ECOMMS Lab 1: Waveform Synthesis and Spectral Analysis

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***Abstract*—This laboratory project contains four parts. Where arbitrary waveforms with specified Signal-to-Noise Ratio (SNR) are produced, along with graphical analysis of the wave form's frequency spectrum.**

I. Introduction and Background

Waveform synthesis surrounds us everywhere. It can penetrate through walls, and allows us to communicate ideas around the world. In other words, radio signals and others such as cellphone signals (3G, 4G, LTE, and Bluetooth) are transmitted for short or long range applications on a daily basis. Therefore, this experiment allows us to have a deeper understanding of these signals and provide an insight as to how these wave forms can communicate and send data.

II. Results

A. Part 1: Digital synthesis of arbitrary wave forms with specified SNR

This part of the experiment, the task was to listen to and analyze different tones/wave forms by varying the SNR, frequency, and amplitude of each signal below

$$s = 0.5 * sin(2 * pi * 466.16 * t)$$

$$s = A_c * [1 + cos(2 * pi * f_m * t)]. * (cos(2 * pi * f_c * t))$$

$$s = A_c * cos(2 * pi * f_c * t) + b * sin(2 * pi * f_m * t)$$

B. Part 2: Comparison between CFT and DFT (FFT)

This section explores the Continuous Fourier Transform and Discrete Fourier Transform of a rectangular pulse signal. A figure of a time domain signal was given as shown in Figure 1. A reconstructed signal is shown in Figure 4 using rectangular pulse function.

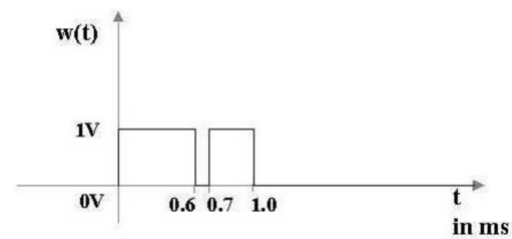

Fig. 1. Time-domain Signal

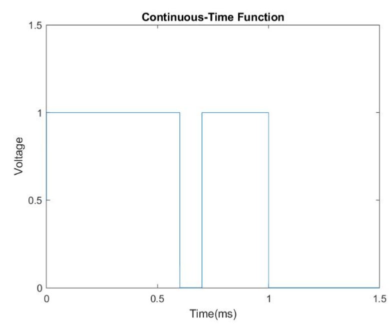

Fig. 2. Time-domain Signal in Matlab

The CFT was calculated analytically by finding the Fourier waveform of the rectangular pulse. This was done by converting the rectangular pulse into it's Fourier waveform. The property of rectangular pulse waveform was used as shown in equation 1.

$$\Pi(\frac{t}{T}) = T|Sa(\pi * f * T)| \tag{1}$$

The rectangular pulse time waveform is described as:

$$w(t) = \pi(\frac{t}{0.06}) + \pi(\frac{t-0.7}{0.3}) \tag{2}$$

From this, each term's Fourier waveform was found. Which is shown below as:

$$W(f) = (0.6sinc(0.6f) + 0.03sinc(0.3f))e^{-j1.4\pi f} \tag{3}$$

The Continuous Fourier Transform of this Fourier Waveform is shown below.

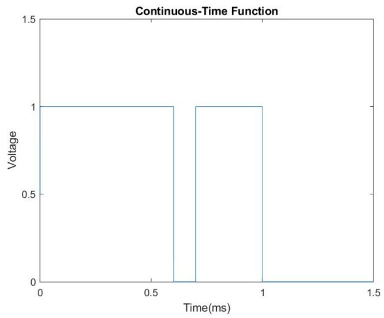

Fig. 3. Time-domain Signal in Matlab

From observing Fourier transform graph, minimum sampling frequency was found to be 10 Hz where the reconstruction of continuous time signal is possible.

The Discrete Fourier Transform was also graphed. This time the fft() function in matlab was utilized. The continuous time signal that was reconstructed in Matlab and was inputted into fast fourier transform function and graphed as shown below Figure **??**. This time a double sided Fourier Transform was graphed.

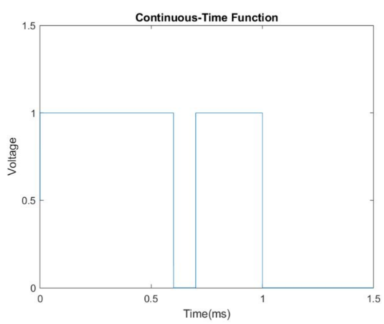

Fig. 4. Time-domain Signal in Matlab

The recovered reconstruction of the discrete time signals can be shown below with the graph of the recovered signal at minimum sampling frequency, below the minimum sampling frequency, and above.

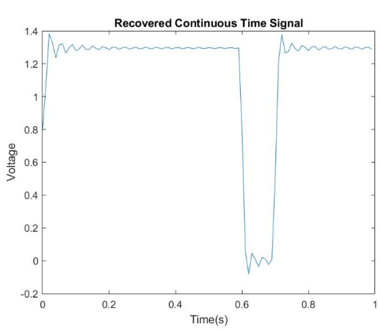

Fig. 5. Time-domain Signal in Matlab

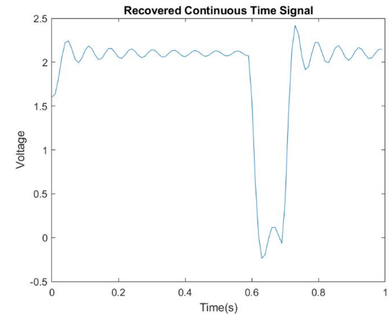

Fig. 6. Time-domain Signal in Matlab

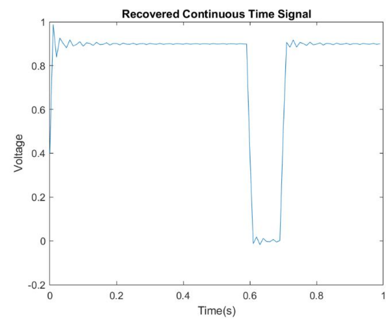

Fig. 7. Time-domain Signal in Matlab

C. Part 3: Spectral Analysis of AM and FM Signals

The first section of Part of 3 called for synthesizing an AM bandpass signal: s(t)=Ac[1+(Am)cos(2fmt)]cos(2fct) where fm=5 kHz, fc=25 kHz. The signal was inputted into Matlab with the set parameters and Ac and Am set to 0.25. The signal was then plotted and can be seen below.

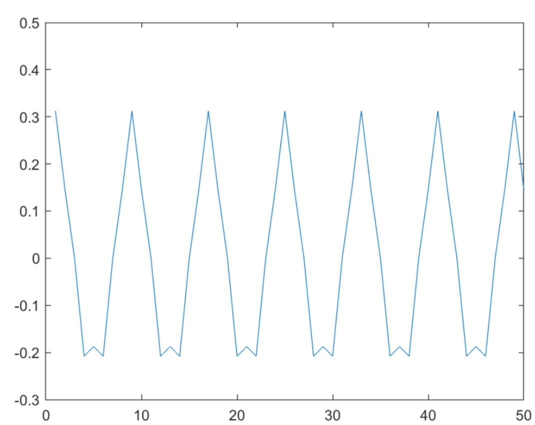

Fig. 8. AM Bandpass Signal

To obtain the spectral components of this signal, Matlab's fft function was used to plot the fft of the signal. This can be seen in the figure below.

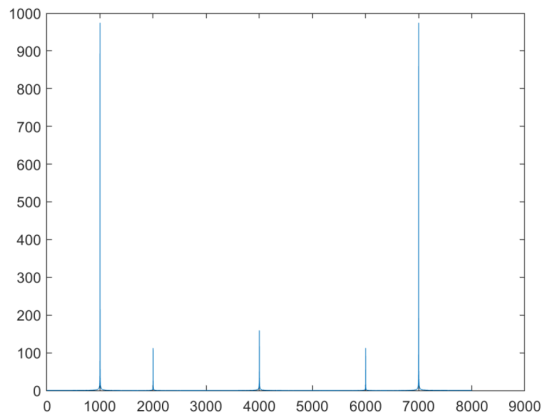

Fig. 9. FFT of AM Bandpass Signal

Noise was then added to the signal and then plotted as well as the fft of the signal with noise added, which can be seen in the figure below.

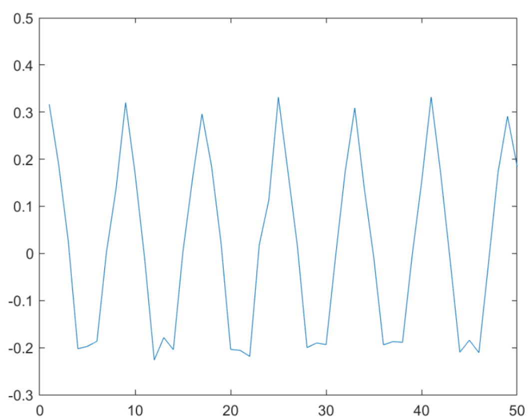

Fig. 10. AM Bandpass Signal with added noise

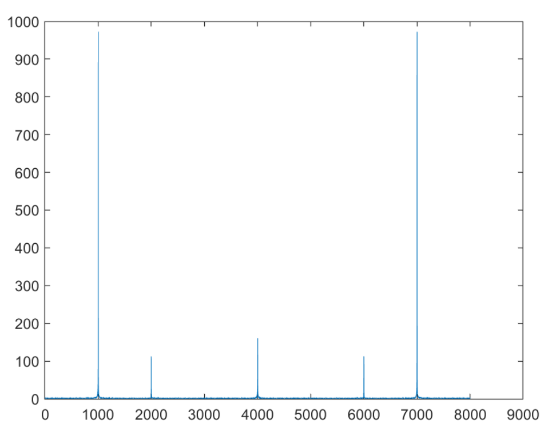

Fig. 11. FFT of AM Bandpass Signal with added noise

Looking at the signal with added noise, it can be seen that thew signal is distorted relative the original signal. Next, Part 3 called for experimenting with different values of fm, fc, and SNR to observe the results.
For the graphs shown above, the SNR was set equal to 20. When decreasing the value of SNR, the noise added to the signal is seen to have a greater effect. With an SNR set to 5, the signal become more distorted and can be seen below.

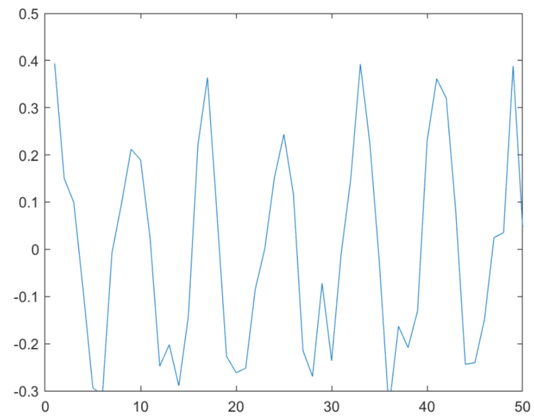

Fig. 12. AM Bandpass Signal with added noise and SNR = 5

As the SNR was increased, the noise began to be reduced and have less of an effect. When the SNR was inscreased to 100, the signal was identical to the original.

The last section of Part of 3 called for synthesizing an FM bandpass signal: s(t)=(Ac)cos[2fct+(bf)(Am)sin(2fmt)], where bf=Frequency Modulation Index. The signal was inputted into Matlab with the set parameters of bf = 10, fm = 5KHz, fc = 25kHz, and Ac and Am set to 0.25. The signal was then plotted and can be seen below.

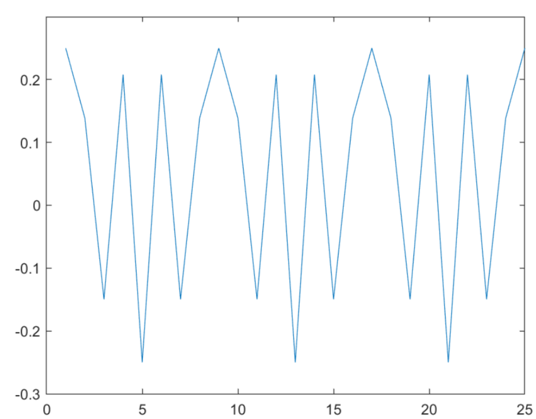

Fig. 13. FM Bandpass Signal

To obtain the spectral components of this signal, Matlab's fft function was used to plot the fft of the signal. This can be seen in the figure below.

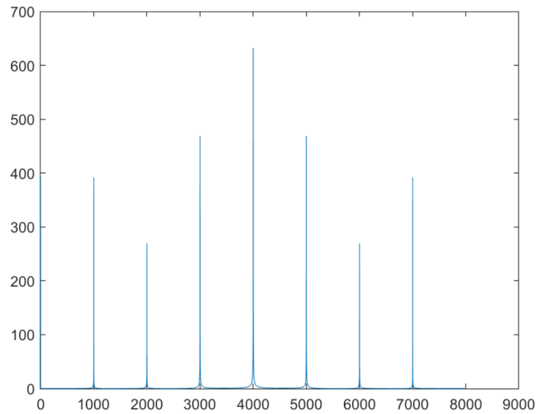

Fig. 14. FFT of FM Bandpass Signal

Noise was then added to the signal and then plotted as well as the fft of the signal with noise added, which can be seen in the figure below.

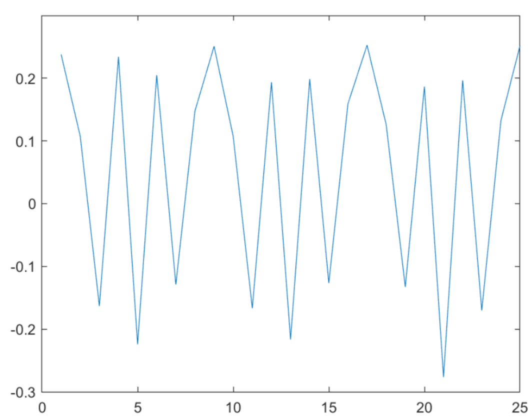

Fig. 15. FM Bandpass Signal with added noise

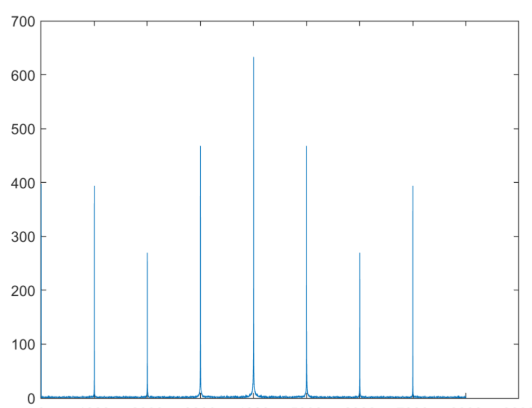

Fig. 16. FFT of FM Bandpass Signal with added noise

Lastly, Part 3 called for experimenting with different values of fm, fc, SNR and also bf to observe the results. For the graphs shown above of the FM signal, the SNR was set equal to 20. When decreasing the value of SNR, the noise added to the signal is seen to have a greater effect. With an SNR set to 5, the signal become more distorted and can be seen below.

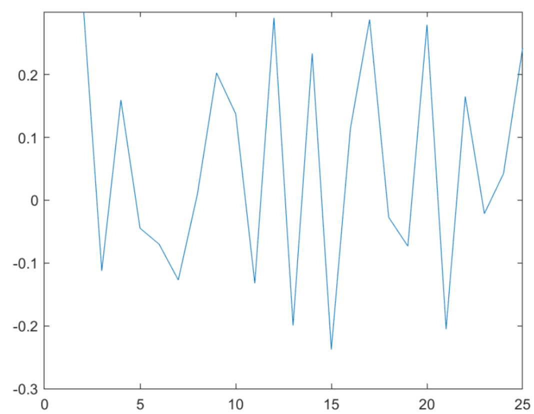

Fig. 17. FM Bandpass Signal with added noise and SNR = 5

As the SNR was increased, the noise began to be reduced and have less of an effect. When the SNR was inscreased to 100, the signal was identical to the original.
The bf was originally set to 10. When increasing and decreasing the value of bf, the appearance of the signal changes drastically.

D. Part 4: Spectral Analysis of Music Signal

Depicted in figure 18, the waveform of a 6 second clip with a sample size of 2^{15} Samples. The sampling frequency for this file (since it is a .wav) is 44.1 kHz.

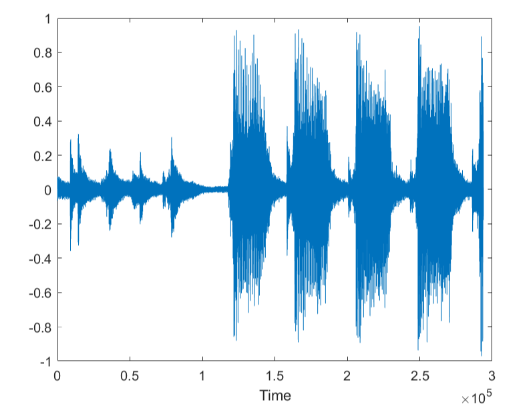

Fig. 18. Small clipping of an electronica version of Wish Lizst

Below is the MATLAB code used to generate these graphs.

```
f='trans.wav';          %file to inspect
[x,sr]=audioread(f);    %reads file
figure(1)
```

```
plot(x)                     %plots waveform
xlabel('Time');
ylabel('');

Ts=1/sr;
N=2^15;                     %Number of Samples
x=x(1:N)';                  %Limits to sample size
time=Ts*(0:length(x)-1);
figure(2)
magx=abs(fft(x));           %FFT magnitude
ssf=(0:N/2-1)/(Ts*N);
plot(ssf,magx(1:N/2))
xlabel('Frequency');
ylabel('Magnitude');
```

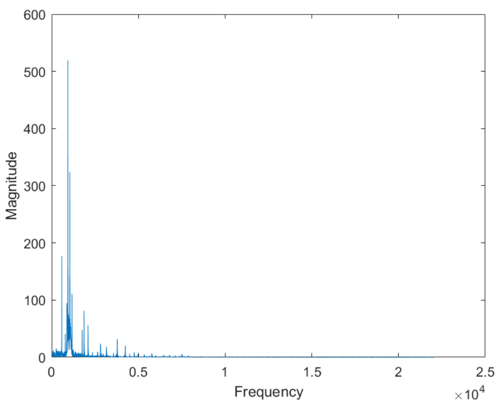

Fig. 19. Fast Fourier Transform of Electronica Wish Lizst

III. Analysis Conclusion

A. Part 1: Digital synthesis of arbitrary wave forms with specified SNR

First, the SNR will be varied. When the SNR is low, there is an audible tone although it is underneath a lot of "static" noise. Then as the SNR is increased, the tonal sound becomes well more clear and eventually the "static" noise is drowned out and not audible.

Next, the Frequency was varied. This in turn will increase/decrease the pitch of the sound output from MATLAB. Although, when working with the other signals given to us for this experiment, there are two frequencies that can be varied, being the carrier, and the message frequency. In essence, the carrier frequency should be much higher than the message frequency. While, varying these keeping that rule in mind, the trend still is the same, when you increase/deacrease the message frequency, the pitch increases/deacreases.

Lastly, when varying the amplitude the graph visually increases/decreases. When increasing the amplitude of a waveform, the graph amplitude increases, which in turn would increase the power/volume of the sound.

B. Part 2: Comparison between CFT and DFT (FFT)

The purpose of Part 2 was to compare the two Fourier transform technique, the Continuous Fourier Transform and Discrete Fourier Transform. The Continuous Fourier Transform, as shown in Figure **??** shows the frequency of the Fourier waveform that was plotted. Comparing that to the double-sided frequency of the DFT, as shown in Figure **??**, it can be seen that the graph of both Transform are similar. For the minimum frequency that the signal could be recovered is at 10 Hz. By observing that most of the information of the signal is before the 10Hz it would not be possible to adequately recover the signal if it was below this frequency. This is evident when the frequency of the sampling frequency was set to 5Hz then after it was set to 20Hz.

C. Part 3: Spectral Analysis of AM and FM Signals

The purpose of Part 3 was to synthesize a provided bandpass AM and FM signal while varying variables to observe the effect each has on the desires FM or AM signal. The first AM signal was, s(t)=Ac[1+(Am)cos(2fmt)]cos(2fct) where fm=5 kHz, fc=25 kHz, and the second FM signal was, s(t)=(Ac)cos[2fct+(bf)(Am)sin(2fmt)]. The variables altered were fm, fc, SNR, and bf. When decreasing the value of SNR, the noise added to the signal is seen to have a greater effect. As the SNR was increased, the noise began to be reduced and have less of an effect. Decreasing fm caused the signal to be more bounded. Decreasing fc, increased the frequency of the signal. When increasing and decreasing the value of bf, the appearance of the signal changes drastically.

D. Part 4: Spectral Analysis of a Music Signal

The purpose of Part 4 was to graph and analyze the frequency spectrum of a short clipping of music. As shown in figure 19 there are clear spikes in the 0-0.5 range, this song includes hard pounding guitar notes, along with various higher pitched piano keys. Therefore, on the FFT graph we can see the various notes played within the song, the most prevalent spike is shown at 932.65 (Hz), which equates to $A_5^\#$ which is a higher pitch piano sound. This piano note is repeatably heard throughout this clip of the song. Along with another clear spike at 586 (Hz), which is a lower darker note D_5. This sounds closely to the heavy guitar notes that appear in this clip.

Essentially the FFT of this waveform is a depiction of all the notes and frequencies that occur in this piece of music. This type of analysis is extremely important when it comes to creating music. This will tell the artist what notes appear in the song, and can help them adjust their instrumentation to match the particular frequencies that they desire.

IV. Appendix