Transform Methods & Signal Processing
Class Exercise 3:
Hand in before lecture, Monday 7th Sept

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Note, questions marked by a (*) are harder than normal questions, and are for masters students. Bonus marks may be awarded to other students who solve these.

1. $\boxed{\text{7 marks}}$ Find (by hand) the DFT of the following signals (using the simplest possible method you can think of):

(a) $(0, 0, 1, 1)$

(b) $(1, 0, -1, 0, 1, 0, -1, 0, 1, 0, -1, 0, 1, 0, -1, 0)$

(c) $(1, 2, 1, 0)$

(d) $(1, i, -1, -i, 1, i, -1, -i)$

2. $\boxed{\text{3 marks}}$ A new type of high-quality digital audio media is proposed. To convert to this media, an analogue audio signal is passed through a low-pass filter, and then sampled at 50 kHz, with a 24 bit fixed point representation of the samples.

(a) What should be the stop-band of the low-pass filter?

(b) If the filter is intended to reduce aliasing noise to the same level as the quantization noise (assuming a full load, with no clipping), then what should be its stop-band attenuation? (refer to lecture 5 if some of this termino0logy is unfamiliar).

(c) Given a sample 0.1 seconds long, what would be the lowest resolvable frequency?

3. $\boxed{\text{10 marks}}$ Read the following web pages, and where needed follow links, or perform your own search for information.

`http://georgegraham.com/compress.html`
`http://en.wikipedia.org/wiki/Loudness_war`
`http://www.cdmasteringservices.com/dynamicrange.htm`
`http://www.cdmasteringservices.com/dynamicdeath.htm`

Measure the RMS power of 3 music tracks (preferably from significantly different years), and provide a table showing the date of release of the music, and its RMS power relative to the maximum possible (unclipped) signal strength. For some helpful hints on how to do this exercise, please see the bottom of this sheet.

In addition, write a one-page article describing your methodology and results, explaining the key issues highlighted in the web pages and relating this to what you have learnt in lectures. Please use your own words. *Do not plagiarise!* Marks will be given for content, and presentation.

4*. $\boxed{\text{5 marks}}$ Prove that the Fourier transform of a Dirac comb (with unit spacing) is a Dirac comb (with unit spacing). Note that the comment in your notes (lecture 3) is not sufficient for a proof. Hint, it helps to also use Fourier series.

Notes on audio capture and manipulation:

- Audio capture may be done using many tools:
  - from CD use a CD 'ripper' (e.g. `http://www.xiph.org/paranoia/`
  - From audio input, e.g. using arecord (for Unix), or GarageBand (for Mac), or Matlab routine `wavrecord` for Windows.
- General audio manipulation tools:
  - `http://audacity.sourceforge.net/about.php`
  - GarageBand for Mac
- I suggest you use Matlab for measuring the power of the signals.
  - `http://www.mathworks.com/access/helpdesk/help/techdoc/ref/wavread.html` to read a WAV file
  - `http://www.mathworks.com/access/helpdesk/help/techdoc/ref/wavwrite.html` to write a WAV file
  - `http://www.mathworks.com/access/helpdesk/help/techdoc/ref/sound.html` to play a sound.
  - The following is an example of the use of these routines

```
file = 'example.wav';
[y, Fs, bits] = wavread(file);
dt = 1/Fs;

fprintf('file = %s\n', file);
fprintf(' sampled at %d bits at %d Hz\n', bits, Fs);

sample_times = (1:length(y(:,1)))/Fs;
figure(1)
plot(sample_times, y);
xlabel('seconds');
```