**Lab 5: DFT**

Q1

The spacing between the circles in the frequency axis of the DFT is 1 Hz.

Q2

The frequency density of the DFT is 2 because the sampling was done for .5 seconds. Therefore, the circles are spaced by 2 Hz and the 2 Hz peak appears on the second circle from the left, as opposed to the third one in the last question.

Q3

Δf=R/N=1/T

Q4

No there is not. Since the Nyquist frequency is always more then 2 Hz, the frequency that is being sampled in the combinations I’ve heard, there is no aliasing, so the sound that is being produced will always be clean and sounds the same as the other sounds of the other combinations.

Q5

From my own experience, it is definitely easier to determine the pitch of a sound if it is played longer. Listening to a tone for a longer period of time would give me more time to hone in on its frequency.

P1

R = 44,100 samples per second

N = 1024 points

1. T = R/N = 44,100/1024 = 43.066 seconds
2. fmax = R/2 = 44,100/2 = 22,050 Hz
3. ∆f = frequency range / number of points = (R/2)/(N/2) = R/N = 1024/44,100 = 0.023 Hz

P2

F(max) = 2000 Hz

∆f = 1 Hz frequency resolution

1. R = 2\*fmax = 2\*2000 = 4000 samples per second
2. N = R/∆f = 4000/1 = 4000 samples
3. T = R/N = 4000/4000 = 1 second

P3

120 beats per minute

1/16th note: 4 notes per beat

A2 fundamental frequency = 110 Hz

1. ∆f = 1/T = 1/((120\*4)/60) = 1/8 Hz
   1. f2 = next semitone = f0(21/12) = 110(21/12) = 116.54
   2. ∆f = 1200\*log2(f2/f1) = 1200\*log2(116.54/110) = 100 cents
2. After analyzing the harmonics of the note to decide if it is in tune or not, the frequency resolution would be smaller than when studying the fundamental.

Q6

Yes. Using the DFT on a waveform to reconstruct it implies that the waveform will be the sum of the components computed by the DFT independently.

Q7

If f1 = 2.5, R = 8, and T = 2, then N = 16. There are 9 dots across the frequency domain of the graph, where the 2.5 Hz peak is on the 5th point from the left. If R is held constant when T = 4, this changes N = 32. This will then double the frequency density, bringing the intervals down from .5 to .25, causing the reading for 2.5Hz to be on 11th point from the left on the frequency domain.

Q8

The low sampling rate in the original file causes heavy distortion in the wave form. The second played back sound uses DFT, which allows the computer to understand the frequency consistency inside of the file, not just it’s amplitude values across time. This allows the computer to accurate reconstruct the file at the specified frequency with a higher resolution with the same amount of information, as opposed to reconstructing the file with the original samples. The sound then is much cleaner then the original sample.

Q9

No, the DFT will not visually represent the phase change in the Frequency Domain window. The DFT uses sine and cosine consistency calculations in order to form the information given in the frequency domain window. Those calculations may vary depending on the phase of a wave, but the conclusions draw on the frequency consistency of a file will be the same.

Q10

Aliasing in a signal works in an oscillating fashion. Frequencies that are recorded above the Nyquist frequency play back sounds that either equal to or lower then the Nyquist frequency. For every Hertz above the Nyquist frequency, the played back sound is that many Hertz lower then the Nyquist frequency. If a frequency is more then double the Nyquist frequency, then for every Hertz it is over double the Nyquist frequency, that frequency is played back as how every many Hertz it is over double the Nyquist frequency. This relationship repeats itself as the recorded frequencies progress over multiples of the Nyquist frequency.

Q11

The phenomenon that occurs within the compound tones is a dissonance between two aliases. Both frequencies are aliased to a frequency below the Nyquist frequency and are then played against each other. This changes the relationship between both frequencies and would cause even heavier alterations to the signals.