# Lab 1: Basic Audio Processing in Python

**Intagliata Emanuele (928166)**

**Surricchio Mattia (920715)**

**Link Colab Notebook:** https://colab.research.google.com/drive/1zgRDklv_4duwJT_YD0o2hm36ckBdiaR_

## IMPORTANT: The first step is always to SAVE A COPY OF THIS NOTEBOOK in your own Google Drive and do the work on your own document. (File --> Save a copy in Drive)

Python is a general-purpose programming language that is popular and easy to use. For new programmers, it is a great choice as a first programming language. In fact, more and more university CS departments are centering their introductory courses around Python.

**The Python Tutorial**

If you are new to Python, you can follow this Python Tutorial.

For the purposes of this course, be sure you understand the following sections:

Section 3: *An Informal Introduction to Python*

- numbers: int, float
- strings
- lists

Section 4: *More Control Flow Tools*

- if statements
- for statements
- range
- functions

Section 5: *Data Structures*

- list comprehensions
- tuples
- dictionaries
- looping techniques

Other resources:

- The Hitchhiker's Guide to Python
- SciPy Lecture Notes
- SciPy Lecture Notes
- learnpython.org

### Exercise 1: Basic operations with arrays

Use numpy to do the following array operations:

Create a row vector v1: $[1, 2, 3]$

In [ ]:

```
#imports
import numpy as np
import matplotlib.pyplot as plt
import librosa
```

**In the following cell we're creating a simple array. It is important to add extra square brackets in its declaration. Declaring the array only as `np.array([1,2,3])` will generate an array with a single dimension: its shape will be `(3,)`.**
**This result will prevent us from any matrix computation, thus we're adding extra square brackets in order to obtain a vector with shape `(1,3)`, which is still one dimensional, but it's interpreted differently in Python with respect to `(3,)`**

In [ ]:

```
import numpy as np

#Here i'm adding extra square brackets to create a 2D array and then use the transpose.
#Creating a standard array with np.array([1,2,3]) creates a 1D element which is not trans
posable with np.transpose()
v1 = np.array([[1,2,3]])
print("Array shape:",v1.shape)
```

```
Array shape: (1, 3)
```

**Create a column vector v2:** $[1, 2, 3]^T$

**We're applying the `transpose` function of `numpy` to our array.**

In [ ]:

```
v2 = np.transpose(v1)
print("Array shape:",v2.shape)
```

```
Array shape: (3, 1)
```

**Create a matrix M of 3 x 3 elements:**

**Now we're declaring a simple 3x3 matrix. The `array` function of `numpy` takes as input tuples of elements between square brackets, this represents the content of a row. Each row is splitted by a comma.**

In [ ]:

```
matrix = np.array([[1,2,3], [4,5,6], [7,8,9]])
print(matrix)
```

```
[[1 2 3]
 [4 5 6]
 [7 8 9]]
```

**Show the first row of M:**

**Now we're using array slicing to select a single row and then a single column. The `:` operator selects all the element in that dimension.**

In [ ]:

```
print(matrix[0,:])
```

```
[1 2 3]
```

**Show the first column of M:**

In [ ]:

```
print(matrix[:,0])
```

```
[1 4 7]
```

**Perform the matrix product v2 times v1:**

**Performing the dot product between the 2 arrays using the `dot` function of `numpy`. Since `v1` has shape `(1,3)` and `v2` has shape `(3,1)` the dot product returns a single scalar value.**
**From the theory, given a matrix of NxM and MxN, the result shape of the dot product between these two matrices is a NxN matrix.**

In [ ]:

```
result = v1.dot(v2)
print(result)
```

```
[[14]]
```

**Perform the matrix product v2 times v1:**

In [ ]:

```
result = v2.dot(v1)
print(result)
```

```
[[1 2 3]
 [2 4 6]
 [3 6 9]]
```

**Perform the element-wise multiplication of v2 and v1, with result having the shape of v2:**

**We're performing di element-wise product using the `multiply` function of `numpy`. We're using `expand_dims()` in order to add an extra dimension to `result` and cast the array from `(3,)` shape to `(3,1)` shape**

In [ ]:

```
result = np.multiply(v2[:,0],v1[0,:])
result = np.expand_dims(result, axis=1)
print('Result: ')
print(result)
print('The result has shape: ')
print(result.shape)
```

```
Result:
[[1]
 [4]
 [9]]
The result has shape:
(3, 1)
```

**Perform the element-wise multiplication of v1 and v2, with result having the shape of v1:**

**Same process as the previous cell**

In [ ]:

```
result = np.multiply(v2[:,0],v1[0,:])
result = np.expand_dims(result, axis=0)
print('Result: ')
print(result)
print('The result has shape: ')
print(result.shape)
```

```
Result:
[[1 4 9]]
The result has shape:
(1, 3)
```

**Generate an array of 20 equally-spaced numbers between 0 and 10:**

**We're using the `linspace` function in order to generate 20 equally spaced elements between 0 and 10. The first two arguments select start and end of the space, the third one select the number of equally spaced elements in the given interval.**

In [ ]:

```
lin_space = np.linspace(0, 10, num=20)
print(lin_space)
```

```
[ 0.          0.52631579  1.05263158  1.57894737  2.10526316  2.63157895
  3.15789474  3.68421053  4.21052632  4.73684211  5.26315789  5.78947368
  6.31578947  6.84210526  7.36842105  7.89473684  8.42105263  8.94736842
  9.47368421 10.        ]
```

**Generate an array of numbers going from 0 to 1.5 in steps of 0.2:**

**Same thing as the previous cell, but the `arange` function takes as third argument the step size instead of the number of equally spaced elements inside the given range**

In [ ]:

```
lin_space = np.arange(0,1.5,0.2)
print(lin_space)
```

```
[0.  0.2 0.4 0.6 0.8 1.  1.2 1.4]
```

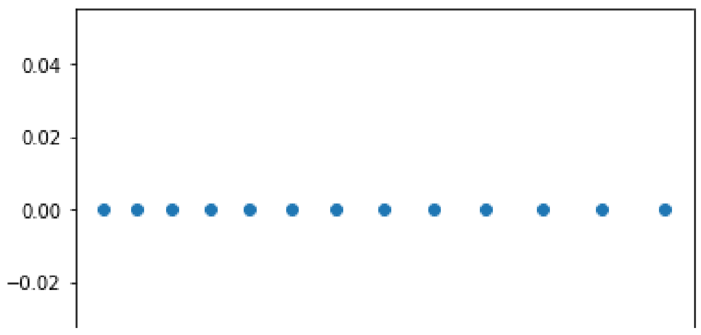
**Generate an array of 13 logarithmically-spaced numbers between 1 and 2 (both included) using the numpy function logspace()**

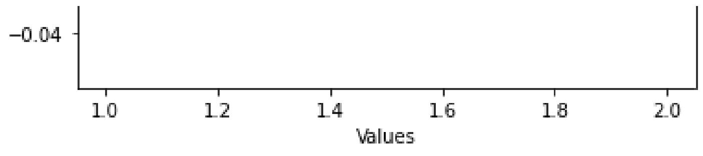
**logspace is used to generate 13 elements in the given range. `logspace(n,m,k)` returns a logspace between `base**n` and `base**m`. Since we're interested in a space between 1 and 2, we're adding `np.log10()` before the arguments.**

In [ ]:

```
log_space = np.logspace(np.log10(1), np.log10(2), num=13, base=10)
y = np.zeros(13)
print('printing values:',log_space)
plt.plot(log_space,y, 'o');
plt.xlabel('Values');
```

```
printing values: [1.         1.05946309 1.12246205 1.18920712 1.25992105 1.33483985
 1.41421356 1.49830708 1.58740105 1.68179283 1.78179744 1.88774863
 2.        ]
```



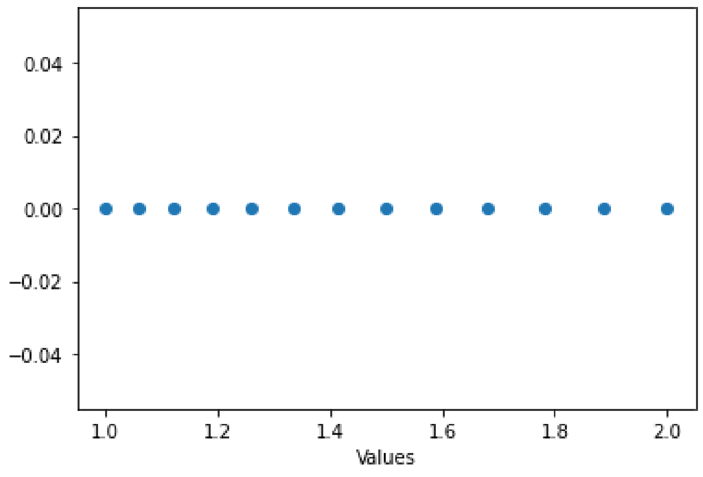


**Generate the same array of 13 logarithmically-spaced numbers using fractional powers of 2:**

In [ ]:

```
log_space = np.logspace(np.log2(1), np.log2(2), num=13, base=2)
y = np.zeros(13)
print('printing values:',log_space)
plt.plot(log_space,y, 'o');
plt.xlabel('Values');
```

```
printing values: [1.         1.05946309 1.12246205 1.18920712 1.25992105 1.33483985
 1.41421356 1.49830708 1.58740105 1.68179283 1.78179744 1.88774863
 2.        ]
```



**The generated array divides logarithmically an octave and can be used to generate fundamental frequencies for the pitches:**

| Index | 0 | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 | 10 | 11 | 12 |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| Pitch | A | A# | B | C | C# | D | D# | E | F | F# | G | G# | $A_2$ |

**Use as a starting fundamental frequency, the one corresponding to A (440 Hz), and generate an array with the fundamental frequencies of each pitch:**

In [ ]:

```
pitches = 440*log_space
print(pitches)
```

```
[440.         466.16376152 493.88330126 523.2511306  554.36526195
 587.32953583 622.25396744 659.25511383 698.45646287 739.98884542
 783.99087196 830.60939516 880.        ]
```

## Exercise 2: Generation of notes

**Assume a sampling frequency of fs = 16000 Hz, and generate a time vector with sampling instants between 0 s and 2 s:**

In [ ]:

```
fs=16000
time = np.arange(0,2,step=1/fs)
time.shape #checked array dimension, seems coherent
```

Out[ ]:

```
(32000,)
```

**Generate an array containing the envelope shape:**

$$a(t) = Ae^{-t/\tau}$$

**with $\tau = 0.4$, $A = 0.5$ and $t$ corresponding to a given time instant:**

In [ ]:

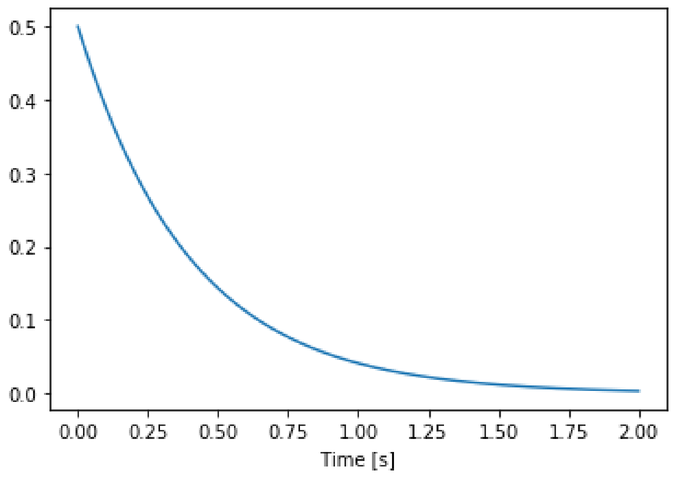
```
tau = 0.4
A = 0.5
a = A*np.exp(-time/tau)
print(a.shape)
```

```
(32000,)
```

**Plot the envelope using Matplotlib:**

In [ ]:

```
import matplotlib.pyplot as plt

plt.plot(time,a);
plt.xlabel(' Time [s] ');
```



**Generate a sawtooth wave by using scipy.signal to generates notes with the model**

$$x(t) = a(t)\text{sawtooth}(2\pi f_k t)$$

**where $a(t)$ is the envelope (see above) and $f_k$ the fundamental frequencies of the $k$-th pitch as in the table of Exercise 1.**

**Generate note signals for pitches A, C#, E and A $_2$:**

**The following cell simply generates 4 different notes using the `signal` function from `scipy`. The `sawtooth` is generated by passing the `frequency pitch`, the `time` axis and `2*pi`. Then the result is multiplied by the envelope `a`, which will change the amplitude of our wave**

In [ ]:

```
from scipy import signal
notes = {}
A1_note = a*signal.sawtooth(2 * np.pi * pitches[0] * time)
print(A1_note.shape)
C_sharp_note = a*signal.sawtooth(2 * np.pi * pitches[4] * time)
```

```
E_note = a*signal.sawtooth(2 * np.pi * pitches[7] * time)
A2_note = a*signal.sawtooth(2 * np.pi * pitches[12] * time)
```

```
(32000,)
```

**Listen to each of the notes by importing "IPython.display as ipd":**

In [ ]:

```
import IPython

#solved the proble of multiple displays https://stackoverflow.com/questions/33048353/more
-than-one-audio-object-in-a-jupyter-ipython-notebook-cell
IPython.display.display(IPython.display.Audio(A1_note,rate=fs))
IPython.display.display(IPython.display.Audio(C_sharp_note,rate=fs))
IPython.display.display(IPython.display.Audio(E_note,rate=fs))
IPython.display.display(IPython.display.Audio(A2_note,rate=fs))
```

**Your browser does not support the audio element.**

**Your browser does not support the audio element.**

**Your browser does not support the audio element.**

**Your browser does not support the audio element.**

**Generate a vector arpegio by concatenating the four notes:**

In [ ]:

```
arpeggio = np.concatenate((A1_note, C_sharp_note, E_note, A2_note))
IPython.display.display(IPython.display.Audio(arpeggio,rate=fs))
```

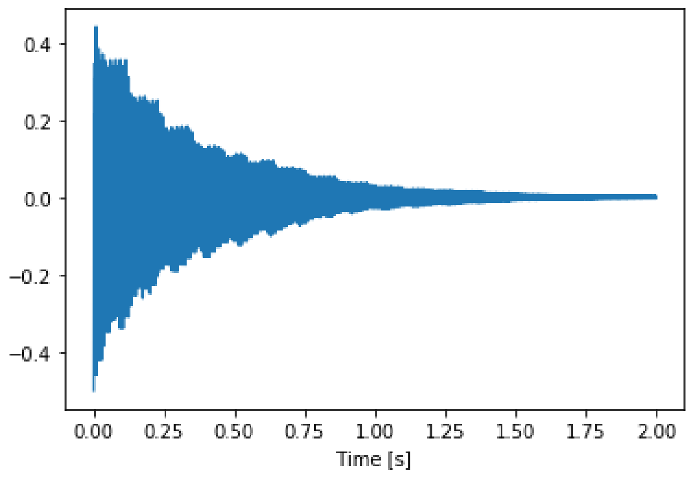
**Your browser does not support the audio element.**

**Generate a vector chord made by summing up the four notes and dividing them by 4 to avoid clipping:**

In [ ]:

```
temp_matrix = np.stack((A1_note, C_sharp_note, E_note, A2_note))
chord = temp_matrix.sum(axis=0)
chord = chord/4
plt.plot(time,chord)
plt.xlabel(' Time [s] ');
IPython.display.display(IPython.display.Audio(chord,rate=fs))
```

**Your browser does not support the audio element.**



## Exercise 3: Linking Colab to Google Drive and Uploading files to Colab

**To link your Google Drive to Colab, use the following commands and follow the instructions. You will have to go to the link to retrieve the authorization code.**

In [ ]:

```
from google.colab import drive
drive.mount('/content/gdrive')
```

```
Mounted at /content/gdrive
```

**If you are able to access Google Drive, your google drive files should be all under: $/content/gdrive/My Drive/**

**You should now have access to any file in your Google Drive. Additionally, you can upload files to the Colab Running Environment by using the interface on the left side.**

**Note: The files uploaded to Colab (that are not within your Google Drive folder) will be deleted upon disconnection.**

**Upload the provided file 'OSR_us_000_0010_8k.wav' via any of the above methods (direct uploading or Google drive access):**

## Exercise 4: STFT Representation

**Now that the file has been uploded, read it with librosa.load, show its sampling rate and its duration.**

In [ ]:

```
import librosa
import os

#----- Use this for local files (not on Colab)
#cwd = os.getcwd()
#file =  'OSR_us_000_0010_8k.wav'
#path = os.path.join(cwd,file)
#-----

signal, sample_rate = librosa.load('/content/gdrive/My Drive/Lab 1 files/OSR_us_000_0010_
8k.wav', sr=None)
print("Sampling rate:", sample_rate)
signal_length = signal.shape[-1]
print("Signal Length in seconds:",signal_length/sample_rate,"s")
```

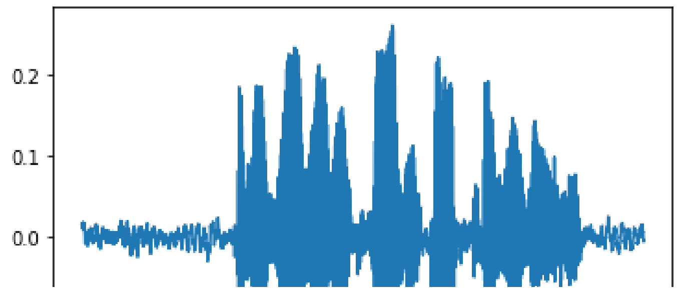
```
Sampling rate: 8000
Signal Length in seconds: 33.623125 s
```

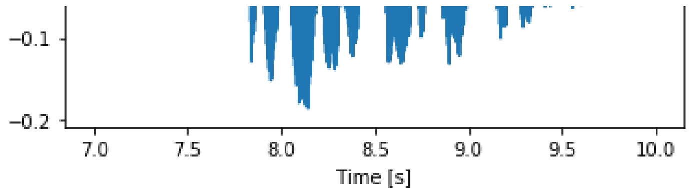
**Cut the signal to show only the third spoken sentence (from t=7 s to t = 10 s). Represent it with respect to time in seconds.**

**The signal is cut between 7s and 10s. The seconds value is multiplied by `sample_rate` in order to get an array index for our signal vector.**

In [ ]:

```
cut_signal = signal[7*sample_rate:10*sample_rate]
time = np.arange(7,10,1/sample_rate)
plt.plot(time,cut_signal);
plt.xlabel(' Time [s] ');
```





**Create a function "enframe" that takes as input:**

- **signal (one-dimensional)**
- **frame length (in samples)**
- **hop size (in samples)**

**and returns a matrix where each column stores a frame of the signal. The function should pad the signal with zeroes if necessary.**

In [ ]:

```
#Ask if we have to pass the cut signal or the complete one

def enframe(signal, frame_length, hop_size):
    signal_length = len(signal)
    #computes the number of frames generated with those inputs in my function
    num_frames = int(np.ceil((signal_length - frame_length)/hop_size)+1)
    #computes the length of the padded signal
    pad_signal_length = (num_frames-1)*hop_size + frame_length
    #generating a zero elements array as big as the difference between pad_signal_length
and signal
    z = np.zeros((pad_signal_length - signal_length))
    #concatenate the padding
    padded_signal = np.append(signal,z)
    #first generates an array of elements from 0 to frame length with step = 1
    #then the tile function generates num_frame copies of this array and stacks them one
on top of the other
    #for each column I have an index that goes from zero to frame length
    inframe_ind = np.tile(np.arange(0, frame_length), (num_frames,1)).T
    #Here it's the same as above, but in each row I have an array of values that goes
    #from zero to num_frames*hop_size with  hop_size steps
    frame_ind = np.tile(np.arange(0, num_frames*hop_size, hop_size), (frame_length,1))
    #Here i get the full indexes matrix. Basically the main idea is to "split" the indexe
s in two different matrixes
    #inframe_ind is the relative indexing inside a single frame, so from 0 to frame_lengt
h
    #frame_ind is the starting point of the n-th frame. So summing relative index + frame
starting point
    #i get an absolute index over the original samples of the signal
    indices = inframe_ind+frame_ind

    frames = padded_signal[indices]
    return frames
```

**Divide the signal into frames of length 512, using a hop size of 256 samples. Show the dimensions of the output.**

In [ ]:

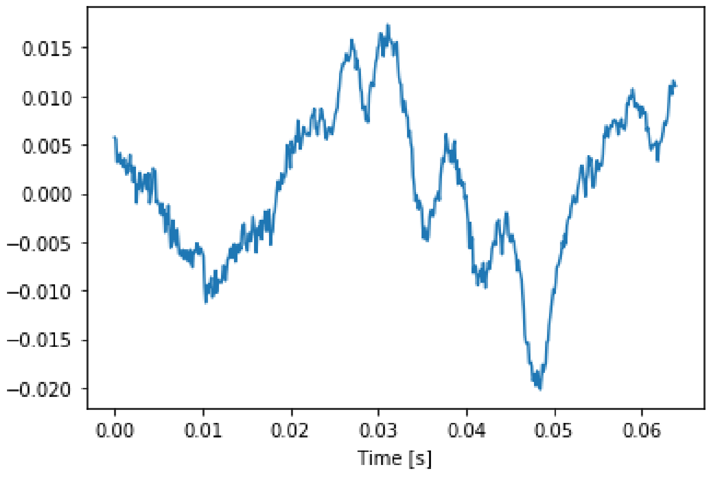
```
frame_length = 512
hop_size = 256
output = enframe(cut_signal, frame_length, hop_size)
print(output.shape)
```

```
(512, 93)
```

**Plot the content of the 10-th frame:**

`output[:,10]` **selects the 10 column. Then the time axis is generated by dividing the sample index by the** `sample_rate`

```
#check if needed time x-axis or not
plt.plot(np.arange(0,frame_length)*1/sample_rate,output[:,10]);
plt.xlabel(' Time [s] ');
```



**Create a Hamming function and apply it to each of your frames:**

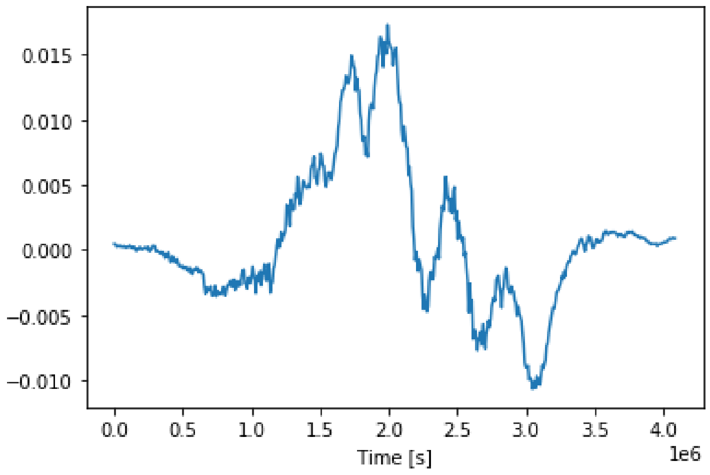
In [ ]:

```
winfunc = np.hamming(frame_length)
winfunc = np.expand_dims(winfunc,1) #to obtain a column vector in order to perform colum
n wise multiplication

windowed_signal = output*winfunc
```

**Plot again the 10-th frame after having applied your Hamming window:**

In [ ]:

```
plt.plot(np.arange(0,frame_length)*sample_rate,windowed_signal[:,10]);
plt.xlabel(' Time [s] ');
```



**Compute the DFT of each column (frame) using numpy's rfft() function. Use as number of FFT points the same value as window length.**

In [ ]:

```
NFFT = len(winfunc)

mag_frames = np.abs(np.fft.rfft(windowed_signal, NFFT, axis=0))
pow_frames = (mag_frames)**2
Nbins = mag_frames.shape[0]
f_bins = np.arange(0,Nbins)*sample_rate/NFFT
```

**Use the following function to represent the result with the indices of each frame and each frequency bin in the horizontal and vertical axes, respectively**

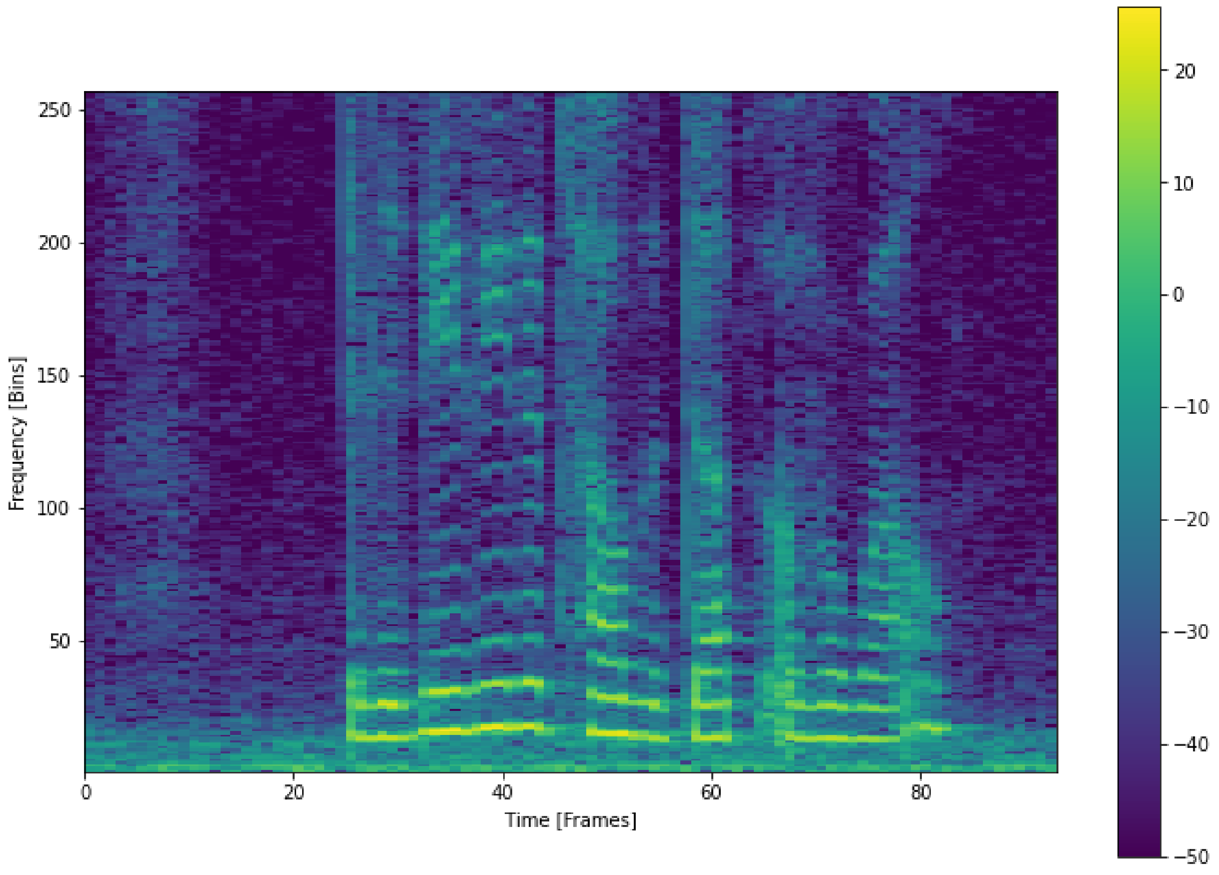
horizontal and vertical axes, respectively.

In [ ]:

```python
def show_specgram(X,ratio,limits,cmin):
    '''This function represents matrix X within a figure having the desired
   aspect ratio. Input:
   X: matrix to represent
   ratio: aspect ratio of axes
   limits: array with [xmin, xmax, ymax, ymin] (y axis is reversed)
   cmin: minimum value corresponding to the bottom of the colormap scale.
    '''
   w, h = plt.figaspect(ratio)
   fig = plt.figure(figsize=(w*2, h*2))
   ax = fig.add_axes([0.1, 0.1, 0.8, 0.8])
   im = ax.imshow(X, extent = limits, cmap='viridis', interpolation='none')
   xleft, xright = ax.get_xlim()
   ybottom, ytop = ax.get_ylim()
   ax.set_aspect(abs((xright-xleft)/(ybottom-ytop))*ratio)
   ax.invert_yaxis()
   if cmin != None:
     cbar = plt.colorbar(im);
     cbar.mappable.set_clim(cmin, None);
   return ax
```

In [ ]:

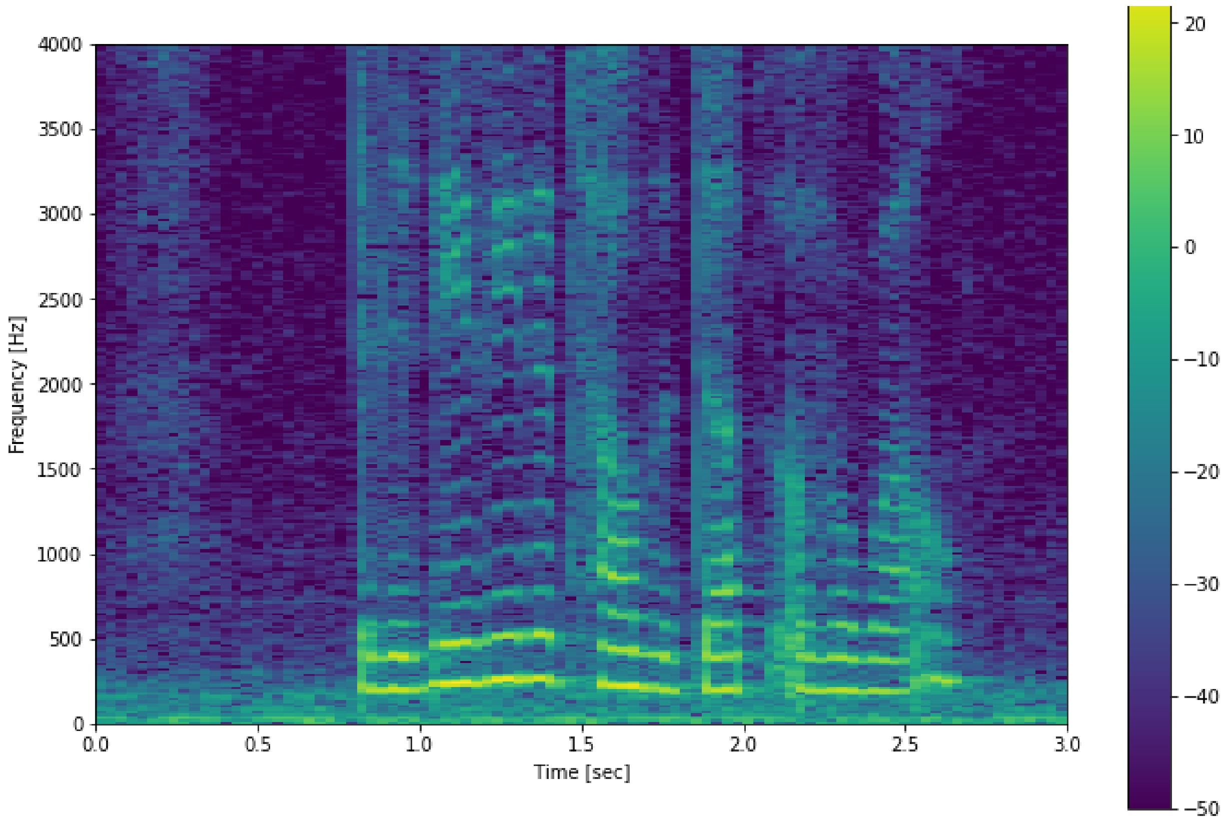
```python
show_specgram(10*np.log10(pow_frames), 0.7, [0, windowed_signal.shape[1], Nbins, 1], -50
);
plt.xlabel(' Time [Frames] ');
plt.ylabel(' Frequency [Bins] ');
```



**Finally, represent again the spectrogram but as a function of the time in seconds and frequency in Hz.**

In [ ]:

```python
show_specgram(10*np.log10(pow_frames), 0.7, [0, 3, f_bins[-1], 0], -50);
plt.xlabel(' Time [sec] ');
plt.ylabel(' Frequency [Hz] ');
```

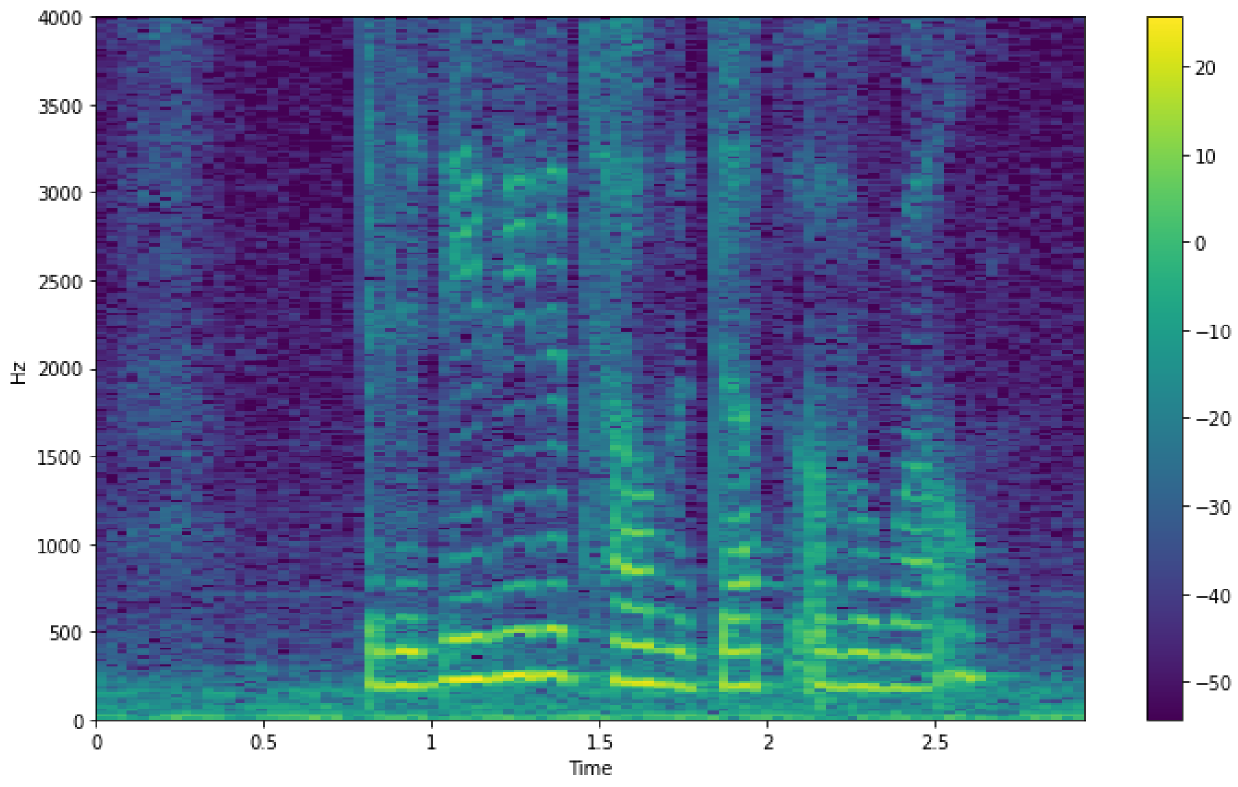


### Exercise 5: Librosa representations (STFT)

**Use librosa to represent the power spectrogram of the signal, as we computed manually in the last exercise.**

In [ ]:

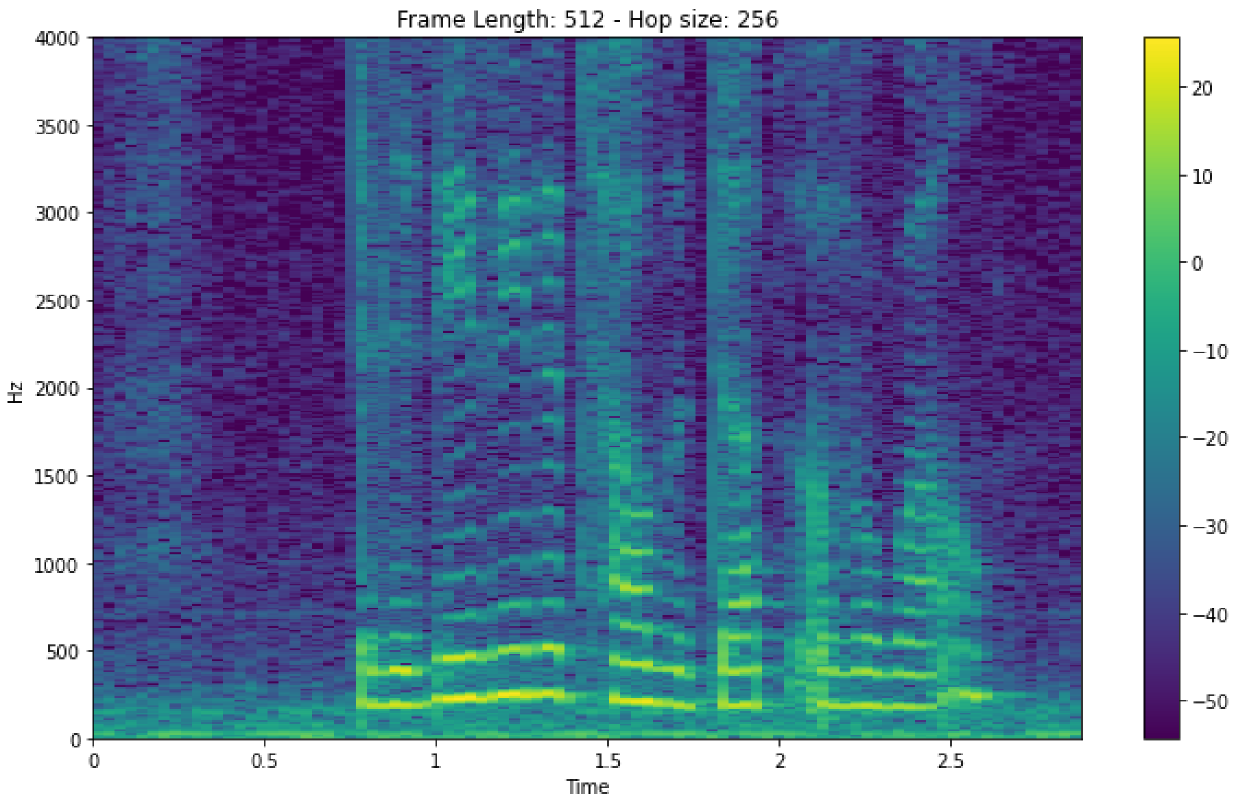
```
import librosa.display
X = librosa.stft(cut_signal, n_fft = NFFT, hop_length = hop_size, win_length=frame_lengt
h, window='hamming', center = False)
Xmag = librosa.amplitude_to_db(np.abs(X))
plt.figure(figsize=(12,7))
librosa.display.specshow(Xmag, sr =sample_rate, hop_length=hop_size, x_axis = 'time', y_
axis = 'linear', cmap = 'viridis' )
plt.colorbar();
```

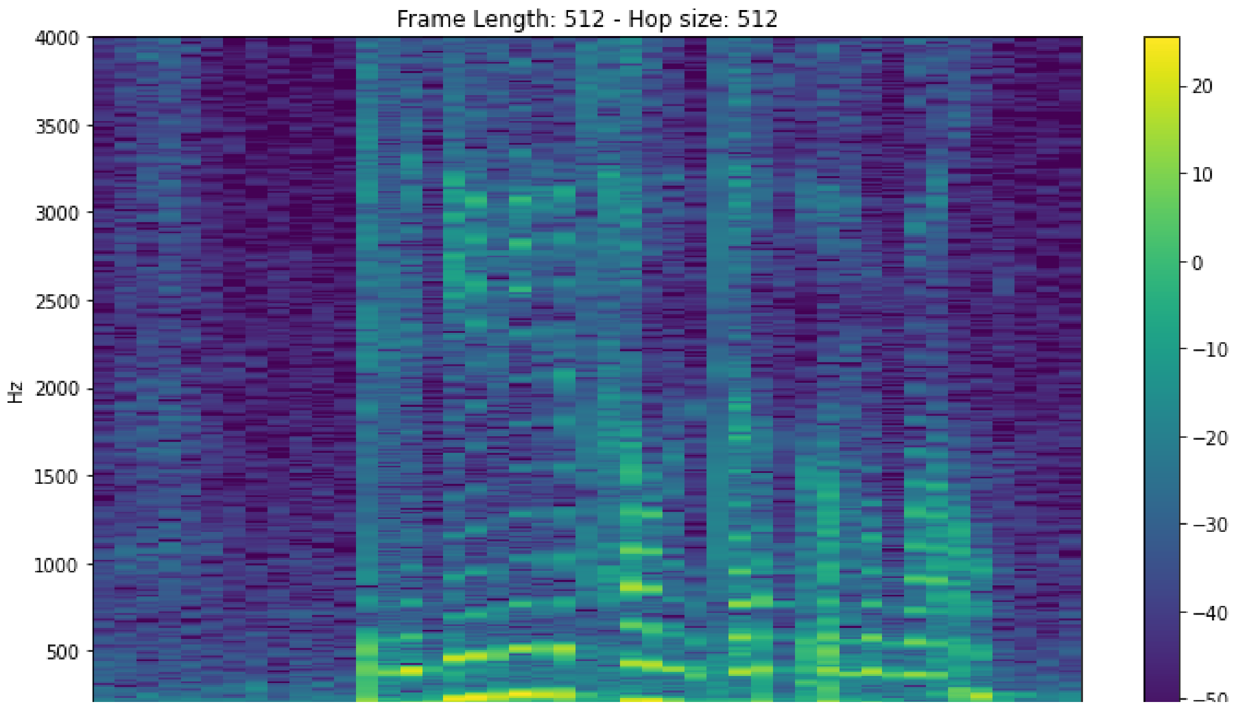


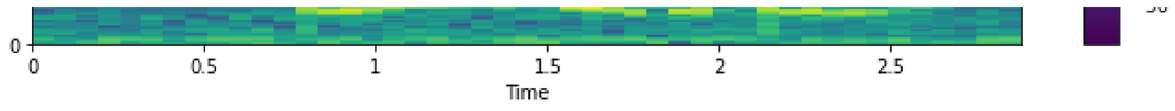
**Experiment with different values of frame length and hop size and discuss the results.**

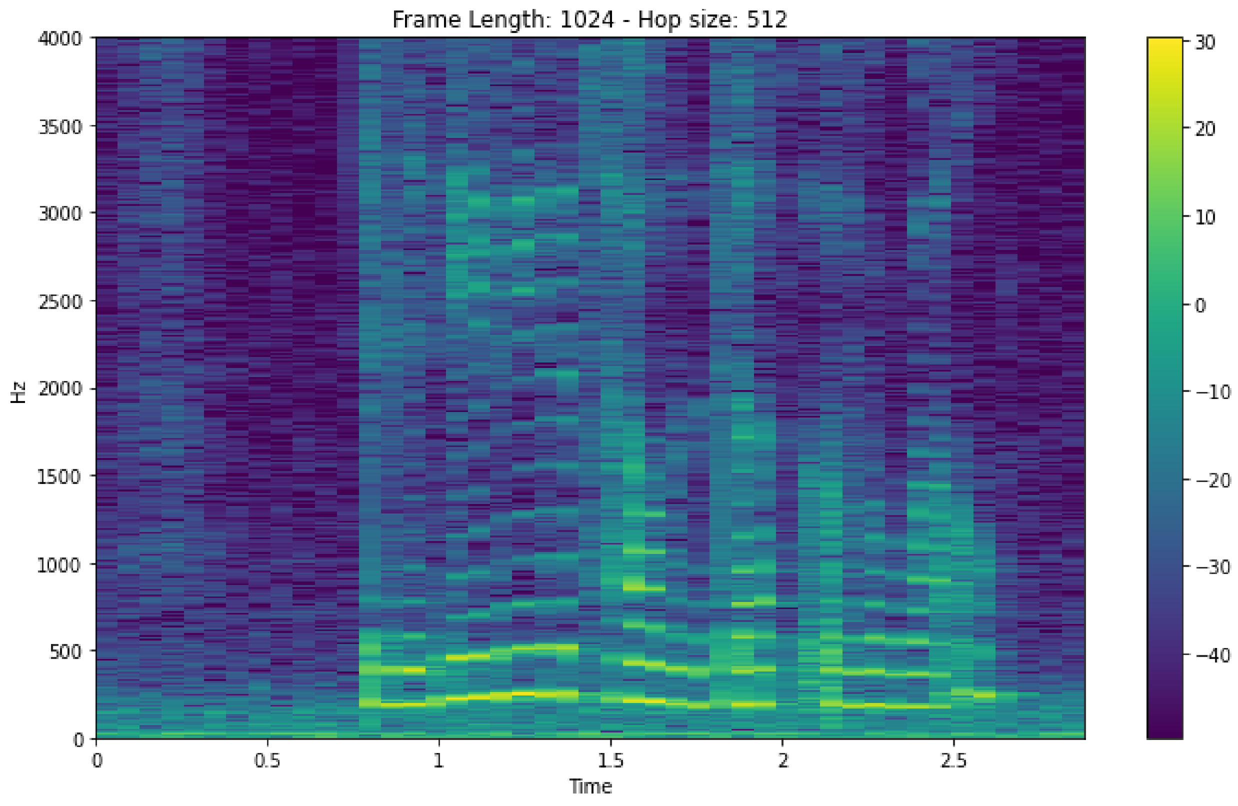
In [ ]:

```
import librosa.display
#changing the frame_length changes the resolution of the FFT (more bins)
#changing the hopsize changes the spatial resolution

frame_length_array = [512, 512, 1024]
hop_size_array = [256, 512, 512]

for i in range(0,3):
  X = librosa.stft(cut_signal, n_fft = 1024, hop_length = hop_size_array[i], win_length=
frame_length_array[i], window='hamming', center = False)
  Xmag = librosa.amplitude_to_db(np.abs(X))
  plt.figure(figsize=(12,7))
  librosa.display.specshow(Xmag, sr=sample_rate, hop_length=hop_size_array[i], x_axis =
'time', y_axis = 'linear', cmap = 'viridis' );
  plt.colorbar();
  plt.title('Frame Length: {} - Hop size: {}'.format(frame_length_array[i], hop_size_arr
ay[i]))
```









***Changing the frame length changes the frequency resolution of the FFT (more bins). An higher frame length corresponds to a lower frequency resolution. Changing the hopsize changes the temporal resolution. A bigger hop size corresponds to a lower temporal resolution.***

## Exercise 6: Mel-Spectrograms

**Compute the mel spectrogram using librosa. Experiment with different values of the parameters and discuss the results.**

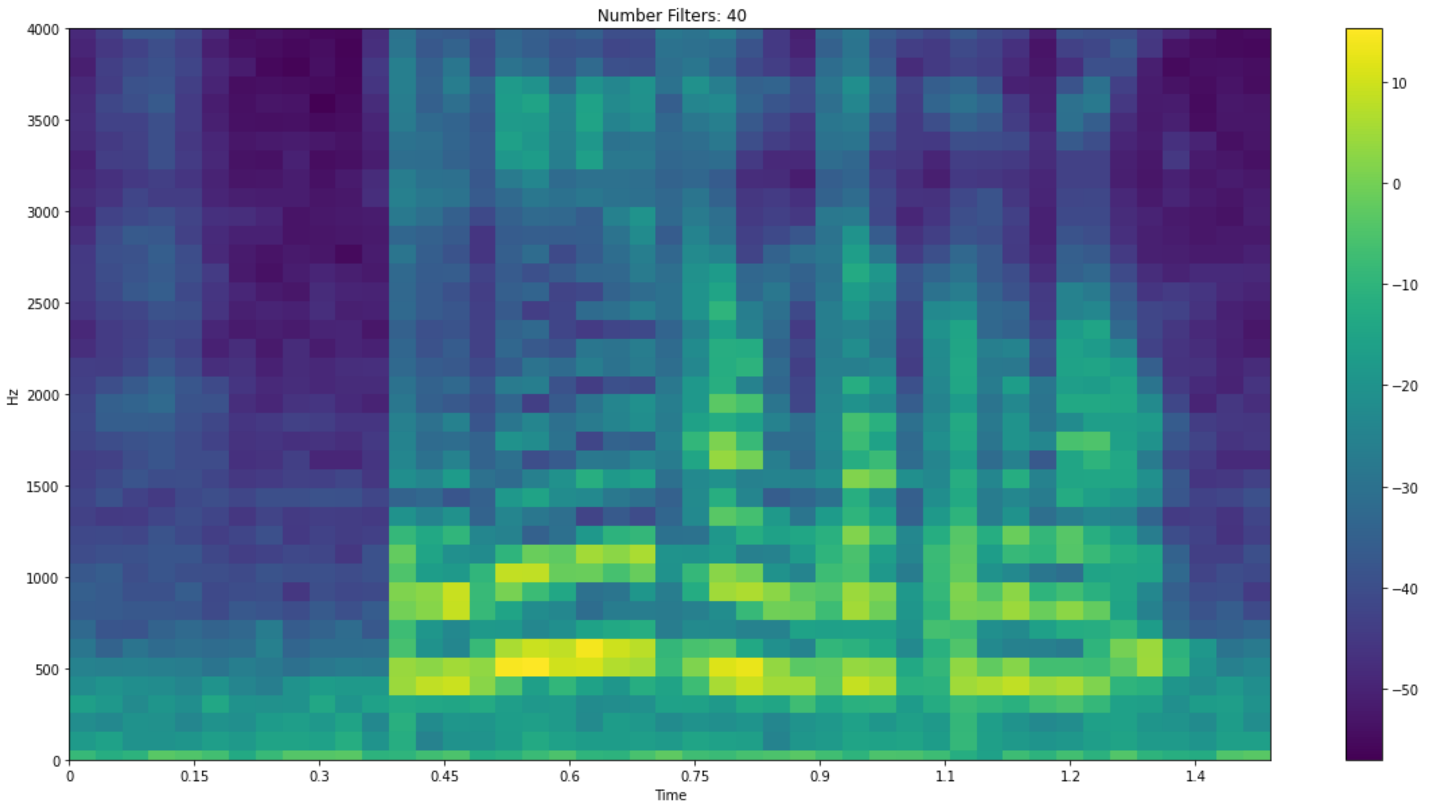
In [ ]:

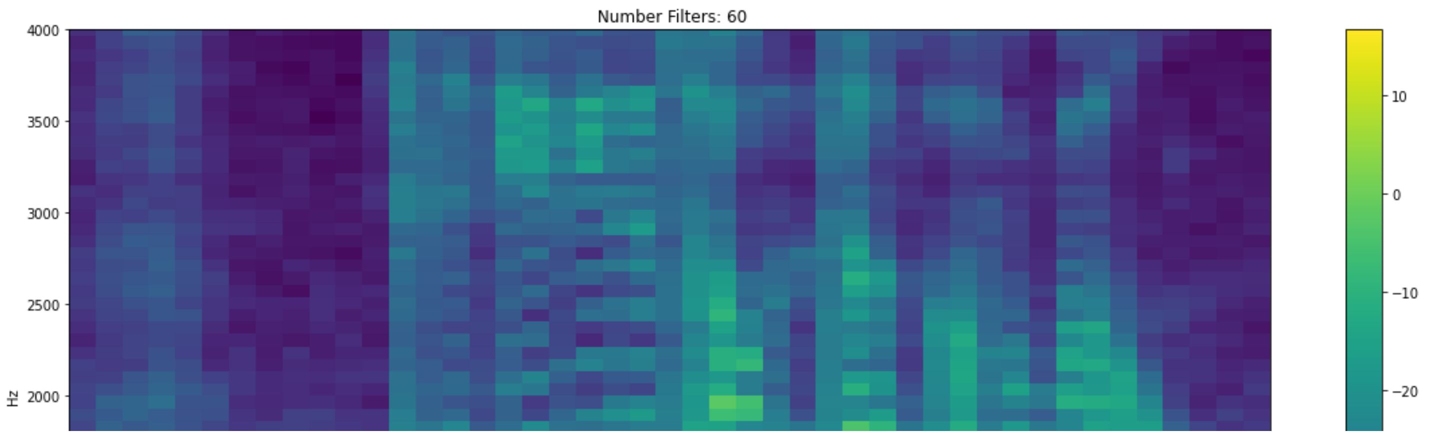
```
# First of all, I define some useful functions

# This converts a frequency from Hz to mels
def Hz_to_Mel(f_hz):
  return (2595 * np.log10(1 + (f_hz / 700)))

# This converts a frequency from mels to Hz
def Mel_to_Hz(f_mel):
  return (700 * (10**(f_mel / 2595) - 1 ))

# This function creates a filterbank
def Mel_filters(nfilt, f_min, f_max, numfft, Fs):
  low_freq_mel = Hz_to_Mel(f_min)
  high_freq_mel = Hz_to_Mel(f_max)

  mel_points = np.linspace(low_freq_mel, high_freq_mel, nfilt + 2) #+2 because for n fil
ters I need n+2 points
  hz_points = Mel_to_Hz(mel_points)
  bin = np.floor((numfft + 1) * hz_points / Fs) # corresponding bin in the DFT (NFTT poi
nts) for frequencies in hz_points

   # Inizialize filterbank
   fbank = np.zeros((nfilt,int(np.floor(numfft/2 +1))))
   for m in range(1, nfilt + 1):
     f_m_minus = int(bin[m-1])
     f_m = int(bin[m])
```

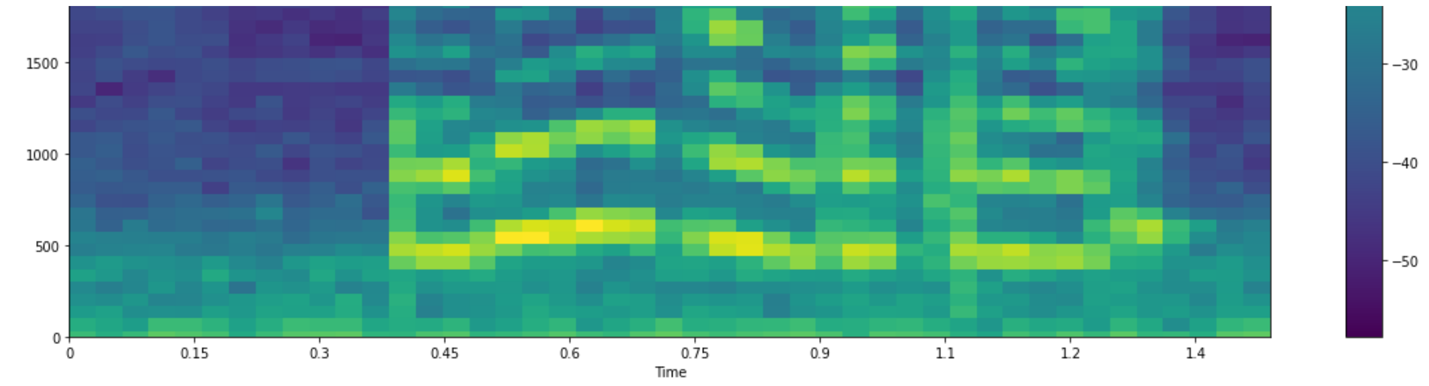
```
    f_m_plus = int(bin[m+1])

    for k in range(f_m_minus, f_m):
      fbank[m - 1, k] = (k - f_m_minus) / (f_m - f_m_minus)

    for k in range(f_m, f_m_plus):
      fbank[m - 1, k] = (f_m_plus - k) / (f_m_plus - f_m)

  return fbank
```

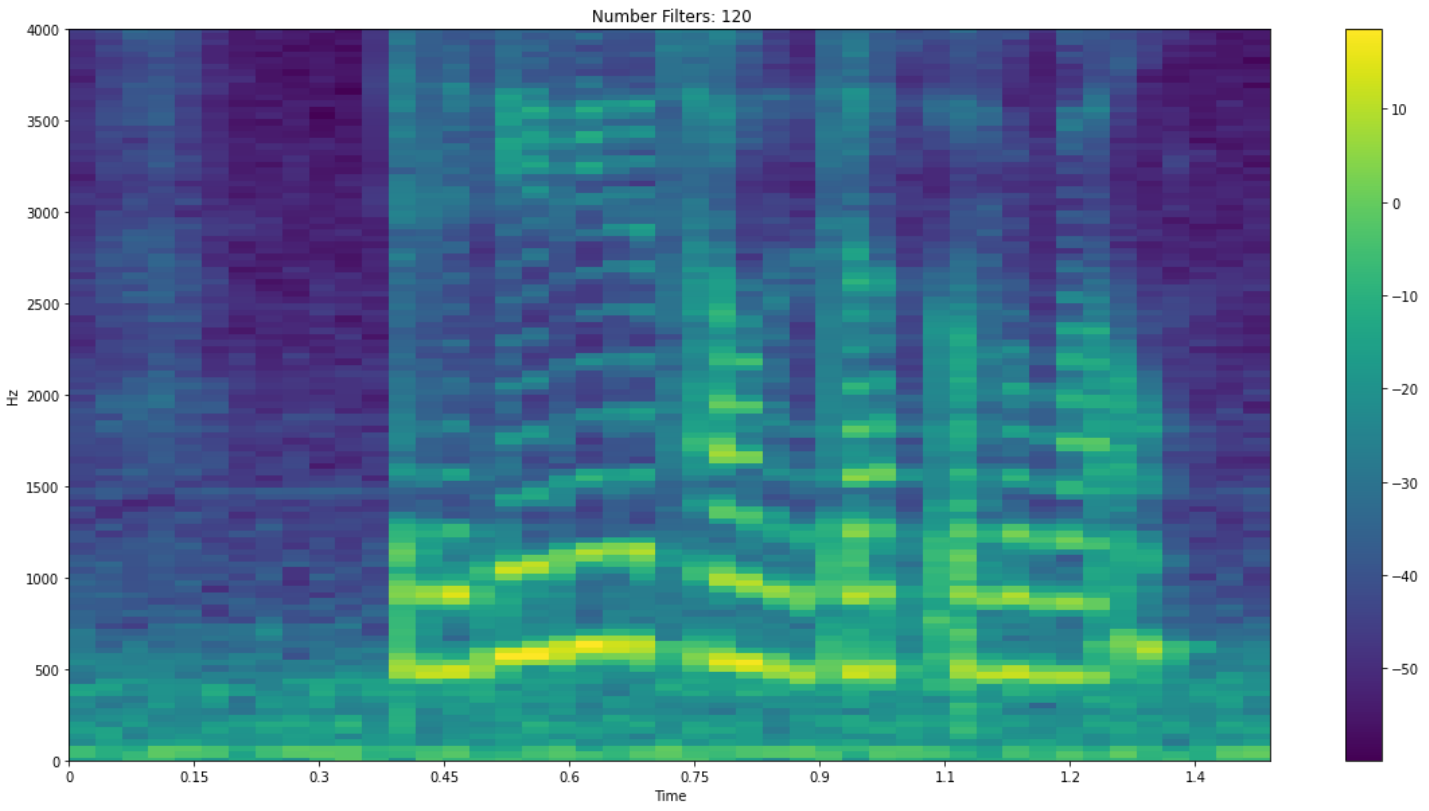
In [ ]:

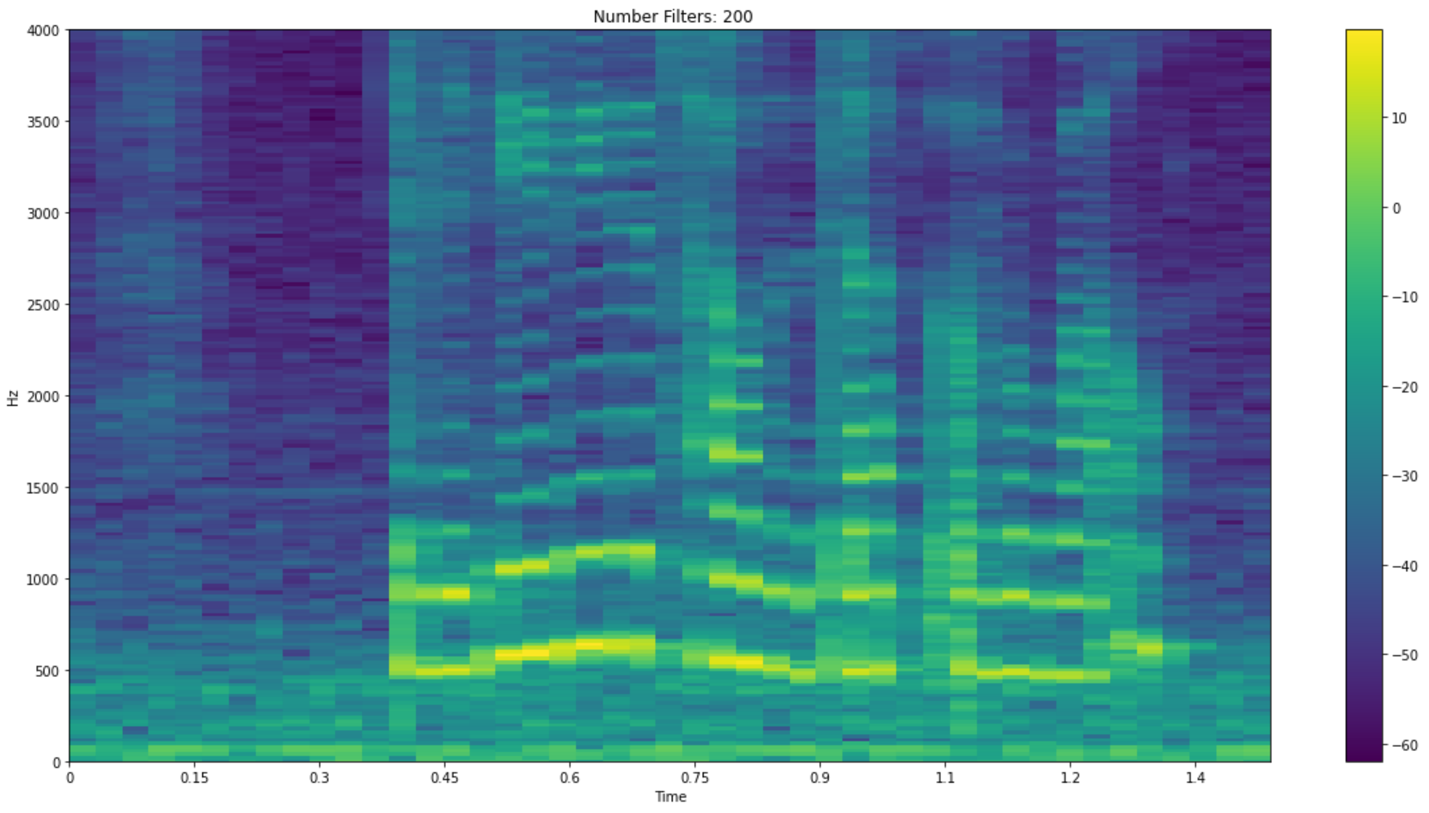
```
nfilt_array = [40, 60 ,120, 200]
fmin = 0
fmax = sample_rate/2
fbank = Mel_filters(nfilt, fmin, fmax, NFFT, sample_rate)


for i in range (0,4):
  nfilt = nfilt_array[i]
  # Let's compute the mel spectrogram (in librosa)
  X_mel_lib = librosa.feature.melspectrogram(sr=sample_rate, S=np.abs(X)**2, n_fft = NFF
T, hop_length= hop_size, n_mels = nfilt, fmin = 0, fmax= sample_rate/2, htk= True)

  # Plot
  plt.figure(figsize=(20,10));
  librosa.display.specshow(10*np.log10(X_mel_lib), sr=sample_rate, hop_length=hop_size,
x_axis = 'time', y_axis = 'linear', cmap = 'viridis' );
  plt.colorbar()
  plt.title(' Number Filters: {}'.format(nfilt))
```











***Changing the number of filters changes the frequency resolution of the FFT (more bins). An higher number of filters corresponds to a lower frequency resolution.***

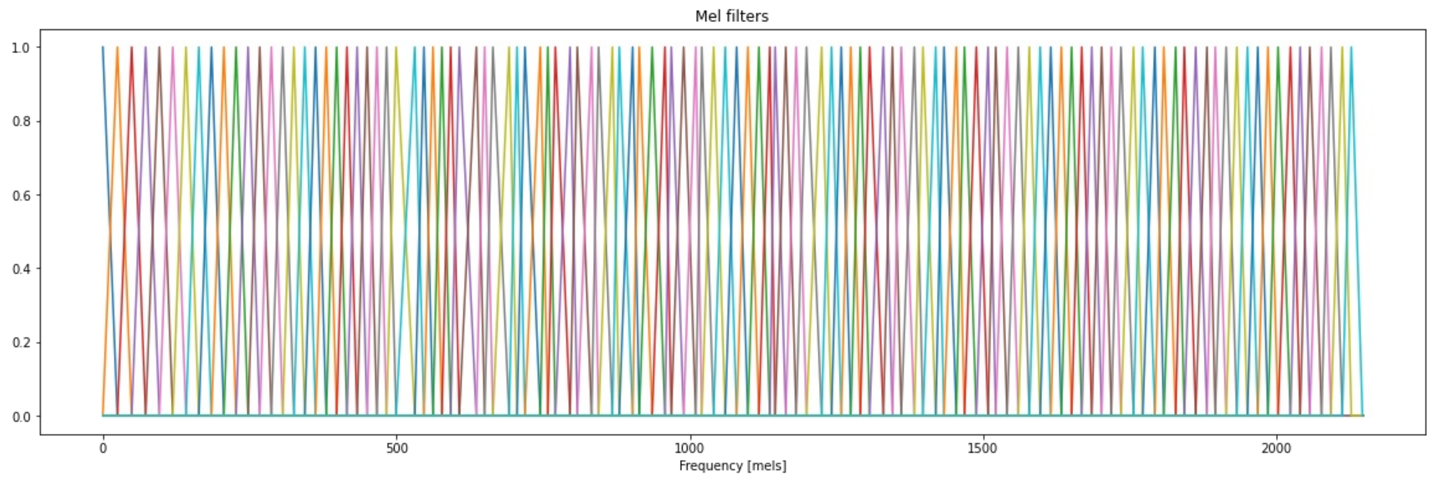
**Show the Mel-scale filterbank used in your last representation:**

In [ ]:

```
plt.figure(figsize=(20,6));
plt.plot(Hz_to_Mel(f_bins), fbank.T)
plt.xlabel('Frequency [mels]')
plt.title('Mel filters')
```

Out[ ]:

```
Text(0.5, 1.0, 'Mel filters')
```

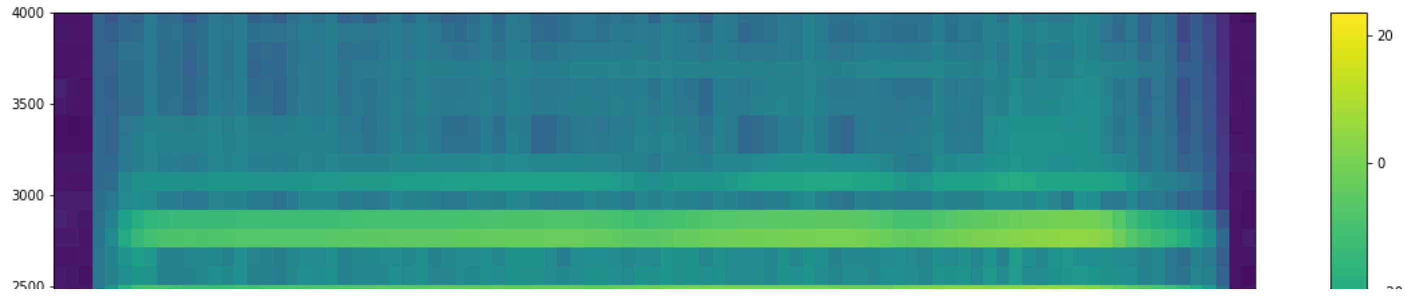


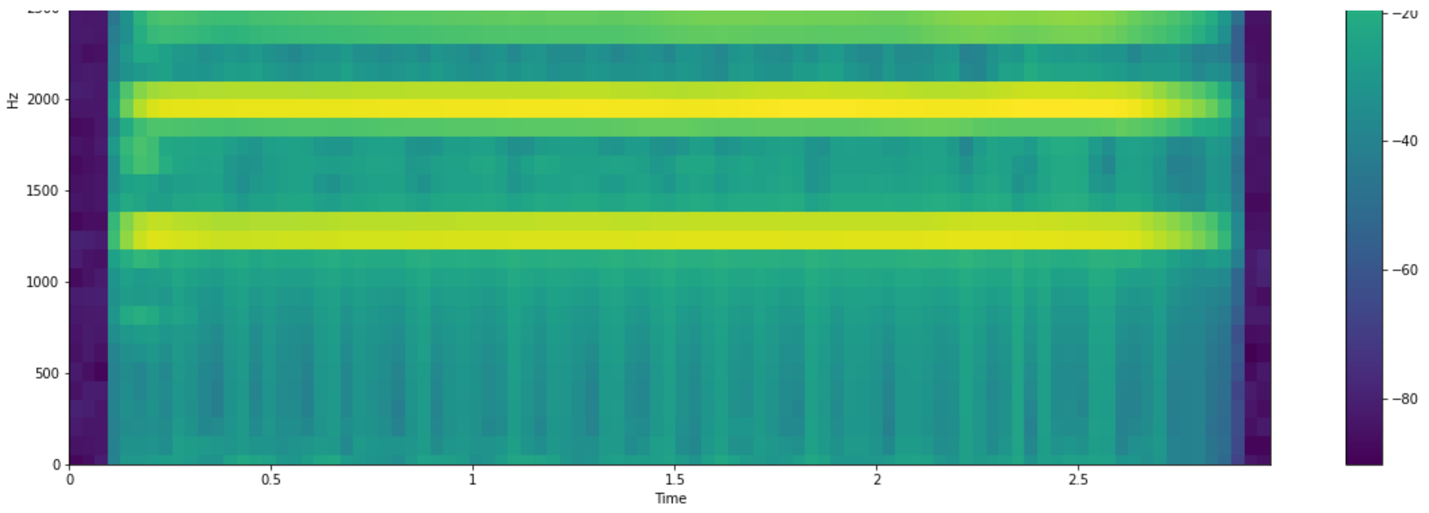
**Now perform a similar analysis with a music signal ("oboe_c6.wav").**

In [ ]:

```
# Imports
x, sr = librosa.load('/content/gdrive/My Drive/Lab 1 files/oboe_c6.wav', sr=None)
#ipd.Audio(x, rate= sr)
print('Sample Rate: ', sr)

# STFT
X = librosa.stft(x, n_fft=1024, hop_length= 256, win_length= 1024, window= 'hamming', ce
nter='False')
Xpow = np.abs(X)**2

# Let's compute the mel spectrogram (in librosa)
nfilt = 40
fmin = 1
fmax = sr/2
fbank2 = Mel_filters(nfilt, fmin, fmax, 1024, sr)

X_mel_lib = librosa.feature.melspectrogram(sr=sr, S=Xpow, n_fft = 1024, hop_length= 256,
n_mels = nfilt, fmin = fmin, fmax= fmax, htk= True)

# Plot
plt.figure(figsize=(20,10));
librosa.display.specshow(10*np.log10(X_mel_lib), sr=sample_rate, hop_length=hop_size, x_
axis = 'time', y_axis = 'linear', cmap = 'viridis' );
plt.colorbar();
```

```
Sample Rate:  22050
```



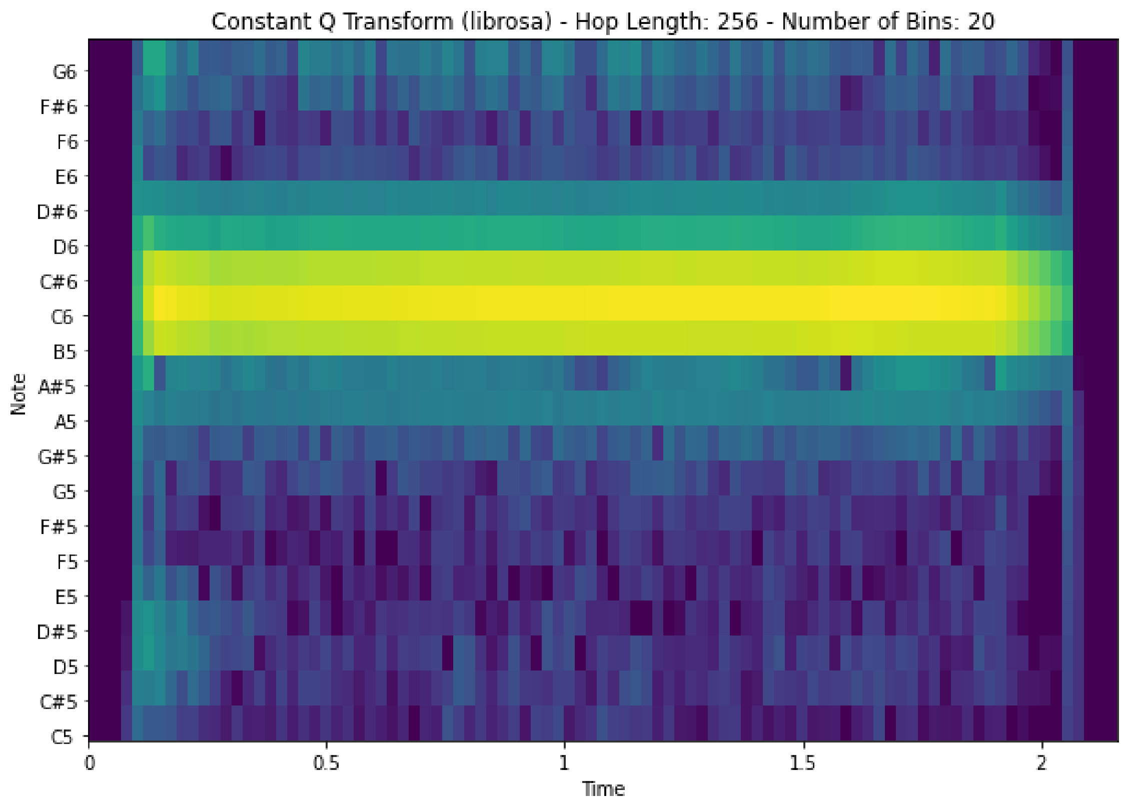


## Exercise 7: Constant-Q Transform

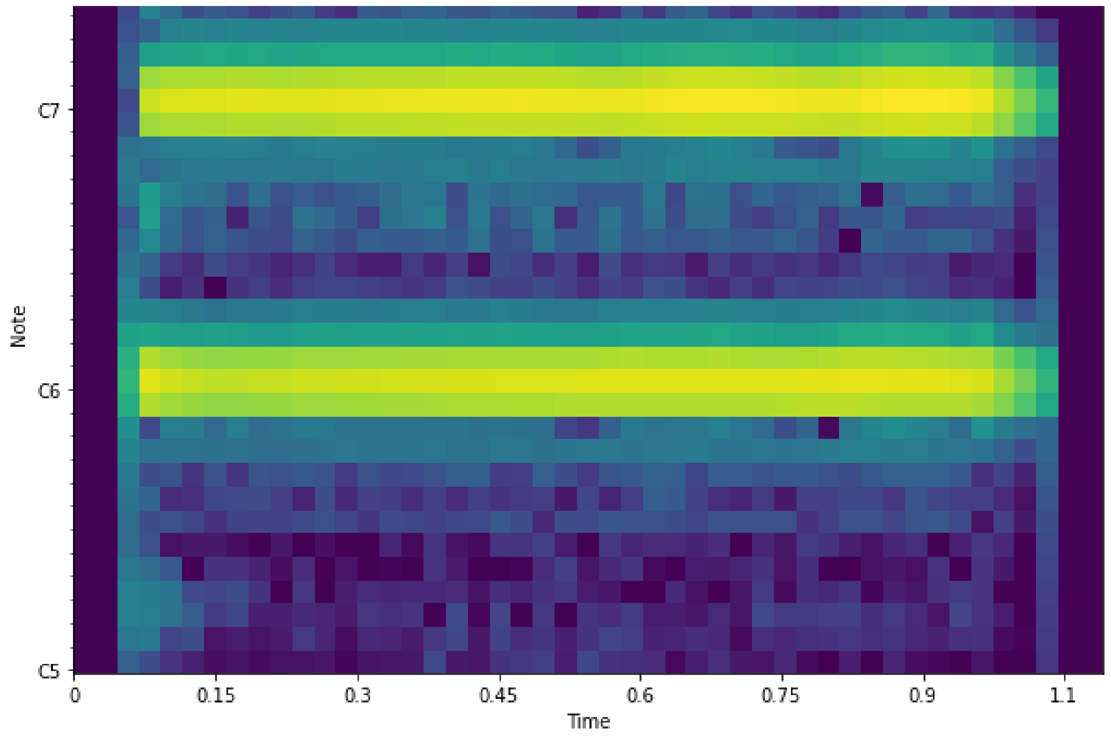
**Compute the constant Q transform using librosa. Experiment with different values of the parameters and discuss the results.**

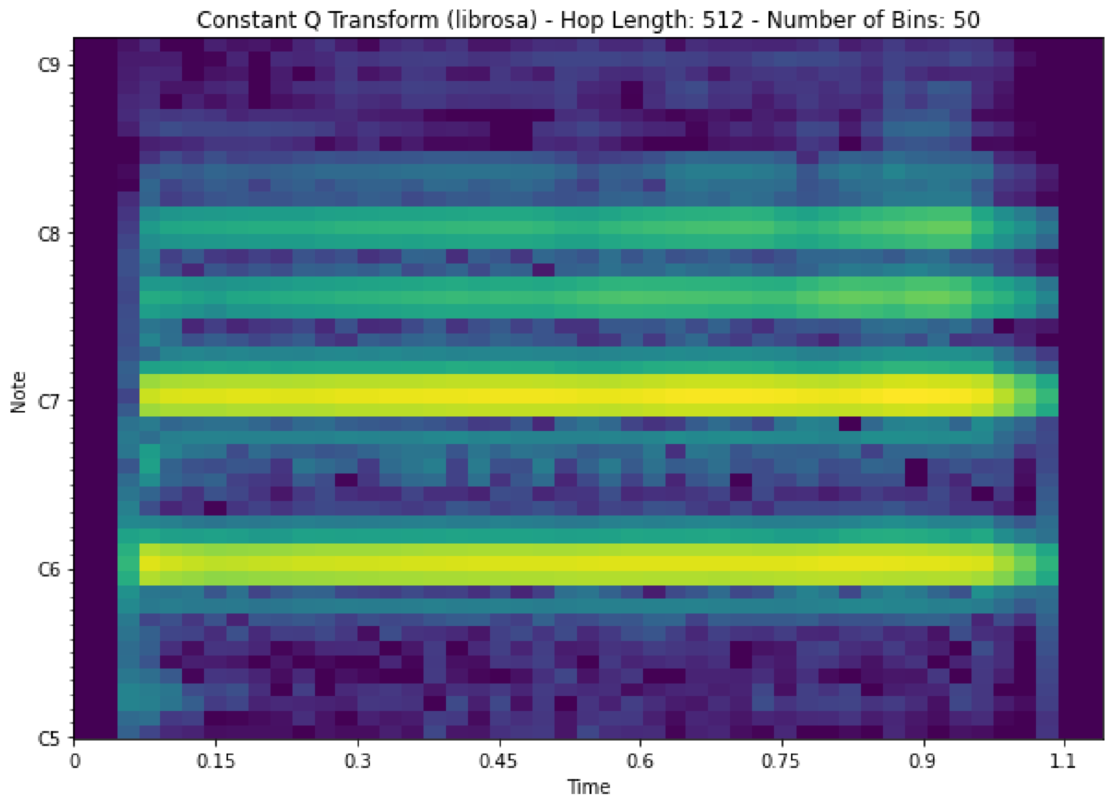
In [ ]:

```
fmin = librosa.midi_to_hz(72)
hop_length_array = [256, 512, 512]
n_bins_array = [20, 30, 50]

for i in range(0,3):
  hop_length = hop_length_array[i]
  n_bins = n_bins_array[i]

  C = librosa.cqt(x, sr=sr, fmin=fmin, n_bins = n_bins, hop_length=hop_length)
  logC = librosa.amplitude_to_db(np.abs(C))
  plt.figure(figsize=(10,7))
  librosa.display.specshow(logC, sr = sr, x_axis='time', y_axis='cqt_note', fmin = fmin
, cmap='viridis' );
  plt.title('Constant Q Transform (librosa) - Hop Length: {} - Number of Bins: {}'.forma
t(hop_length, n_bins));
```



Constant Q Transform (librosa) - Hop Length: 512 - Number of Bins: 30





**Changing the hopsize changes the temporal resolution. A bigger hop size corresponds to a lower temporal resolution.**

**The parameter nbins corresponds to the number of frequency bins, starting at fmin (according to [librosa documentation](#)).**

**Increasing the number of bins will result in an higher number of shown notes. In other words an higher number of bins allow us to see more high frequency harmonics.**

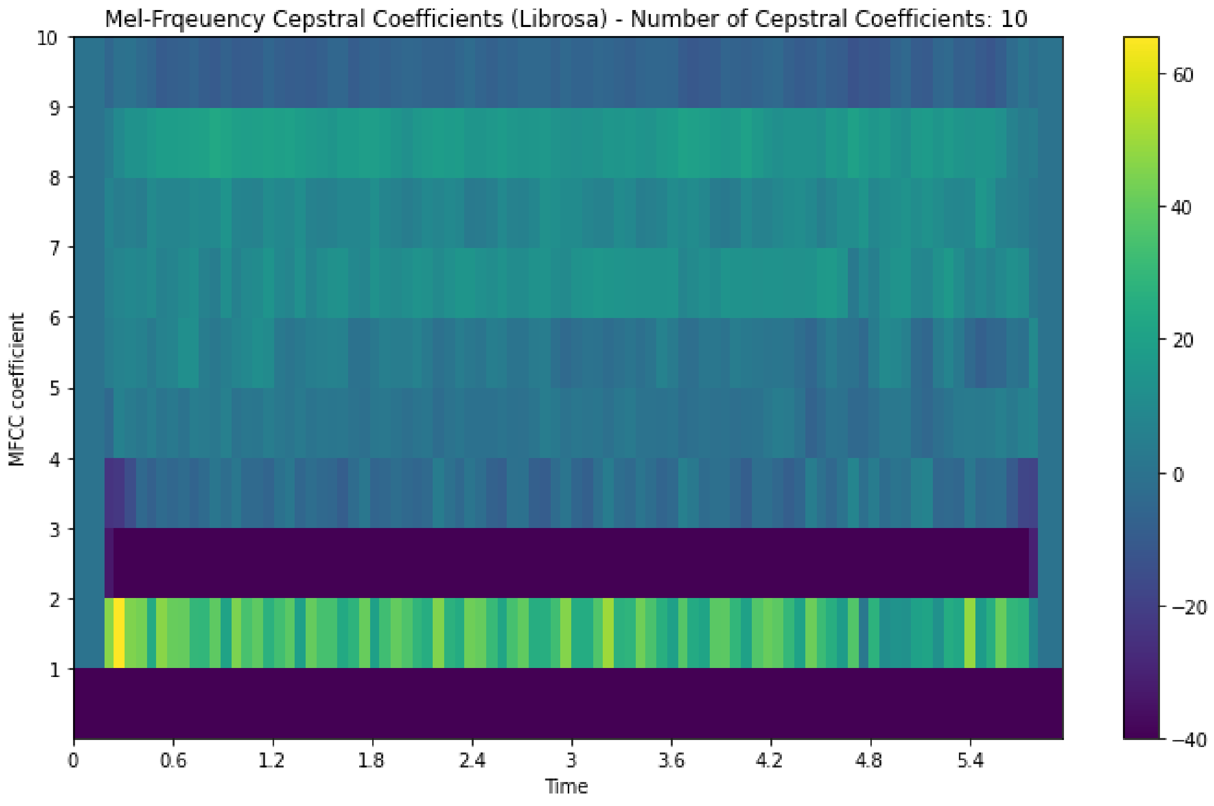
## Exercise 8: MFCCs

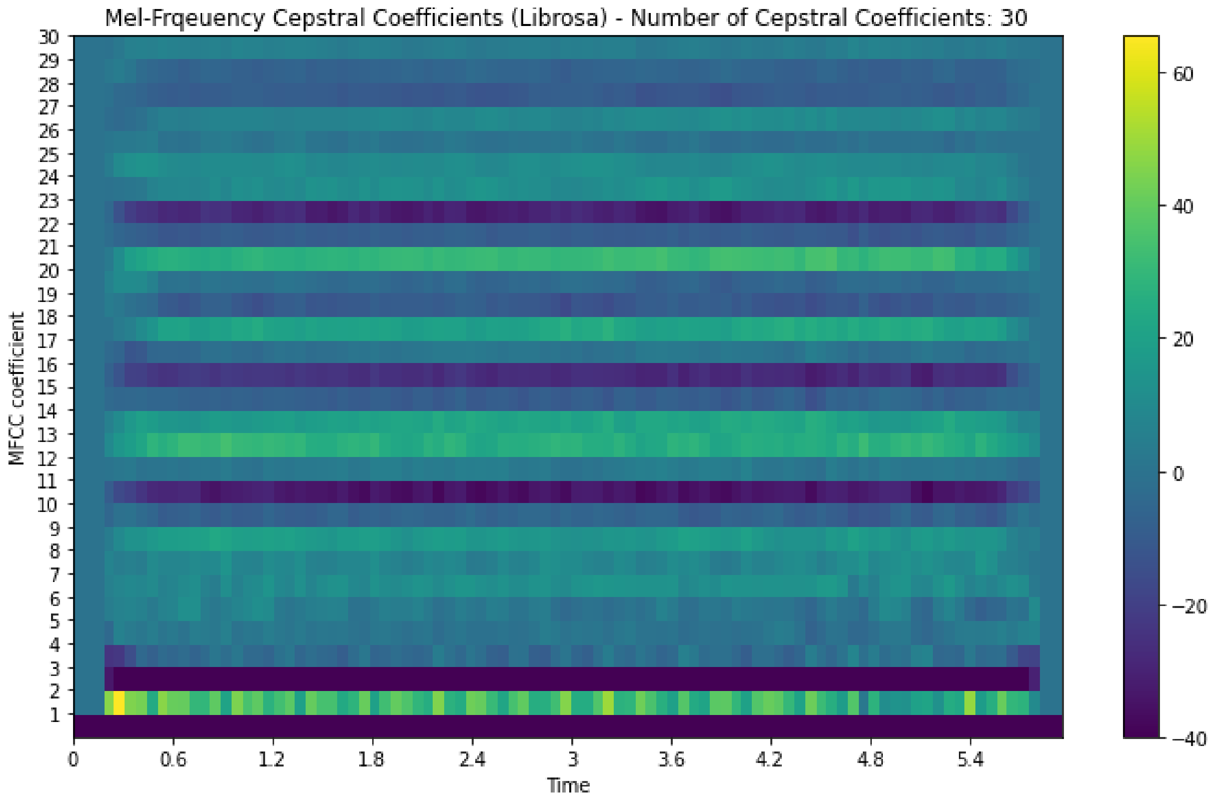
**Compute MFCCs using librosa. Experiment with different values of the parameters and discuss the results.**

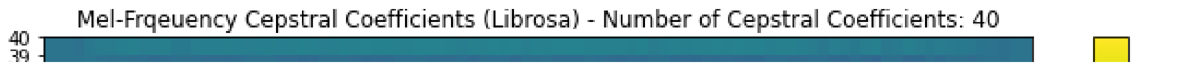
In [ ]:

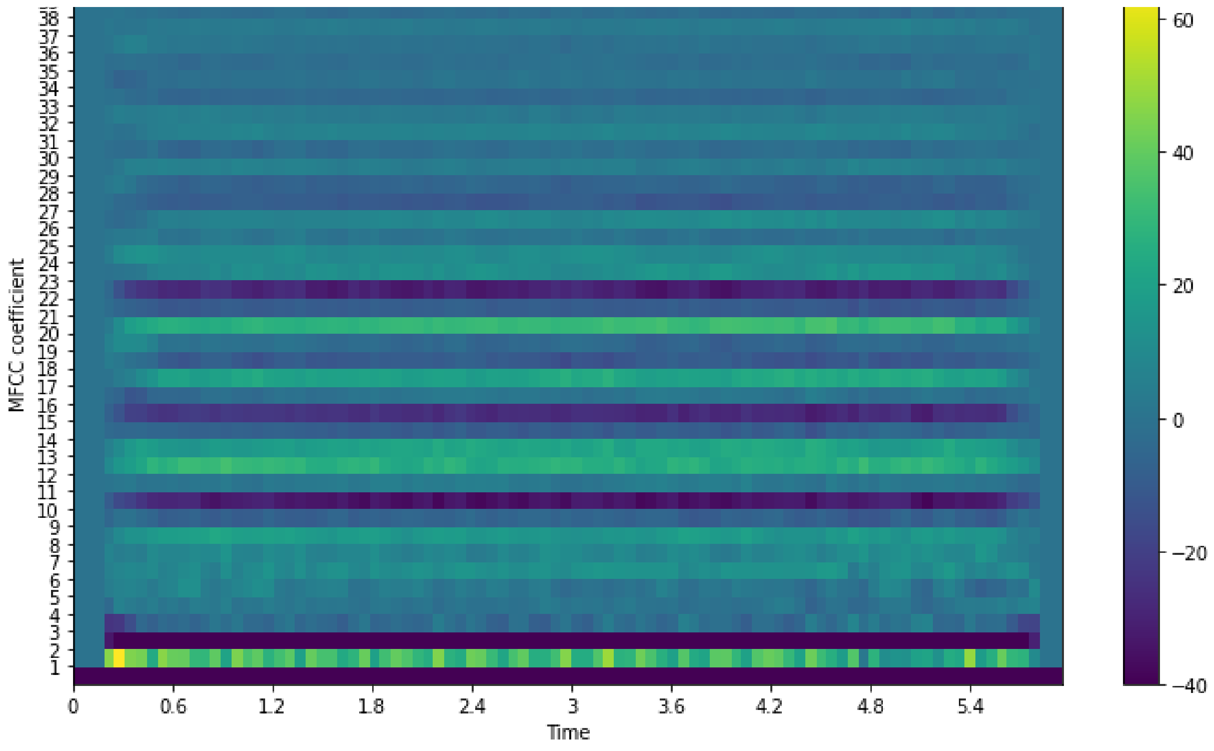
```
num_ceps_array = [10, 30, 40]
```

```
for elm in num_ceps_array:
  num_ceps = elm;
  X_mfcc = librosa.feature.mfcc(S = librosa.power_to_db(X_mel_lib), sr = sample_rate, n_
mfcc= num_ceps)

  plt.figure(figsize=(12, 7))
  librosa.display.specshow(X_mfcc, sr = sample_rate, hop_length = hop_length, x_axis='ti
me', cmap='viridis');
  plt.ylabel('MFCC coefficient')
  plt.yticks(np.arange(1,num_ceps+1))
  plt.clim(-40, None);
  plt.colorbar();
  plt.title('Mel-Frqeuency Cepstral Coefficients (Librosa) - Number of Cepstral Coefficie
nts: {}'.format(elm));
```









**Increasing the Number of cepstral coefficients increases the number of MFCCs to return.**

**Increasing the Number of cepstral coefficients does not increase the frequency resolution.**

**Changing the hopsize changes the temporal resolution. A bigger hop size corresponds to a lower temporal resolution.**