Lab 2 – DT signal frequency, Fourier transform

Entry test example questions

1. $x_a(t) = cos(2\pi f_a t)$ was sampled with sampling period T_s. Plot the { spectrum | N-point DFT } of $x[n]$ (f_a, T_s or f_s given, their proportion rational or irrational..., N given)

2. A signal $x(n)$ with known Fourier spectrum $X(\theta)$ has been {inverted in time | decimated | modulated | ...}. Express mathematically what happened to the spectrum.

3. Calculate a DFT of a simple finite signal (by pen and paper...)

Exercises

(First exercise is a repeat from end of Lab 1 - if you have done it then, skip now.) *Italics denote optional tasks.*

1. Implement $y(n) = a \cdot y(n-1) + x(n)$, accepting a and initial y as parameters. Test the response with one value of a such as $0 < a < 1$ and:

   (a) zero initial condition and $\delta(n)$ at input

   (b) non-zero initial cond. but zero at input

   (c) non-zero initial cond and $\delta(n)$ at input

2. Experiment with different values of a. $(1, -1, > 1, < 0$ etc.).

3. Plot a DT sinusoid with normalized frequency $f_n = f/f_s$ equal to 0.1, 0.3, 0.5, 0.9, 1.1, 2.1 ($\theta = 2\pi f_n$, $x(n) = sin(n\theta)$). Note number of samples in period. Explain the plots - try to draw (by pencil) the underlying CT signal on your plot copied from screen. *If you are brave enough, draw the underlying CT signal with Matlab using 9 additional samples between original ones.*

4. Use program `anator` to display real-time signal and its spectrum – measure a sinusoid with different relations of f and f_s ($f < f_s/2$, $f \approx f_s/2$, $f > f_s/2$, etc.). Comment the plots. ($f_s \approx 38kHz$)
   (anator-¿device-¿signal analyzer...)

5. Simulate 2 ms of samples of a single square impulse of 1 ms length, sampled with:

   (a) 1 MHz

   (b) 10 kHz

   (c) 10 kHz, but use 4 ms of samples

Remark: first, calculate by pencil and imagine (or even sketch) the signal, then produce it using ones(), zeros(), and [] operators in Matlab. Claculate (with Matlab) and plot amplitude of FFT's of all signals on one graph, keeping the real-world frequency axes the same and scaling the 5a signal 100 times down. *Find out from the FFT definition why the scaling is necessary (compare different length FFTs of a DC signal).*

Think of 5a as "almost CT" signal and comment the spectrum differences.

6. Plot an FFT of 1024 points of following signals:

   (a) a 512 points square impulse

   (b) other (narrower) square impulses

   (c) sine wave (integer and non-integer number of periods in window)

   (d) $e^{jn\theta_c}$ (how many peaks do you see? why?) Try different values of $0 < \theta_c \leq \pi$.

   (e) a 32-point square impulse beginning at 0

   (f) a 32-point square impulse beginning at $N_0 > 0$

   (name the effects, note the number of zero places in spectrum etc.)

7. Plot a spectrum of 512 samples of sine wave. Then, zero-pad them to 1024 and 2048 samples. Compare the results. Compute IFFT. (plot real part of IFFT to cut off arithmetic errors). Hint: `fft(x,L)` automatically zero-pads signal x to length L.

8. Compute spectra of different windows. Note mainlobe width, sidelobe attenuation etc.

   (If you have enough time, use Matlab: `hamming, bartlett, blackman, hanning, kaiser`, otherwise use Windows program "anator").

9. Do the following experiments to see the effect of windowing:

   (a) Plot a spectrum of 512 samples of sine wave. Choose the frequency to see the rectangular window effect clearly. If necessary, use zero-padding to see the spectrum better.

   (b) Use different window shapes, trying to obtain good, clear plot of the spectrum.

   (c) Demonstrate the signal separation properties of different windows - plot a spectrum of a sum of two sinusoids with similar frequencies and amplitudes, then with very different frequencies and amplitudes.

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