where :
  units = dimension of output of this layer, input_shape = dimension of input to this layer. And, return_sequences = boolean value. If true, it returns only the last output of output sequence, else returns the full output sequence.

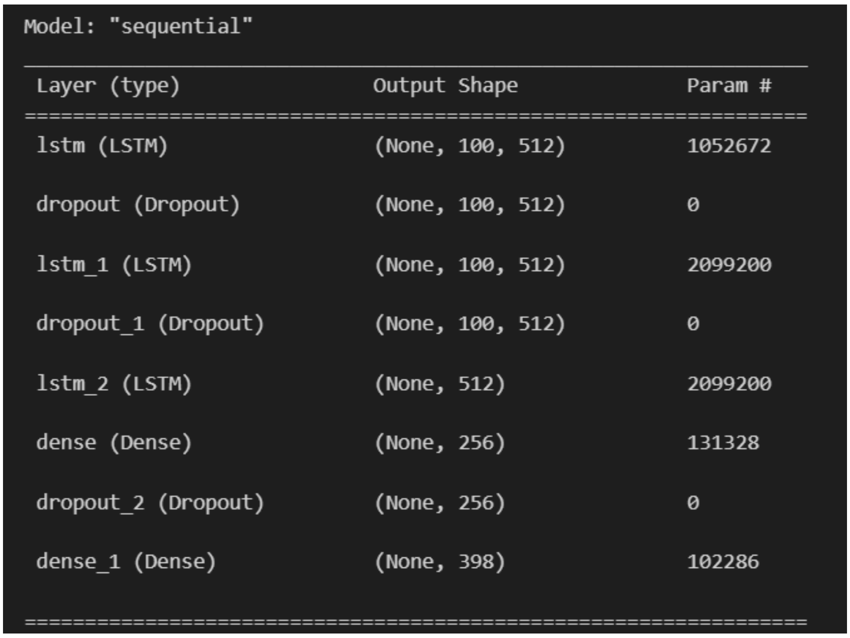
2) Dropout Layer

- Dropout refers to data, or noise, that is dropped intentionally from a neural network to improve processing, to reach to better results.
- Similar to human brain, the neural-network units randomly process countless inputs, and give countless outputs at the same time. So, it might happen that an intermediate output thing gets passed to another neural-network unit even before the end output gets calculated. And, some of these processes result in noise creation.
- Dropout(x) : The syntax of this argument, where x refers to the fraction of previous layer's output to drop. So, value of x lies between 0 and 1.

3) Dense Layer

- Dense layer has a deep connection in neural network, i.e. each neuron in dense layer is receiving inputs from all previous layer neurons. So, it helps in developing a better correlation in neural network.
- This layer performs a matrix-vector multiplication, where values used as correlation-factors are trained with the help of backpropagation.
- Dense(units, activation = <*activation-function*>).

  Above is the syntax of this layer where, units = output-size of this layer. And activation function helps to develop output though complex functions, based on input. By default, it uses linear function (i.e. linear regression model).
- But in final layer of neural network, prediction is needed in terms of probability-distribution, so used "softmax" activation function. In other layers, "linear" activation function is used, as simple linear regression model is needed to predict next note.



Model: "sequential"

Layer (type)	Output Shape	Param #
lstm (LSTM)	(None, 100, 512)	1052672
dropout (Dropout)	(None, 100, 512)	0
lstm_1 (LSTM)	(None, 100, 512)	2099200
dropout_1 (Dropout)	(None, 100, 512)	0
lstm_2 (LSTM)	(None, 512)	2099200
dense (Dense)	(None, 256)	131328
dropout_2 (Dropout)	(None, 256)	0
dense_1 (Dense)	(None, 398)	102286

Figure 2 : The Neural Network model (using these layers)

Data Processing :

- In the dataset, there is only 2 types of music elements considered, i.e. notes and chords.
- In case of note, extracted the pitch of music element and converted it to string. And then further mapped the string with an integer, using hashmap data structure.

Processing a Note :-

```python
note1 = elements_of_song[3]
print(note1.pitch)
print(type(note1))
# This gives the note in form of a class
print(type(note1.pitch))
# Get the string from the class
currNote = str(note1.pitch)
print(currNote)
# This will recover the note-name from class
```

[118] ✓ 0.1s Python

```
C5
<class 'music21.note.Note'>
<class 'music21.pitch.Pitch'>
```

- In case of chord, extracted the constituent notes using "chord.normalOrder" command, which will give an array of random integers corresponding to notes, and converted it to string similar to notes. So that, it can be mapped in the same way as notes, so that all music elements can be treated similarly.

Processing a Chord :-

```python
chord1 = elements_of_song[1]
print(chord1)
print(type(chord1))
# This is a chord, let's figure this out.. how to process this
print(chord1.normalOrder)
# chord.normalOrder --> Gives the list of nodes in it.
# 2 --> A4
# 6 --> D5
# 9 --> F#4
# (Following some pattern of indexing.. have to figure it out)
print(type(chord1.normalOrder))
# Convert the chord-list into a string, concatenated with "+"
currChord = "+".join(str(x) for x in chord1.normalOrder)
print(currChord)
```

[119] ✓ 0.0s Python

```
<music21.chord.Chord C5 E4>
<class 'music21.chord.Chord'>
[0, 4]
<class 'list'>
0+4
```

- After retrieving all music elements from all the music files, created an array comprising of only unique music elements from the collection. And then mapped those elements with their index. This mapping was required because we have some random collection of strings, but the LSTM model works on numerical data, so we can convert our data to numerical using this map.

Preparing Sequenctial Data for LSTM :-

Choosing a sequence length which states that how many elements are considered in a LSTM layer, which is 1st layer of the neural network (created further).

Markdown

```python
sequenceLength = 100
# Will give 100 elements to a layer, and will predict output for next layer using them.
uniqueNotes = sorted(set(notes))
countNodes = len(uniqueNotes)
print("No. of elements in uniqueNotes = ", len(uniqueNotes))
print("Some elements of uniqueNotes array are :-")
count = 0
for ele in uniqueNotes:
    print(ele)
    count += 1
    if count > 7:
        break
print("...")
```

[127] ✓ 0.0s Python

```
No. of elements in uniqueNotes =  398
Some elements of uniqueNotes array are :-
0
0+1
0+1+3
0+1+5
```

- Then normalized the input-data, i.e. shrinked the values of all the datapoints betweens 0 and 1. Initially they were the indices of the unique-element array, so the values were from 0 to size of array (only integer values). And for algorithms, there were many datapoints possible in between them, as they work in high precision. So, normalization will help in reducing additional noise, produced by varied range of data points, i.e. data having high variance.

Making Prediction :

- As the sequenceLength was specified as 100, so the dimension of the input array must be 100 i.e., we have to give 100 music elements as input.
- For each element of input, first calculated it's normalized value, and then made prediction using the trained sequential model.
- The sequential model will predict the probability distribution of all unique elements as softmax activation function is used in the

last dense layer. And from that array, the element with maximum probability distribution is considered as the next predicted music element.

- In each step, we will keep incrementing the index with maximum probability value to the input array, which will increase the size to 101. And to check down the size, will keep discarding the oldest music element, i.e. at index 0. As the correlation of newly predicted element or upcoming predictions with the oldest element will be the lowest.
- Now, the music element corresponding to the index having maximum probability is retrieved using reverse mapping, which was created using the same unique notes array, which was used to create unique music element strings to integer mapping.

```python
# Trying to generate (numIteration)-elements of music
numIteration = 200

for noteIdx in range(numIteration):
    predictionInput = np.reshape(pattern, (1,len(pattern), 1))
    predictionInput = predictionInput / float(countNodes)
    # Making prediction
    prediction = model.predict(predictionInput, verbose=0)
    # Taking the element with max. probability
    idx = np.argmax(prediction)
    # index (unique-note index too) corresponding to max. probability element
    result = intNoteMap[idx]
    # Appending this element to prediction-array
    predictionOutput.append(result)
    pattern.append(idx)
    # slicing out the oldest element (0th index)
    pattern = pattern[1:]
    # Size of pattern remained constant at 100 (as needed by model).
    #     (as added 1 element, & removed 1)
```
[160] ✓ 2m 56.7s Python

Figure 3 : Prediction using model

Generating Music out of Predicted-data :

- After revere mapping, we have elements in terms of strings. As the dataset is considered, there can be two possibilities i.e., notes and chords.
- For discriminating notes and chords, the character ‘+’ was added in chords for concatenating the integers corresponding

to constituent notes. And then processing of constituent notes and the note elements are similar.

- For generating the notes, used the Note() method of note class of music21 library of python, which is basically reverse generating the note using which we generated the integer using the same method of same class.
- Also setting the instrument of each note, as it might be storing a garbage value by default. As the input dataset was only of piano, so manually setting instrument of all the notes as piano. Also setting the offset manually, which is the instance of that music element in the audio file's duration.
- As we are setting the offset manually, therefore the output music will be mostly sound flat seeing the rhythm. To make better music prediction, we can store the offset of input music elements in another array, and also the instrument of different music elements in another array, and similarly other factors. If these factors are considered, we can replicate the rhythm and music arrangement patterns of input music.

Creating Music-Elements from String-array :-

```python
offset = 0
# offset --> instance-time of particular element (note/chord)
# Have to iterate over all elements of predictionOutput
#   --> Checking whether is a note or chord ?

for element in predictionOutput:
    # If element is a chord :-
    if('+' in element) or element.isdigit():
        # Possibilites are like '1+3' or '0'.
        notesInChord = element.split('+')
        # This will get all notes in chord
        tempNotes = []
        for currNote in notesInChord:
            # Creating note-object for each note in chord
            newNote = note.Note(int(currNote))
            # Set it's instrument
            newNote.storedInstrument = instrument.Piano()
            tempNotes.append(newNote)
        # This chord can have x-notes
        # Create a chord-object from list of notes
        newChord = chord.Chord(tempNotes)
        # Adding offset to chord
        newChord.offset = offset
        # Add this chord to music-elements
        outputMusicElements.append(newChord)
    # If element is a note :-
    else:
        # We know that this is a note
        newNote = note.Note(element)
        # Set off-set of note
        newNote.offset = offset
        # Set the instrument of note
        newNote.storedInstrument = instrument.Piano()
        # Add this note to music-elements
        outputMusicElements.append(newNote)
    offset += 0.5
    # Fixing the time-duration of all elements
```

[166] Python

Figure 4 : Working code of music generation from predicted data

- From the music elements, combined them with an offset in between, and made out a midi file out of it. To visualize the input and output music out of the neural network model, plotted it side by side to each other. Where the offset of output music starts from where the input music ends.

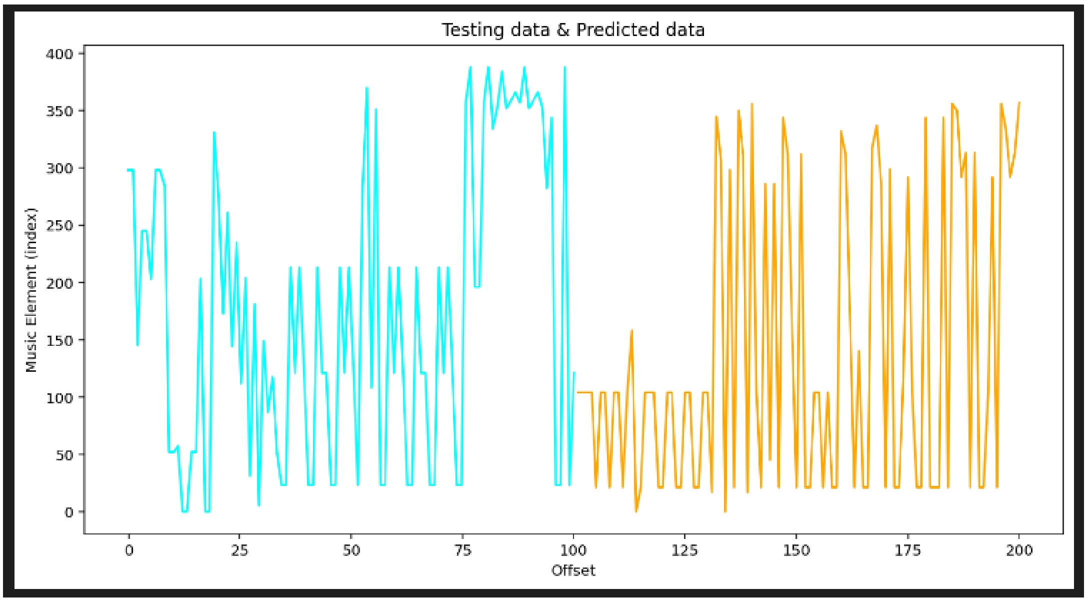


Figure 5 : Input music VS Output music

Conclusions :-

The predicted music follows some correlation trends of input music, but at the same time in most cases they are far from ideal music sounds. This is because we have done this only for one instrument, and for multiple instruments, the combination count increases exponentially. So, it is difficult to accumulate such cases, to predict a practical sounding music, specifically the training of neural network model, which will be the biggest task to handle. Also the offset is set manually, so every note of same length doesn't sound musically good for most of the times.

References :-

http://www.piano-midi.de/midi_files.htm (for midi files dataset)

https://abc.sourceforge.net/NMD/ (for midi files dataset)

Full Code :-

In [1]:

```
from music21 import converter, instrument, note, chord, stream
import glob
import pickle
import numpy as np
```

Read a Midi File

In [2]:

```
song1 = converter.parse("midi_songs/8.mid")
print(type(song1))
```

```
<class 'music21.stream.base.Score'>
```

In [3]:

```
song1
```

Out[3]:

```
<music21.stream.Score 0x223b220e040>
```

In [4]:

```
# song1 --> object of stream.Score type
#       --> will contain music in form of notes and chords
song1.show('midi') # Shows the song in playable format
```

In [5]:

```
# So, the chords and notes are stored in nested forms of containers
# .. to simplify this, store all of them in a single list
# ==> Flatten the elements.
elements_of_song = song1.flat.notes
print(len(elements_of_song))
print(elements_of_song)
print(type(elements_of_song))
```

```
336
<music21.stream.iterator.StreamIterator for Score:0x223cbdea760 @:0>
<class 'music21.stream.iterator.StreamIterator'>
```

In [6]:

```
count = 0
print("Following are some elements of song :-")
for e in elements_of_song:
    print(e, e.offset, type(e))
    count += 1
    if count > 7:
        break
    # e.offset --> will tell the time-duration of element
```

```
Following are some elements of song :-
<music21.note.Note C> 0.0 <class 'music21.note.Note'>
<music21.chord.Chord C5 E4> 0.0 <class 'music21.chord.Chord'>
<music21.note.Note C> 0.0 <class 'music21.note.Note'>
<music21.note.Note C> 0.0 <class 'music21.note.Note'>
<music21.chord.Chord C5 E4> 0.0 <class 'music21.chord.Chord'>
<music21.note.Note C> 0.0 <class 'music21.note.Note'>
<music21.note.Note G> 1.5 <class 'music21.note.Note'>
<music21.note.Note G> 5/3 <class 'music21.note.Note'>
```

Get All the Notes, from all the Midi Files

In [7]:

```
notes = []
count = 0
for file in glob.glob("midi_songs/*.mid"):
    midi = converter.parse(file) # Convert file into stream.Score Object
    if count < 10:
        print("parsing %s"%file)
    elements_to_parse = midi.flat.notes
    count += 1

    for elex in elements_to_parse:
        # If the element is a Note,  then store it's pitch
        if(isinstance(elex, chord.Chord) == True):
            notes.append("+".join(str(n) for n in elex.normalOrder))
        elif(isinstance(elex, note.Note) == True):
            noteString = str(elex.pitch)
            notes.append(noteString)
            # If the element is a Chord, split each note of chord and join them with +
print("...")
```

```
parsing midi_songs\0fithos.mid
parsing midi_songs\8.mid
parsing midi_songs\ahead_on_our_way_piano.mid
parsing midi_songs\AT.mid
parsing midi_songs\balamb.mid
parsing midi_songs\bcm.mid
parsing midi_songs\BlueStone_LastDungeon.mid
parsing midi_songs\braska.mid
parsing midi_songs\caitsith.mid
parsing midi_songs\Cids.mid
...
```

In [8]:

```
print("Length of notes-array = ", len(notes))
print("Some elements of note array :-")
count = 0
for n in notes:
    print(n)
    count += 1
    if count > 7:
        break
print("...")
```

```
Length of notes-array =  60764
Some elements of note array :-
4+9
E2
4+9
4+9
4+9
4+9
4+9
4+9
...
```

Saving the file, containing all Notes

In [9]:

```
import pickle

with open("notes", 'wb') as filepath:
    pickle.dump(notes, filepath)
```

In [10]:

```
# 'wb' --> Write-binary mode (to write data in a file)
# 'rb' --> Read-binary mode (to read data from a file)
with open("notes", 'rb') as f:
    notes = pickle.load(f)
    # This will load whole file-data to variable notes
```

Count of Unique Elements in Music :-

In [11]:

```
# In 'wb' and 'rb', same file needs to be referenced.
# Else, Will give error --> "Ran out of data".
print(len(set(notes)))
# This will print unique no. of elements.
# i.e. --> Unique notes/chords in all files.
numElements = len(set(notes))
```

```
398
```

Preparing Sequential Data for LSTM :-

In [12]:

```
sequenceLength = 100
uniqueNotes = sorted(set(notes))
countNodes = len(uniqueNotes)
print("No. of elements in uniqueNotes = ", len(uniqueNotes))
print("Some elements of uniqueNotes array are :-")
count = 0
for ele in uniqueNotes:
    print(ele)
    count += 1
    if count > 7:
        break
print("...")
```

```
No. of elements in uniqueNotes =  398
Some elements of uniqueNotes array are :-
0
0+1
0+1+3
0+1+5
0+1+6
0+2
0+2+3+7
0+2+4+5
...
```

Mapping Strings (unique-elements) to Integer values :-

In [13]:

```
# As ML models work with numerial data only, will map each string with a number.
noteMap = dict((ele, num) for num, ele in enumerate(uniqueNotes))
count = 0
for ele in noteMap:
    print(ele, " : ", noteMap[ele])
    count += 1
    if count > 7:
        break
print("...")
```

```
0  :  0
0+1  :  1
0+1+3  :  2
0+1+5  :  3
0+1+6  :  4
0+2  :  5
0+2+3+7  :  6
0+2+4+5  :  7
...
```

--> As sequenceLength is 100, will take first 100 data to input, and 101st data as output. --> For next iteration, take (2-101) data points as input, and 102nd data as output. --> So on... Sliding window (of size 100) as input, & next 1 data as output.

--> So, total we will get (len(notes) - sequenceLength) datapoints.

In [14]:

```
networkInput = [] # input-data
networkOutput = [] # will try to get output, using input
for i in range(len(notes) - sequenceLength):
    inputSeq = notes[i : i+sequenceLength] # 100 string-values
    outputSeq = notes[i + sequenceLength] # 1 string-value
    # Currently, inputSeq & outputSeq has strings.
    # Use map, to convert it to integer-values.
    # ..as ML-algorithm works only on numerical data.
    networkInput.append([noteMap[ch] for ch in inputSeq])
    networkOutput.append(noteMap[outputSeq])
```

In [15]:

```
print(len(networkInput))
print(len(networkOutput))
```

```
60664
60664
```

Create ready-data for Neural Network :-

In [16]:

```
import numpy as np
```

In [17]:

```
# n_patterns = int(len(networkInput)/100)
# No. of rows divided by 100.. as 100 columns, so Distributing data in 3-D format
n_patterns = len(networkInput)
networkInput = np.reshape(networkInput, (n_patterns, sequenceLength, 1))
# LSTM recieves input data in 3-dimensions
print(networkInput.shape)
```

```
(60664, 100, 1)
```

Normalize this data

In [18]:

```
# As the values are from 0 - uniqueNodes
# For better precision, converting data in range [0 - 1]
normNetworkInput = networkInput / float(numElements)
print("Some elements of normNetworkInput[0] array :-")
count = 0
for ele in normNetworkInput[0]:
    print(ele)
    count += 1
    if count > 10:
        break
print("...")
```

```
Some elements of normNetworkInput[0] array :-
[0.48743719]
[0.92713568]
[0.48743719]
[0.48743719]
[0.48743719]
[0.48743719]
[0.48743719]
[0.48743719]
[0.48743719]
[0.2638191]
[0.48743719]
...
```

In [19]:

```
# Now, values are in range [0 - 1]
# Network output are the classes, encoded into 1-vector
```

In [20]:

```
from keras.utils import np_utils
```

In [21]:

```
networkOutput = np_utils.to_categorical(networkOutput)
print(networkOutput.shape)
# This will convert output-data to a 2-D format
# In which each key(old-output value) has 229 categorical values
# And, the one which matches has some kind of flag marked to it.
```

```
(60664, 398)
```

Create Model

Download & Import Packages

In [22]:

```
from keras.models import Sequential
from keras.layers import *
from keras.callbacks import ModelCheckpoint, EarlyStopping
```

Creating a Sequential Model :-

In [23]:

```
model = Sequential()
```

Adding Layers to the Model :-

In [39]:

```
# And, this model has first layer as LSTM layer.
model.add(LSTM(units=512, input_shape=(normNetworkInput.shape[1], normNetworkInput.shape[2]), return_sequences=True))
# As this is the 1st layer, so we need to provide the input-shape (in argument)
# Here we are passing (100,1) as input_shape, as all data-points have shape (100,1)
# Also, we have to do return_sequences=True, as this isn't the last layer, also have further layers.

# After the 1st layer, adding a Dropout
model.add(Dropout(0.3))

# Also adding another LSTM layer.
model.add(LSTM(512, return_sequences=True))
# return_sequences=True --> returns only last output of output seq.

# Again adding a Dropout
model.add(Dropout(0.3))
```

```python
# And, now 1-more LSTM layer.
model.add(LSTM(512))
# return_sequences=False(default) --> returns whole output-seq.

# And, adding a Dense-layer.
model.add(Dense(256))

# Again adding a Dropout.
model.add(Dropout(0.3))

# Now, the final layer.
#    (Adding dense layer with no. of neurons = countNodes)
#    (Also having an "softmax" activation function)
model.add(Dense(numElements, activation="softmax"))
```

Compiling the model :-

In [40]:

```python
model.compile(loss="categorical_crossentropy", optimizer="adam")
# loss="categorical_crossentropy" --> since it has 229 classes.
# Not specifying any metrics (like accuracy), as it would not be a good metrics to evaluate.
```

This is our Model :-

In [42]:

```python
model.summary()
```

```
Model: "sequential"
_________________________________________________________________
 Layer (type)                Output Shape              Param #   
=================================================================
 lstm (LSTM)                 (None, 100, 512)          1052672   
                                                                 
 dropout (Dropout)           (None, 100, 512)          0         
                                                                 
 lstm_1 (LSTM)               (None, 100, 512)          2099200   
                                                                 
 dropout_1 (Dropout)         (None, 100, 512)          0         
                                                                 
 lstm_2 (LSTM)               (None, 512)               2099200   
                                                                 
 dense (Dense)               (None, 256)               131328    
                                                                 
 dropout_2 (Dropout)         (None, 256)               0         
                                                                 
 dense_1 (Dense)             (None, 398)               102286    
                                                                 
=================================================================
Total params: 5,484,686
Trainable params: 5,484,686
Non-trainable params: 0
_________________________________________________________________
```

Training the Model :-

In [41]:

```python
import tensorflow as tf
```

In [52]:

```python
# (Entire code commented out, to prevent created model, from starting fit again, and old work getting wasted)
# Creating callbacks for fitting model.

checkpoint = ModelCheckpoint("model3.hdf5", monitor='loss', verbose=0, save_best_only=True, mode='min')
# 1st arg  -->  where the model will be saved
# 2nd arg  -->  We have to monitor the loss
# 5th arg  -->  As monitoring loss, so mode = "min", as loss should be minimum.

# We can also create an earlystopping callback, but lets only keep the checkpoint.

# Fitting the model :-

model_his = model.fit(normNetworkInput, networkOutput, epochs=10, batch_size=64, callbacks=[checkpoint])
# No. of epochs = 10 (trying for model3)
# batch size = 64
# No. of epochs = 100 (for model4 .. trained in google-colab)
# Then imported that model to this file...
```

```
Epoch 1/10
948/948 [==============================] - 4003s 4s/step - loss: 4.8030
Epoch 2/10
948/948 [==============================] - 3837s 4s/step - loss: 4.7745
Epoch 3/10
948/948 [==============================] - 3842s 4s/step - loss: 4.7721
Epoch 4/10
948/948 [==============================] - 3839s 4s/step - loss: 4.7687
Epoch 5/10
948/948 [==============================] - 12148s 13s/step - loss: 4.7669
Epoch 6/10
948/948 [==============================] - 3739s 4s/step - loss: 4.7658
Epoch 7/10
948/948 [==============================] - 3994s 4s/step - loss: 4.7650
Epoch 8/10
948/948 [==============================] - 3497s 4s/step - loss: 4.7646
Epoch 9/10
948/948 [==============================] - 3606s 4s/step - loss: 4.7646
Epoch 10/10
948/948 [==============================] - 3747s 4s/step - loss: 4.7644
```

Load Model :-

In [42]:

```python
from keras.models import load_model
```

In [43]:

```python
model = load_model("model4.hdf5")
```

Predictions :-

In [44]:

```
sequenceLength = 100
```

In [45]:

```
networkInput = [] # input-data
for i in range(len(notes) - sequenceLength):
    inputSeq = notes[i : i+sequenceLength] # 100 string-values
    # Currently, inputSeq & outputSeq has strings.
    # Use map, to convert it to integer-values.
    # ..as ML-algorithm works only on numerical data.
    networkInput.append([noteMap[ch] for ch in inputSeq])
```

In [46]:

```
print("Some elements of networkInput[0] array :-")
count = 0
for ele in networkInput[0]:
    print(ele)
    count += 1
    if count > 7:
        break
print("...")
```

```
Some elements of networkInput[0] array :-
194
369
194
194
194
194
194
194
...
```

In [47]:

```
print(len(networkInput[300]))
```

```
100
```

In [48]:

```
# Each data-point has 100-elements (in networkInput)
# We will give these 100-elements as input, & it will generate 1-output.
# Will add this 1-output in input, & discard oldest element from input. (again getting to 100 input-elements)
# This way, we will keep predicting 1-element each time.
```

In [49]:

```
startIdx = np.random.randint(len(networkInput)-1)
# This will get any random data-point-index from the input-data
# Data at each random data-point-index means --> 100 elements.
```

In [50]:

```
print(startIdx)
```

```
16473
```

In [51]:

```
networkInput[startIdx]
print("Some elements of networkInput[startIdx] array are :-")
count = 0
for ele in networkInput[startIdx]:
    print(ele)
    count += 1
    if count > 7:
        break
print("...")
```

```
Some elements of networkInput[startIdx] array are :-
364
394
364
364
394
364
364
394
...
```

In [52]:

```
# Above 100-element np-array, is the start sequence.
# Right now, we have :-
# element --> integer  mapping
# What is also required is :-
# integer --> element  mapping.
```

In [53]:

```
intNoteMap = dict((num,ele) for num,ele in enumerate(uniqueNotes))
# This will have  (integer --> element) mapping.
# uniqueNotes --> has all unique-elements
# noteMap --> has  (element --> integer) mapping.
# countNodes --> count of unique-elements.

# print(intNoteMap)
# Commented, as this is just "integer" --> "music-element" mapping.
```

Generate Input-music in playable format :-

In [54]:

```
# Taking the initial input-index pattern
pattern = networkInput[startIdx]
predictionOutput = []
```

In [55]:

```python
inputMusicElements = []
inputMusic = []
inputMusic = (intNoteMap[ele] for ele in pattern)
```

In [56]:

```python
offset = 0
# offset --> instance-time of particular element (note/chord)
# Have to iterate over all elements of predictionOutput
#   --> Checking whether is a note or chord ?

for element in inputMusic:
    # If element is a chord :-
    if('+' in element) or element.isdigit():
        # Possibilites are like '1+3' or '0'.
        notesInChord = element.split('+')
        # This will get all notes in chord
        tempNotes = []
        for currNote in notesInChord:
            # Creating note-object for each note in chord
            newNote = note.Note(int(currNote))
            # Set it's instrument
            newNote.storedInstrument = instrument.Piano()
            tempNotes.append(newNote)
        # This chord can have x-notes
        # Create a chord-object from list of notes
        newChord = chord.Chord(tempNotes)
        # Adding offset to chord
        newChord.offset = offset
        # Add this chord to music-elements
        inputMusicElements.append(newChord)
    # If element is a note :-
    else:
        # We know that this is a note
        newNote = note.Note(element)
        # Set off-set of note
        newNote.offset = offset
        # Set the instrument of note
        newNote.storedInstrument = instrument.Piano()
        # Add this note to music-elements
        inputMusicElements.append(newNote)
    offset += 0.5
    # Fixing the time-duration of all elements
```

In [57]:

```python
# For playing them, have to create a stream-object
#    ..from the generated music-elements
midiInputStream = stream.Stream(inputMusicElements)
# Write this midiStream on a midi-file.
midiInputStream.write('midi', fp="testInput8.mid")
# 1st arg  -->  File-type
# 2nd arg  -->  File-name
```

Out[57]:

```
'testInput8.mid'
```

In [58]:

```python
midiInputStream.show('midi')
```

Making Prediction :-

In [59]:

```python
# Trying to generate (numIteration)-elements of music
numIteration = 200

for noteIdx in range(numIteration):
    predictionInput = np.reshape(pattern, (1,len(pattern), 1))
    predictionInput = predictionInput / float(countNodes)
    # Making prediction
    prediction = model.predict(predictionInput, verbose=0)
    # Taking the element with max. probability
    idx = np.argmax(prediction)
    # index (unique-note index too) corresponding to max. probability element
    result = intNoteMap[idx]
    # Appending this element to prediction-array
    predictionOutput.append(result)
    pattern.append(idx)
    # slicing out the oldest element (0th index)
    pattern = pattern[1:]
    # Size of pattern remained constant at 100 (as needed by model).
    #     (as added 1 element, & removed 1)
```

In [60]:

```python
print("No. of elements in predictionOutput = ", len(predictionOutput))
print("Some elements of predictionOutput array are :-")
count = 0
for ele in predictionOutput:
    print(ele)
    count += 1
    if count > 7:
        break
print("...")
```

```
No. of elements in predictionOutput =  200
Some elements of predictionOutput array are :-
A5
C3
9+11+2+4
0+6
C#3
11+1+4+5
C3
0+4+6
...
```

Analyzing Prediction :-

In [61]:

```
# Let's see, what our model has predicted
print("Measures of Dispersion of data :- \n")
print("Minimum value = ", np.amin(prediction))
print("Maximum value = ", np.amax(prediction))
print("Range of values = ", np.ptp(prediction))
print("Variance = ", np.var(prediction))
print("Standard Deviation = ", np.std(prediction))
print("Length of 1st Prediction-element = ", len(prediction[0]))
print("Count of unique elements = ", countNodes)
```

```
Measures of Dispersion of data :-

Minimum value =  5.747982e-12
Maximum value =  0.89277625
Range of values =  0.89277625
Variance =  0.0022270184
Standard Deviation =  0.047191296
Length of 1st Prediction-element =  359
Count of unique elements =  398
```

Generate Music out of Predicted-data :

What required is to get a Midi File :-

In [62]:

```
outputMusicElements = []
# Array to store notes & chords.
```

Trying to Create a Note (from string) :-

In [63]:

```
tempStr = 'C4'
# Just copying from the predictionOutput display
# Creating a note-object (using note-package)
note.Note(tempStr)
```

Out[63]:

```
<music21.note.Note C>
```

In [64]:

```
# Music-note is generated.
# Similarly we can do for multiple elements.
newNote = note.Note(tempStr)
# Also, the note will have a off-set (timing)
# By default, offset was 0. (setting it manually here)
newNote.offset = 0
# And, the note will have an instrument
#    Can set, using storedInstrument package
newNote.storedInstrument = instrument.Piano()
# outputMusicElements.append(newNote)
# Above element is commented out, as it will get unwanted music like this random created note.
```

In [65]:

```
print(newNote)
```

```
<music21.note.Note C>
```

Creating Music-Elements from String-array :-

In [66]:

```
offset = 0  # instance-time of particular element (note/chord)

for element in predictionOutput:
    # If element is a chord :-
    if('+' in element) or element.isdigit():
        notesInChord = element.split('+')
        tempNotes = []
        for currNote in notesInChord:
            newNote = note.Note(int(currNote))
            newNote.storedInstrument = instrument.Piano()
            tempNotes.append(newNote)
        # Create a chord-object from list of notes
        newChord = chord.Chord(tempNotes)
        newChord.offset = offset
        # Add this chord to music-elements
        outputMusicElements.append(newChord)
    # If element is a note :-
    else:
        newNote = note.Note(element)
        newNote.offset = offset
        newNote.storedInstrument = instrument.Piano()
        # Add this note to music-elements
        outputMusicElements.append(newNote)
    offset += 0.5
```

In [67]:

```
print("No. of elements in outputMusicElements = ", len(outputMusicElements))
print("Some elements of outputMusicElement array are :-")
count = 0
for ele in outputMusicElements:
    print(ele)
    count += 1
    if count > 7:
        break
print("...")
```

```
No. of elements in outputMusicElements =  200
Some elements of outputMusicElement array are :-
<music21.note.Note A>
<music21.note.Note C>
<music21.chord.Chord A B D E>
<music21.chord.Chord C F#>
<music21.note.Note C#>
<music21.chord.Chord B C# E F>
<music21.note.Note C>
<music21.chord.Chord C E F#>
...
```

Trying to Play the Output Music :-

In [68]:

```
# For playing them, have to create a stream-object
#    ..from the generated music-elements
midiStream = stream.Stream(outputMusicElements)
# Write this midiStream on a midi-file.
midiStream.write('midi', fp="testOutput8.mid")
# 1st arg  -->  File-type , 2nd arg  -->  File-name
```

Out[68]:

```
'testOutput8.mid'
```

Loading the output-midi file :

In [69]:

```
midiStream.show('midi')
# Show the music in playable-format
```

In [70]:

```
outputMusic8 = converter.parse("testOutput8.mid")
print(type(outputMusic8))
```

```
<class 'music21.stream.base.Score'>
```

In [71]:

```
outputMusic8.show('midi')
# This will show the music in playable-format
```

Plotting inputMusicElements VS outputMusicElements :

In [72]:

```
import matplotlib.pyplot as plt
```

In [73]:

```
inputMusicNums = networkInput[startIdx]
print("Some elements of inputMusicNums :-")
count = 0
for ele in inputMusicNums:
    print(ele)
    count += 1
    if count > 7:
        break
print("...")
```

```
Some elements of inputMusicNums :-
364
394
364
364
394
364
364
394
...
```

In [74]:

```
outputMusicNums = []
for ele in predictionOutput:
    outputMusicNums.append(noteMap[ele])
print("Some elements of outputMusicNums are :-")
count = 0
for ele in outputMusicNums:
    print(ele)
    count += 1
    if count > 7:
        break
print("...")
```

```
Some elements of outputMusicNums are :-
328
350
318
24
344
91
350
21
...
```

Plot inputMusicNums VS outputMusicNums (prediction visualization) :-

In [75]:

```
y1 = np.array(inputMusicNums)
y1 = y1[:100]
y2 = np.array(outputMusicNums)
y2 = y2[:100]
print(y1.shape)
print(y2.shape)
```

```
(100,)
(100,)
```

In [76]:

```
plt.rcParams["figure.figsize"] = [10.00, 5.50]
plt.rcParams["figure.autolayout"] = True
x1 = np.linspace(0,100,100)
x2 = np.linspace(101,200,100)

plt.plot(x1,y1,color="aqua",label="Input music-data")
plt.plot(x2,y2,color="orange",label="Predicted music-data")
```

```
plt.xlabel("Offset")
plt.ylabel("Music Element (index)")
plt.title("Testing data & Predicted data")
plt.show()
```