ADSP **高等數位訊號處理**, Spring 2017 Homework #3

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- (1) MATLAB code (請參考最底下)
- (2) Suppose that there are three vocal signals: (i) $\cos(200\pi t)$ (ii) $\sin(1000\pi t)$ (iii) $\sin(4000\pi t)$
- (a) Which voice sounds louder?
 - 三個信號聽起來音量應該相同,因為amplitude一樣。
- (b) Which voice signal can be propagated in a longer distance? 波長越長就可以傳越遠。因為 (i)的頻率是100hz、波長最長,所以可以傳最遠。
- (3) In addition to DCT, which is adopted by MP3, write at least three possible ways that can compress a music signal more efficiently.
- 1. 把每個音的頻率記錄起來,還有他的harmonic frequency也紀錄、這樣就可以回復 原始音色。
 - 2. 每半度音差 21/12倍,利用頻率之間關係壓縮。
 - 3. 歌曲重複部分直接代換,就不用一直壓縮同旋律的片段。
- (4) In the JPEG process, (a) why the DCT is used instead of the DFT and the KLT for transformation? (b) Why the input image is separated into several 8x8 blocks before using the DCT?
 - (a) DCT的特點像是係數都是實數(方便運算)是可以贏過DFT的;而KLT因為需要紀錄transform matrix耗空間、所以用DCT也較佳。
 - (b) 運算複雜度下降、用8x8就可以讓buffer size較小、而且可以表現該部分特性,影像切成小塊的,在做DCT就可以看出哪些地方高頻、哪些低頻。

(5) (a) Why the normalized root mean square error (NRMSE) may not reflect the similarity between two images?

因為NRMSE他本身是受pixel的差異影響的、而且和x, y有關,只要 pixel value不同就會造成NRMSE太大,所以兩張影像直接比pixel value的話可能用NRMSE就不太好。

(b) Can the NRMSE measure the similarity between two audio signals? Why?

答案是無法,因為對人來說雖然只是有time delay晚一點出來沒什麼差別,而frequency shift 音高變化就聽得出來、amplitude只聽得出大小聲,但是在NRMSE就會因為點對點差異,也就是y, x 在時間上的delay、或是frequency shift, amplitude大小這些種種因素都可以讓NRMSE大幅改變,所以不適合用來表現兩個聲音的相似度。

(6) Suppose that $P(x = n) = e^{-\lambda} \lambda^n / n!$ for $\mathbf{n} = \mathbf{0}, \mathbf{1}, \mathbf{2}, \mathbf{3}, \dots$ where $\lambda = \mathbf{0}.98$. Also suppose the **length**(\mathbf{x}) = **10000**. Estimate the range of the total coding lengths in binary system when using (i) the Huffman code and (ii) the arithmetic code.

附錄: Problem(1) Matlab code

```
clc;
clear;
% score=[1,1,5,5,6,6,5]; %1:Do,2:Re,3:Mi,....
% beat=[1, 1, 1, 1, 1, 1, 2]; % tempo
score=[1,1,5,5,6,6,5, 4, 4, 3, 3, 2, 2, 1]; %1:Do,2:Re,3:Mi,.....
beat=[1, 1, 1, 1, 1, 1, 2, 1, 1, 1, 1, 1, 1, 2]; % tempo
name = 'twinkle';
getmusic(score, beat, name);
function getmusic(score, beat, name)
    fs = 8000;
    t = 0:1/fs:0.25;
    tspace = .05;
    fr = 2^{(1/12)};
    A4 = 440;
    B4 = A4*fr^2;
    C4 = A4*fr^{-9};
    D4 = A4*fr^{(-7)};
    E4 = A4*fr^{-5};
    F4 = A4*fr^{(-4)};
    G4 = A4*fr^{(-2)};
    C5 = A4*fr^3;
    keySet = \{1, 2, 3, 4, 5, 6, 7, 8\};
    valueSet = \{\cos(C4*2*pi*t), \cos(D4*2*pi*t), \ldots
        cos(E4*2*pi*t), cos(F4*2*pi*t), ...
        cos(G4*2*pi*t), cos(A4*2*pi*t), ...
        cos(B4*2*pi*t), cos(C5*2*pi*t);
    mapObj = containers.Map(keySet,valueSet);
    xspace = zeros(size(tspace));
    res = [];
    for i = 1:size(score, 2)
        for j = 1:beat(i)
          res = [res, mapObj(score(i)), xspace];
        end
    end
    str = '.wav';
    tmp = strcat(name, str);
    audiowrite(tmp, res, fs);
end
```