TI Project Applications of Fast Fourier Transform

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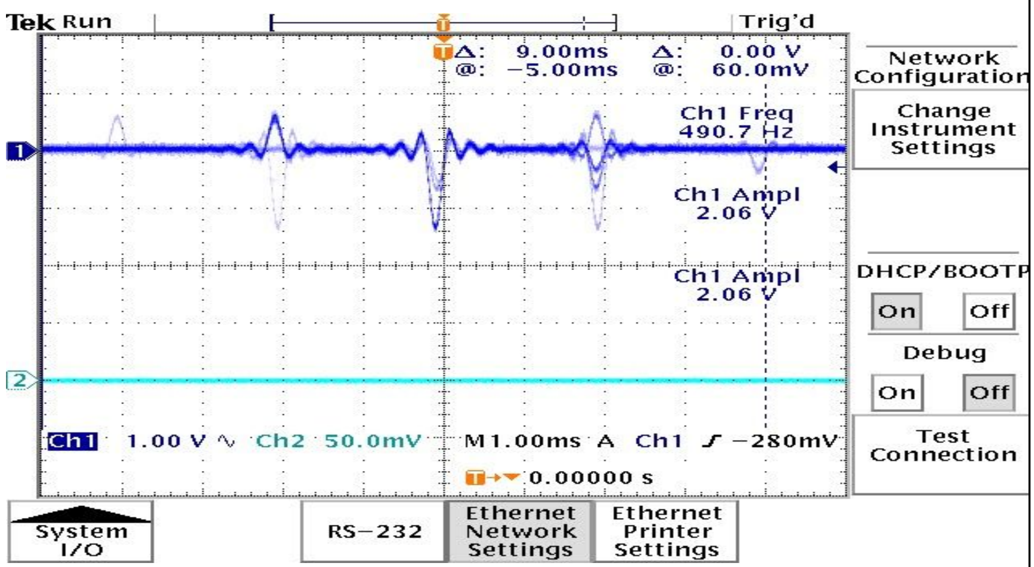
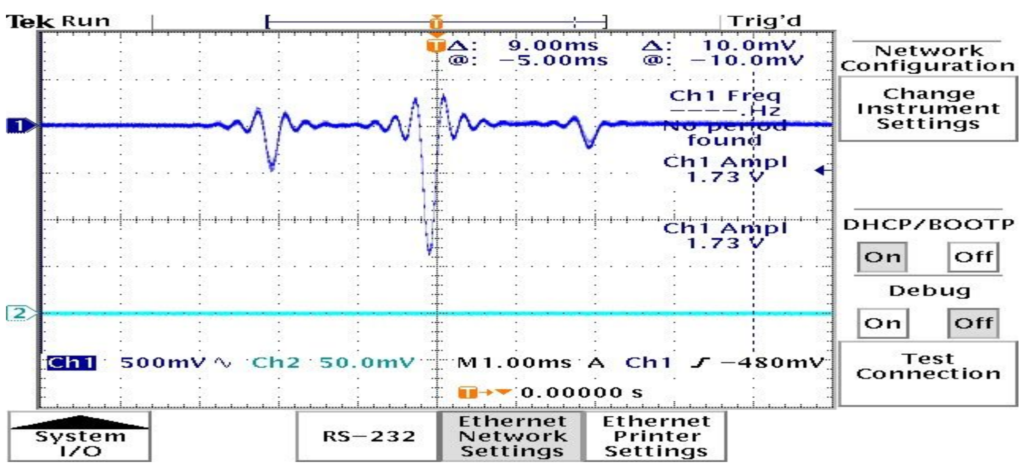
Objective:
- Real time spectrum analyzer.
- Pitch estimation of voiced speech segments.

Real time spectrum analyzer

- Open the “CPU Load Graph” under the DSP/BIOS menu in code composer studio, toggle switch 3 on/off a couple of times and check the CPU load. What is the CPU load when the spectrum is calculated?

The CPU Load when the spectrum is not calculated, ie, when Switch3 is off, is 24.49% ; whereas it rises to about 25.03% when it is turned on, ie, when the spectrum is calculated.

- Copy the code and modify such that the data acquired from the ADC and stored in the buffers (i.e.: gBufferRcvPingL) is multiplied by a specific window. Try each of the above windows and compare the result on the scope with the previous output (rectangular window where all the samples are multiplied by one). Comment about the obtained results

<table>
<tr><td>Window</td><td>Sine Wave convolved with the window</td></tr>
<tr><td>Rectangular</td><td>
</td></tr>
<tr><td>Hamming</td><td>
</td></tr>
</table>

Hanning	
Kaiser	

Observations :

We can see from the Oscilloscope plots of all the above windows convolved with the input sinewave that:
For Rectangular Window, since it has the shortest width of mainlobe and prominent side-lobes, there is lot of spectral leakage associated with the peaks in the spectrum.
For the Hamming and Hanning windows, eventhough the mainlobe width is bigger, they have very small side lobes, hence there is less spectral leage due to the side-lobes, thus the peaks can be extracted much more better than with the rectangular window.
The Kaiser window has the best recovering capability as it helps to maintain the width of the mainlobe and very small side-lobes in a good ratio to recover the peaks in the frequency domain.

- Now change the input signal from sine to square, what do you observe?

Window	Square Wave convolved with the window
Rectangular	

Hamming	
Hanning	

Kaiser	

Observations :

We can see from the Oscilloscope plots of all the above windows convolved with the input squarewave that:
The main difference between convolving the windows with a sine or a square-wave is that, in the frequency domain, you are convolving a digital sinc along the peaks, but with a square wave, you are still convolving a digital since, but instead of the impulse peaks in the frequency domain, you will have a sinc functions as the DFT of a square wave is a digital sinc function , which is the reason who will see two peaks at each position at the frequency domain for each of the windows. The rest of the impact of the windows remain the same as with convolving with the sine wave.
For Rectangular Window, since it has the shortest width of mainlobe and prominent side-lobes, there is lot of spectral leakage associated with the peaks in the spectrum.
For the Hamming and Hanning windows, even-though the mainlobe width is bigger, they have very small side lobes, hence there is less spectral leage due to the side-lobes, thus the peaks can be extracted much more better than with the rectangular window.
The Kaiser window has the best recovering capability as it helps to maintain the width of the mainlobe and very small side-lobes in a good ratio to recover the peaks in the frequency domain.

- ❖ Use a.wav and i.wav to calculate first three formant frequencies.

For a.wav

The first 3 Formant Frequencies are:

Formant Frequency = Index* Sampling Rate you used* Sampling Rate of the Codec / N
Sampling Rate of codec - 48KHz ; Sampling rate used - 8KHz, N - 256
F1 -> 400 Hz
F2 -> 800 Hz
F3 -> 2550 Hz

For i.wav

The first 3 formant frequencies are:
F1 -> 600 Hz
F2 -> 900 Hz
F3 -> 2550 Hz

Pitch Estimation

- ❏ What is the obtained pitch?

The obtained Pitch is the index at which the maximum peak of the autocorrelation function obtained due to the IFFT of the power spectrum which is in-turn obtained after taking the DIF-FFT of the input signal.

As we can see, the index at which the maximum peak occurs is at 67.

Now, we can tell this only through observation. But we can also tell this through coding it through an Assembly function which finds the index at which the maximum peak occurs.

Below is the Output:

The Maximum peak’s index is stored in the temporary register T0. As we can see from the Memory window below, at T0, is a Hexadecimal value of 0043, which corresponds to 67 in decimal, which is the expected result, and hence the pitch of the signal.

Pitch value stored in T0.

Conclusion

From this Project, we saw the implication of obtaining the peaks of sine and a square wave from the frequency domain due to convolution with various windows, and verified which window was best at this. Then we went to find out the first 3 formant frequencies of 2 given input samples corresponding to \a and \i vowels.
Lastly we estimated the pitch of the signal through the autocorrelation function obtained through DIF-IFFT of the Power spectrum, which is obtained through DIF-FFT of the input signal.