Lab 1 – DT signals, LTI systems, frequency

For help, use `help <subject>`, note that UPPERCASE is used to mark keywords in help only, not in Matlab....

For plotting DT signals, use markers (`plot(n,x,'o')` etc.). For the continuous couterparts, use lines.

Italics denote optional tasks.

1. Using matlab, plot 100 samples of 200 Hz square wave sampled at 2 kHz. Note the period and normalized frequency.

2. The same for sine wave.

3. Write an m-file with a function automating the generation of a sinusoid with given f, f_s and number of computed samples. *Provide defaults for parameters. Plot result if output is unused.*

4. Use A/D converter to get signals (as in 1 and 2) from a generator. Compare simulated and real-world plots. Use Matlab's command: `y=getdata(Nsamples_in_block, [Kblocks, [Tsampling, [leave_bias]]])` (Tsampling is in seconds, "[ ]" denote optional arguments).

5. Label an x-axis of above plot with time units, then repeat with sample indices (hint: `plot(xvalues, yvalues, 'marker');`).

6. Simulate $sin\theta_0 n$ where $\theta_0 = 0.1 \cdot 2\pi$, make plot.

7. Find another $\theta_1 \gg \theta_0$ such that $sin\theta_1 n = sin\theta_0 n$ for $n \in \mathrm{I}$. Try to use $\theta_1 < 20 \cdot \theta_0$

8. *Prove the θ's inequality, plotting (cont. line) the "visibly continuous" sinusoids – i.e. putting 10 times more samples at non-integer points in "time" (use `n10=0:.1:(N-1)` for "continuous" and `n=0:(N-1)` for "discrete" time)*

9. Plot $x(n) = sin\theta n$ where $\theta = p \cdot 2\pi$, p being a rational number. Choose p value and n range so that at least two periods $x(n + N) = x(n)$ are visible, and avoid undersampling effects.

10. Explain the plot, comparing the periods of the underlying CT signal and DT signal.

11. *Use real (non-rational) p_1 close to p. Explain the difference.*

12. write m-files implementing lecture examples of DT systems:

multiplier	$y(n) = 3 \cdot x(n)$
two sample averager	$y(n) = \frac{x(n)+x(n-1)}{2}$
M sample averager	$y(n) = \frac{1}{M}\sum_{k=0}^{M-1} x(n-k)$
compressor	$y(n) = x(Mn)$
FIR filter	$y(n) = \sum_{k=0}^{M} h(k) \cdot x(n-k)$
square value	$y(n) = (x(n))^2$

Note: FIR filter is a lecture example limited to finite length $h[k]$

13. Make some experiments testing L and TI properties of above systems.

14. Plot impulse responses of all systems of item 12

15. Implement an accumulator and test it with $\delta[n]$ and $u[n]$.

16. Implement $y(n) = a \cdot y(n-1) + x(n)$, accepting a and initial y as parameters. Test impulse response with zero initial condition, initial cond. response, then the combination of both for $0 < a < 1$.

17. Experiment with different values of a.

18. *Implement "from scratch" a convolution of two series. Compare results with* `conv`*. Check timing (*`help etime`*), and* `flops`

19. Use convolution (`conv()`)to find a response of an M sample averager to a sequence with four nonzero samples. Check results against the implementation of item 12.

20. *The same with system of item 16. Q: can you do it exactly?*

21. Use program `anator` to display real-time signal and its frequency (well, FFT, but you'll learn it later) – measure a sinusoid with different relations of $1/Ts$ and f

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