

Homework 2: Gábor transforms

DUE: Friday, Feb. 7, 2020

Part I

In this homework, you will analyze a portion of Handel's Messiah with time-frequency analysis. To get started, you can use the following commands (note that Handel is so highly regarded, that MATLAB has a portion of his music already in MATLAB! I'm sure they'll add some Beyonce and T-Swift at some point, but you'll have to suffer Handel for now):

```
load handel
v = y'/2;
plot((1:length(v))/Fs,v);
xlabel('Time [sec]');
ylabel('Amplitude');
title('Signal of Interest, v(n)');
```

This code gives you plots the portion of music you will analyze. To play this back in MATLAB, you can use the following commands:

```
p8 = audioplayer(v,Fs);
playblocking(p8);
```

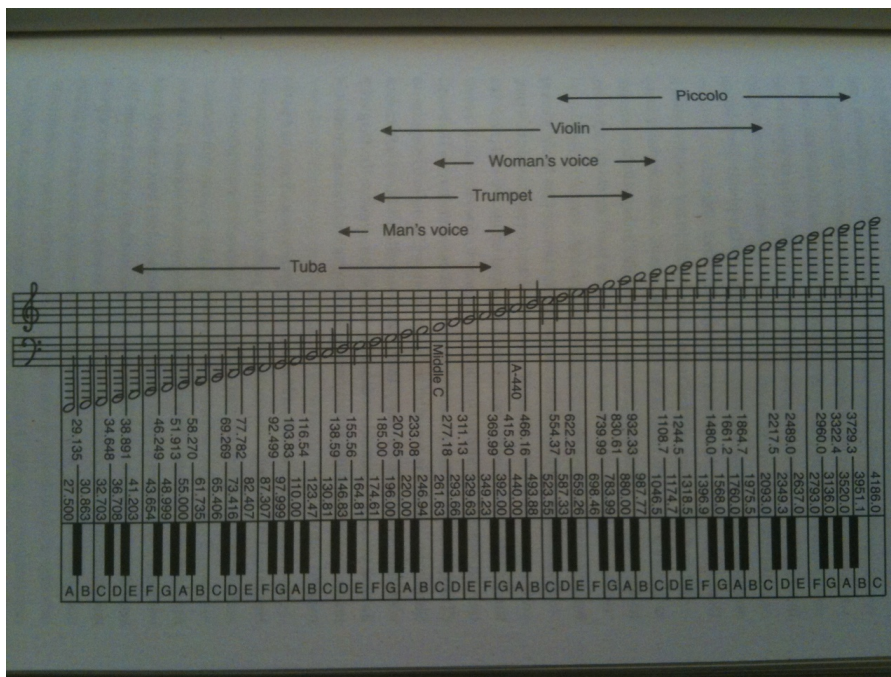
This homework is rather open ended in the sense that I want you to explore the time-frequency signature of this 9 second piece of classic work. Things you should think about doing:

1. Through use of the Gábor filtering we used in class, produce spectrograms of the piece of work.
2. Explore the **window width** of the Gábor transform and how it effects the spectrogram.
3. Explore the spectrogram and the idea of **oversampling** (i.e. using very small translations of the Gábor window) versus potential **undersampling** (i.e. using very course/large translations of the Gábor window).
4. Use **different Gábor windows**. Perhaps you can start with the Gaussian window, and look to see how the results are effected with the Mexican hat wavelet and a step-function (Shannon) window.

Don't be cheap on time here, i.e. don't be lame. This is an opportunity for you to have a creative and open ended experience with MATLAB and data analysis. Please do a nice job writing things up and labeling your resulting plots. I believe this homework can be a really engaging and educational experience if you devote some time to it.

Part II

From the website, download the two files **music1.wav** and **music2wav**. These files play the song *Mary had a little lamb* on both the recorder and piano. These are **.wav** files I generated using my iPhone. To important and convert them, use the following commands for both pieces (NOTE: basically both pieces are converted to a vector representing the music, thus you can easily edit the music by modifying the vector).



```
tr_piano=16; % record time in seconds
y=wavread('music1'); Fs=length(y)/tr_piano;
plot((1:length(y))/Fs,y);
xlabel('Time [sec]'); ylabel('Amplitude');
title('Mary had a little lamb (piano)'); drawnow
p8 = audioplayer(y,Fs); playblocking(p8);
```

```
figure(2)
tr_rec=14; % record time in seconds
y=wavread('music2'); Fs=length(y)/tr_rec;
plot((1:length(y))/Fs,y);
xlabel('Time [sec]'); ylabel('Amplitude');
title('Mary had a little lamb (recorder)');
p8 = audioplayer(y,Fs); playblocking(p8);
```

1. Through use of the Gábor filtering we used in class, reproduce the music score for this simple piece. See Fig. 1 which has the music scale in Hertz. (note: to get a good clean score, you may want to filter out overtones... see below).
2. What is the difference between a recorder and piano? Can you see the difference in the time-frequency analysis? Note that many people talk about the difference of instruments being related to the *timbre* of an instrument. The timbre is related to the overtones generated by the instrument for a center frequency. Thus if you are playing a note at frequency ω_0 , an instrument will generate overtones at $2\omega_0, 3\omega_0, \dots$ and so forth.