Middle East Technical University
Electrical and Electronics Engineering
Fall 2014

EE 430 Project

Preliminary Work: Spectrogram Viewer

As a first step of the project, you are going write a MATLAB code to display the spectrogram of a discrete-time signal. A spectrogram is a color or grey scale plot of the magnitude of the short-time Fourier transform (STFT) on the time-frequency plane. Somewhat informally, let $\omega[n]$ (window function, tapering function) be a discrete-time sequence having a finite duration centered around $n = n_0$. For example, if all samples of $\omega[n]$ are the same then it is called a *rectangular* window[1]. Let $x[n]$ be the signal you wish to analyze. By multiplying $\omega[n]$ and $x[n]$ you extract a portion of $x[n]$ around n_0. By doing so, one focuses on the spectral properties of $x[n]$ over a particular time interval. Then, compute the magnitude of the discrete Fourier transform[2] (DFT) of the product. This is considered as a spectral description at time n_0. The magnitude values of this DFT can be coded in color or grey scale and plotted on the time-frequency plane (take horizontal axis as time and vertical axis as frequency). Proceeding with a sequence of n_0 values, a spectrogram can be plotted. The window length and the amount of shift between consecutive windows are the fundamental parameters. The shift of the window should <u>at most</u> be equal to its length. The type of window function is also important. It may have a significant role in some tasks.

Note that MATLAB has its own "spectrogram" command. You are going to write your own. You may use all other MATLAB commands. You can compare your results to those of MATLAB and other tools you can find from the web.

A signal and its spectrogram are shown in Figure 1 and Figure 2, respectively. Note that the horizontal axes (time) in both figures have the same scale.

[1] There are other well-known window functions like Hamming, Hanning, Tukey, Cosine, Triangular, Gaussian, Blackman, Kaiser…

[2] DFT values are the samples of DTFT.

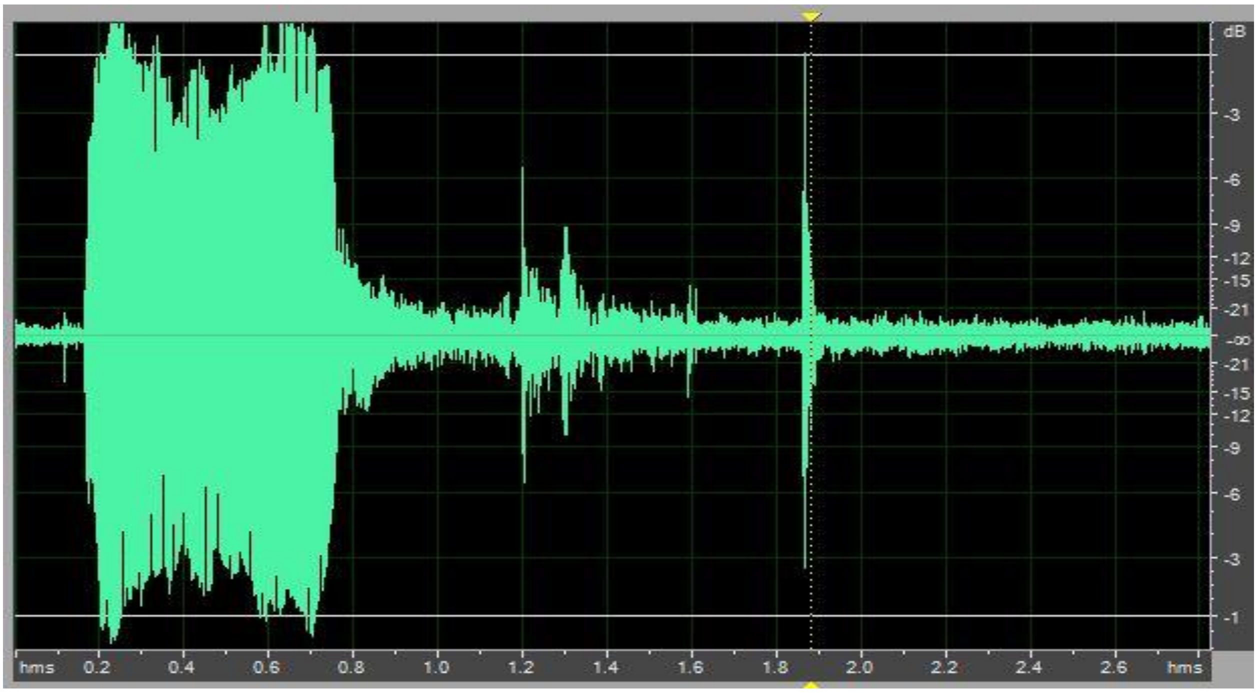


Figure 1: A signal to be analyzed.

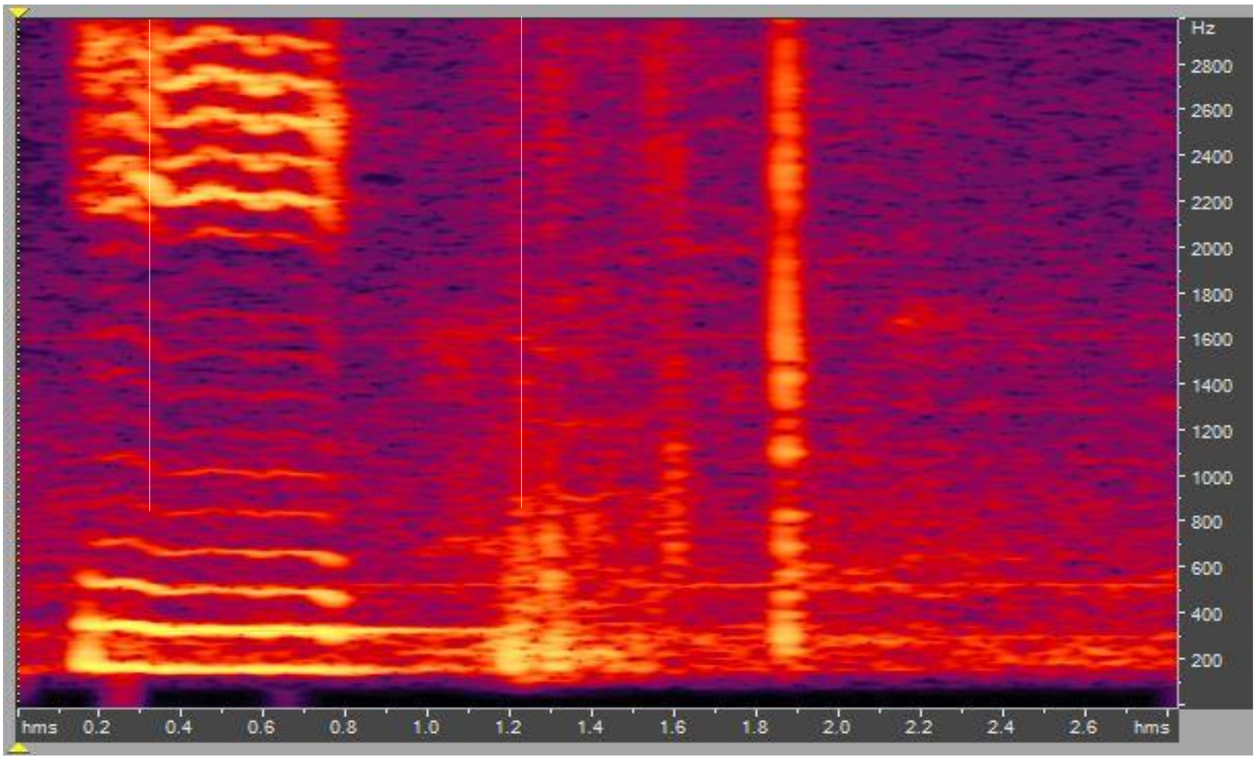


Figure 2: Spectrogram of the signal in Figure 1.

Figures 3 and 4 display the enlarged views of the signal around the time instants indicated by the white lines in Figure 2 on the left and right, respectively.

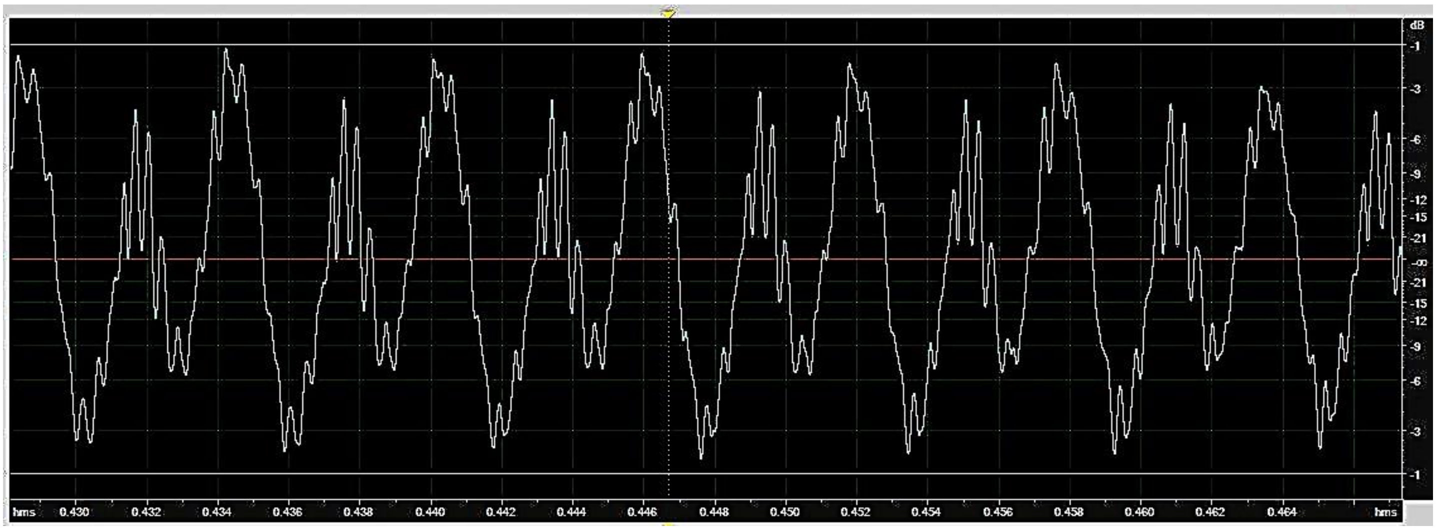


Figure 3: The signal details around the time instant of the white line on the left in Figure 2.

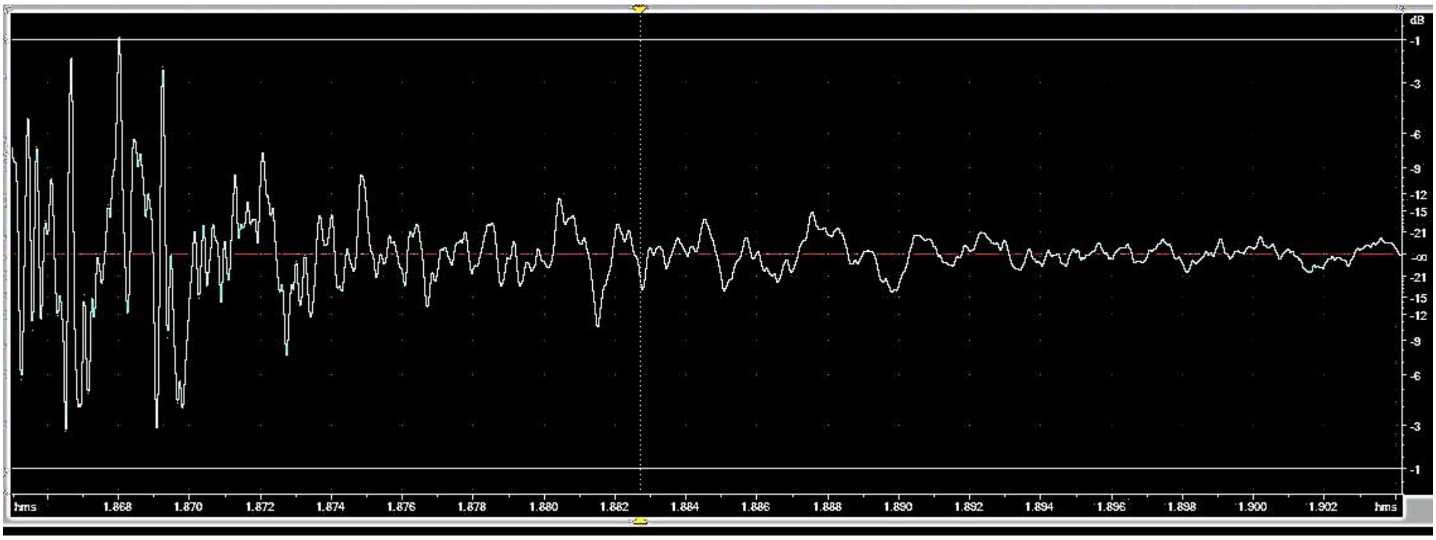


Figure 4: The signal details around the time instant of the white line on the right in Figure 2.

The magnitudes of the STFTs computed at the time instants indicated by the white lines in Figure 2 are shown in Figures 5 and 6.

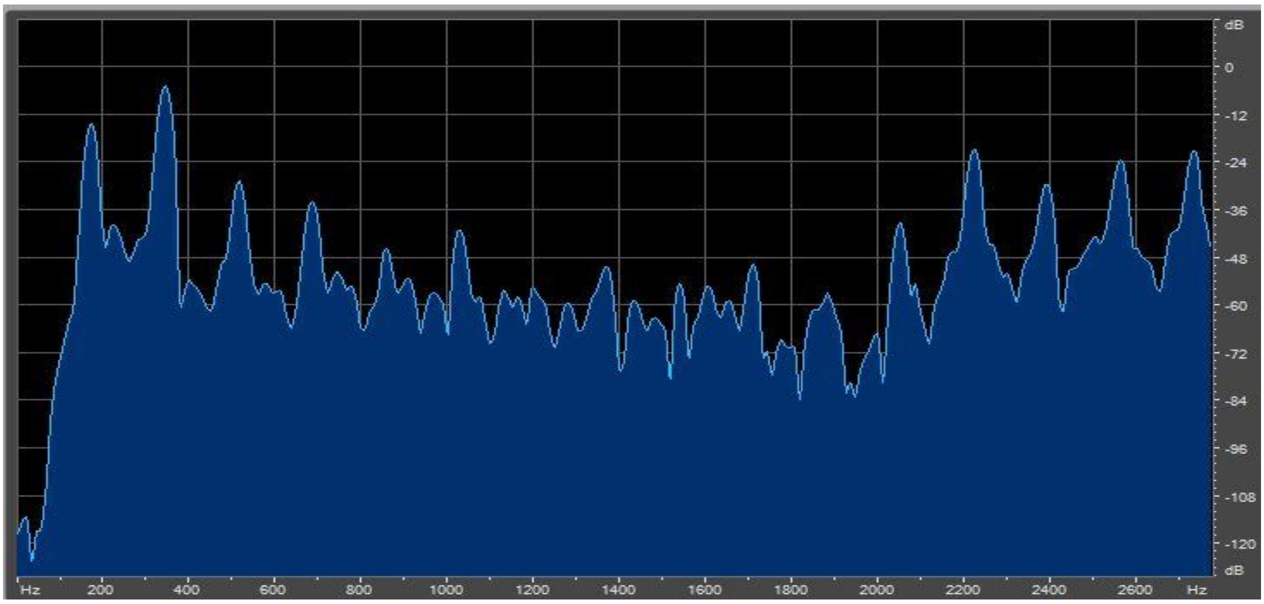


Figure 5: Magnitude of the STFT computed at the time instant indicated by the white line on the left in Figure 2.

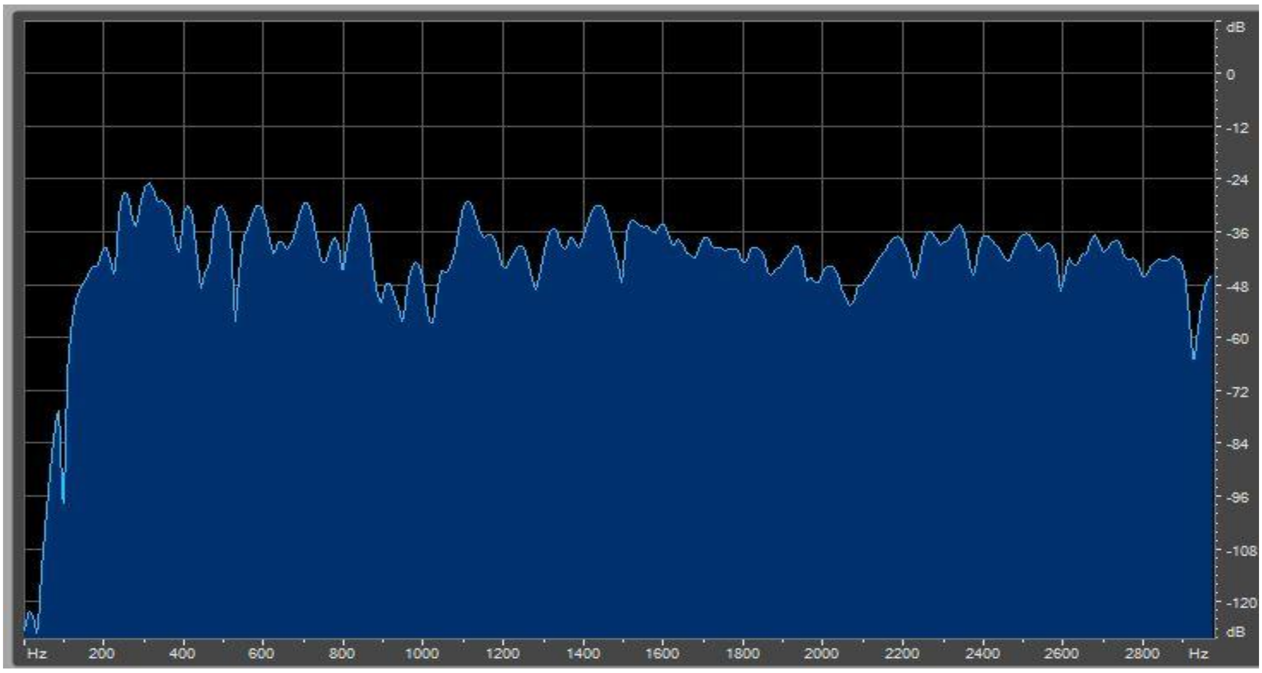


Figure 6: Magnitude of the STFT computed at the time instant indicated by the white line on the right in Figure 2.

Things to do:

1) In your spectrogram code, window length, window type must be variable. For window type, use the alternatives (Hamming, Hanning, Tukey, Cosine, Triangular, Gaussian, Blackman, Kaiser…) provided by MATLAB.

2) Try different window lengths and state the differences of the spectrograms obtained by using different window lengths.

3) Try different amounts of window overlap and state the differences of the spectrograms obtained by using different amounts of window overlap.

4) Prepare a <u>short</u>, <u>descriptive</u>, <u>clearly written (in language and format)</u> report.

<u>Deadline for the submission and demonstration of the spectrogram code</u>: December 05, 2014.

Project Topics

There are three topics. The first one, “frequency estimation”, is mandatory for all, then choose also one of the other two projects: “musical instrument tuner” or “estimation of liquid level in a bottle by blowing at the bottle mouth”.

<u>Deadline for the submission and demonstration of the project work</u>: Jan. 12th and 16th , 2014.

1) Frequency Estimation.

Estimation of the frequency of a digitized sinusoidal signal in noise. Estimation error will be measured in percent of the true frequency. Variables/parameters which have to be taken into account in assessing the accuracy of estimation:

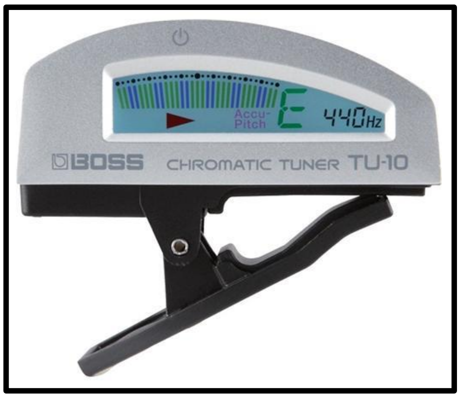
1) Frequency of sinusoid, f_0, in Hz.
2) Length of observation interval, K_O, in number of periods of the sinusoidal signal.
3) Sampling rate, f_s, and its value, K_S relative to the frequency of sinusoidal signal
4) SNR (signal-to-noise ratio): $\frac{\sigma_S^2}{\sigma_N^2}$. σ_S^2 is the power of the sinusoidal signal, σ_N^2 is the random noise power. Noise is white, zero-mean and Gaussian.
5) Number of bits, B, used to represent signal samples.

Things to do:
After some research, theoretical study and self-experimentation, you are going develop your method. Then prepare a <u>short</u>, <u>descriptive</u>, <u>clearly written (in language and format)</u> report where you describe your method and provide experimental results to reveal the performance/accuracy of your method according to the variables described above.

It is totally your own responsibility to find a measure to expose the accuracy of your method. Be clear and specific in the contents and the format of the tables, figures, plots, etc., and the numerical variables involved.

2) Musical Instrument Tuner (aid).

Aids for musical instrument tuning (“tuner” for short) are commercially available, small, handheld devices. Some samples are shown below.



When tuning a musical instrument, tuner is placed close to the instrument and a sound is produced by the instrument, for example, by pulling a guitar string or by hitting a piano key. The tuner picks the sound of the instrument via its microphone (tuner may also be placed on the body of the instrument). By analyzing the instrument's sound, the tuner finds the musical note that is closest to the produced sound and it visually indicates the deviation of the produced sound from the closest matching note. Then, the musician adjusts (tenses or relaxes) the tension of the string to observe a matching indication on the screen of the tuner.

The fundamental frequencies of musical notes are given in Table 1. Any octave range in this table can be generated by using a scale factor of $2^{\frac{k}{12}}$, $k = 1,2,\ldots 12$ (the value 12 is due to the 12-tone chromatic scale). An example is given in Table 2 for the range 440-880 Hz.

Table 2			
k	$2^{\frac{k}{12}}$	$440\times 2^{\frac{k}{12}}$	
0	1	$440\times 2^{\frac{0}{12}} =$	440
1	1,06	$440\times 2^{\frac{1}{12}} =$	466,16
2	1,12	$440\times 2^{\frac{2}{12}} =$	493,88
3	1,19	$440\times 2^{\frac{3}{12}} =$	523,25
4	1,26	$440\times 2^{\frac{4}{12}} =$	554,37
5	1,33	$440\times 2^{\frac{5}{12}} =$	587,33
6	1,41	$440\times 2^{\frac{6}{12}} =$	622,25
7	1,50	$440\times 2^{\frac{7}{12}} =$	659,26
8	1,59	$440\times 2^{\frac{8}{12}} =$	698,46
9	1,68	$440\times 2^{\frac{9}{12}} =$	739,99
10	1,78	$440\times 2^{\frac{10}{12}} =$	783,99
11	1,89	$440\times 2^{\frac{11}{12}} =$	830,61
12	2,00	$440\times 2^{\frac{12}{12}} =$	880,00

Table 1: Fundamental frequencies for equal-tempered scale
This is table created using A_4 = 440 Hz
Speed of sound = 345 m/s = 1130 ft/s = 770 miles/hr
"Middle C" is C_4

Note	Frequency (Hz)	Wavelength (cm)		Note	Frequency (Hz)	Wavelength (cm)
C0	16,35	2100		D4	293,66	117
C#0/Db0	17,32	1990		D#4/Eb4	311,13	111
D0	18,35	1870		E4	329,63	105
D#0/Eb0	19,45	1770		F4	349,23	98,8
E0	20,6	1670		F#4/Gb4	369,99	93,2
F0	21,83	1580		G4	392	88
F#0/Gb0	23,12	1490		G#4/Ab4	415,3	83,1
G0	24,5	1400		A4	440	78,4
G#0/Ab0	25,96	1320		A#4/Bb4	466,16	74
A0	27,5	1250		B4	493,88	69,9
A#0/Bb0	29,14	1180		C5	523,25	65,9
B0	30,87	1110		C#5/Db5	554,37	62,2
C1	32,7	1050		D5	587,33	58,7
C#1/Db1	34,65	996		D#5/Eb5	622,25	55,4
D1	36,71	940		E5	659,26	52,3
D#1/Eb1	38,89	887		F5	698,46	49,4
E1	41,2	837		F#5/Gb5	739,99	46,6
F1	43,65	790		G5	783,99	44
F#1/Gb1	46,25	746		G#5/Ab5	830,61	41,5
G1	49	704		A5	880	39,2
G#1/Ab1	51,91	665		A#5/Bb5	932,33	37
A1	55	627		B5	987,77	34,9
A#1/Bb1	58,27	592		C6	1046,5	33
B1	61,74	559		C#6/Db6	1108,73	31,1
C2	65,41	527		D6	1174,66	29,4
C#2/Db2	69,3	498		D#6/Eb6	1244,51	27,7
D2	73,42	470		E6	1318,51	26,2
D#2/Eb2	77,78	444		F6	1396,91	24,7
E2	82,41	419		F#6/Gb6	1479,98	23,3
F2	87,31	395		G6	1567,98	22
F#2/Gb2	92,5	373		G#6/Ab6	1661,22	20,8
G2	98	352		A6	1760	19,6
G#2/Ab2	103,83	332		A#6/Bb6	1864,66	18,5
A2	110	314		B6	1975,53	17,5
A#2/Bb2	116,54	296		C7	2093	16,5
B2	123,47	279		C#7/Db7	2217,46	15,6
C3	130,81	264		D7	2349,32	14,7
C#3/Db3	138,59	249		D#7/Eb7	2489,02	13,9
D3	146,83	235		E7	2637,02	13,1
D#3/Eb3	155,56	222		F7	2793,83	12,3
E3	164,81	209		F#7/Gb7	2959,96	11,7
F3	174,61	198		G7	3135,96	11
F#3/Gb3	185	186		G#7/Ab7	3322,44	10,4
G3	196	176		A7	3520	9,8
G#3/Ab3	207,65	166		A#7/Bb7	3729,31	9,3
A3	220	157		B7	3951,07	8,7
A#3/Bb3	233,08	148		C8	4186,01	8,2
B3	246,94	140		C#8/Db8	4434,92	7,8
C4	261,63	132		D8	4698,64	7,3
C#4/Db4	277,18	124		D#8/Eb8	4978,03	6,9

The sounds produced by musical instruments contain harmonic components. Conceptually, each note is associated with sinusoidal components at the fundamental frequency and its multiples (like in Fourier series representation). Aperiodic/random signal components also exist in the composition. Even if these aperiodic/random components did not exist, the sound would still be far from being periodic due to the amplitude variation of the "sinusoidal" components. As you know, once you pull the string, the sound intensity decays quickly. These are some of the facts that may cause difficulties in frequency analysis.

Basically, the task is the determination (estimation) of the fundamental frequency of the sound produced by the instrument, and give some indication to the user. The processing time is a concern.

Short-time Fourier transform (STFT) is a useful tool for frequency analysis. The signal to be analyzed is first "windowed" to obtain a short segment, and then its Discrete Fourier transform (DFT) is computed. The choice of the window function may be critical.

In your project studies, in addition to our textbook and reference books, you may refer to books on acoustics, physics of musical sounds and frequency estimation methods (for example, "The Physics of Musical Instruments", N. H. Fletcher, Thomas D. Rossing, Springer Verlag, "Introduction to Spectral Analysis", P. Stoica, R. Moses, Prentice Hall).

During your demonstration, the performance of the system you designed and implemented will be assessed by comparing your system's response and that of a professional tuner device, in tuning a guitar's strings. A user friendly interface will be very useful to use your software. You can examine those of mobile applications like "PitchLab Guitar Tuner" and "Tuner – gStrings". There will be guitars that you can use in the lab.

Things to do:

1) We have two guitars. You may use for your work.
2) Think about your setup and prepare it.
3) Work it out. Design your method. Try to make something a bit more refined than just peak picking in the frequency domain. Consider the nature, characteristics of the sounds produced by the instrument. Resilience to sounds in the environment would be an extra feature.
4) Measure and present the accuracy of your method in your report. The performance of your method has to be measured statistically. Decide on the number of trials you need to state the accuracy according to some numerical precision.
5) A simple but functional user interface.
6) Prepare a <u>short</u>, <u>descriptive</u>, <u>clearly written (in language and format)</u> report.

3) Estimation of liquid level in a bottle by blowing at the bottle mouth.

A bottle can be considered as a tube with nonuniform cross section and closed at one end. A uniform tube has an acoustic resonance frequency inversely proportional to the length of the tube; harmonics also exist. Geometrical nonuniformity of a bottle yields multiple resonance frequencies. Depending on its specific geometry, each bottle has its own characteristics. The interaction of the acoustic waves and the material the bottle is made of yields energy loses. Therefore, the material also contributes to the formation of the resonance characteristics.

When a bottle is blown at its mouth, the nature of the sound heard is a result of its resonance characteristics. Loosely speaking, the envelope of the magnitude spectrum of the sound displays peaks around the resonance frequencies. This is what “resonance characteristics” roughly mean to us. The peakynesses, separations of the peaks change according to the physical dimensions and geometry of the cavity in the bottle.

It would be appropriate to work with one or just a few specific bottles throughout your study. To choose the bottle(s) you will use in the demonstration, you can do preliminary experiments with a variety of bottles, and observe and compare their spectral characteristics, and select one/those which you conclude to better suit to your job.

Upon choosing bottle(s), you can form a database by recording bottle-blowing sounds at different liquid levels. It would be wise to have multiple recordings for any liquid level due to the randomness of the blowing process. You have to observe the spectral characteristics of these recordings and identify the relationship(s) between the liquid level and spectral characteristics. Then, you can decide on the spectral properties which can be used to estimate the liquid level. For example, you may observe strong tonal (sinusoidal) components (sharp peaks) or the spectrum may have a smoother envelope. In both cases the frequencies of the peaks may be related to the liquid level or you may discover other relationships. In the case of sharp peaks, determining their frequencies can be an important part of your method. If the spectrum is smoother then not only the frequencies but also the bandwidths of the peaks may be considered.

You can make a short survey on the methods of modeling spectral characteristics while you are thinking about your own approach. For example, in the context of speech studies, all-pole models, Gaussian mixture models, mel (melody) frequency cepstral coefficients (MFCC) are well known in modeling the vocal tract[3] characteristics. If you feel somewhat reluctant or find it too time-consuming to try to understand such “mysterious” things, don’t worry, you can still find your own way to make a good liquid level estimator without them. This is not a difficult problem!

Things to do:

1) Determine your bottle.
2) Think about your setup and prepare it.
3) Work it out. Design your method. Try to make something a bit more refined than just peak picking in the frequency domain. Consider the nature, characteristics of the sounds produced by the bottle. Resilience to sounds in the environment would be an extra feature.
4) Measure and present the accuracy of your method in your report. The performance of your method has to be measured statistically. Decide on the number of trials you need to state the accuracy according to some numerical precision. Percent deviation may be a performance score.
5) A simple but functional user interface.
6) Prepare a <u>short</u>, <u>descriptive</u>, <u>clearly written (in language and format)</u> report.

[3] Vocal tract is also a nonuniform-cross-section-tube type acoustic filter.