HW2 Report

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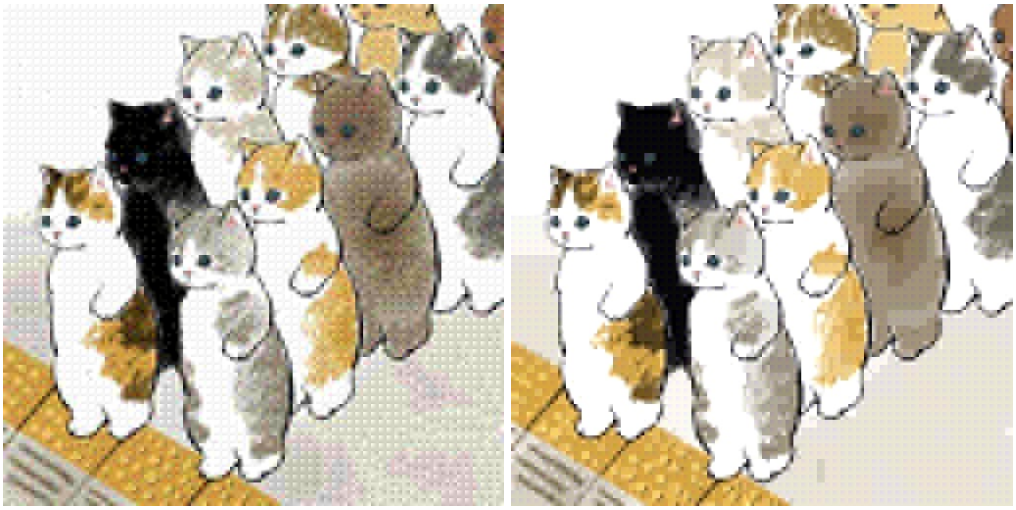
1. DCT image compression

Compare the differences between the results with DCT compression performed in two color spaces in (a) and (b) and discuss the results.

理論上，(b)的做法有先將圖片轉至YCbCr空間，圖片的亮度和色度先被分離，讓我們在做2D-DCT壓縮時高頻分量集中在色度，而低頻集中在亮度，壓縮的效果會相比於(a)的作法更好。

1. cat_n2m4 (a) / (b)

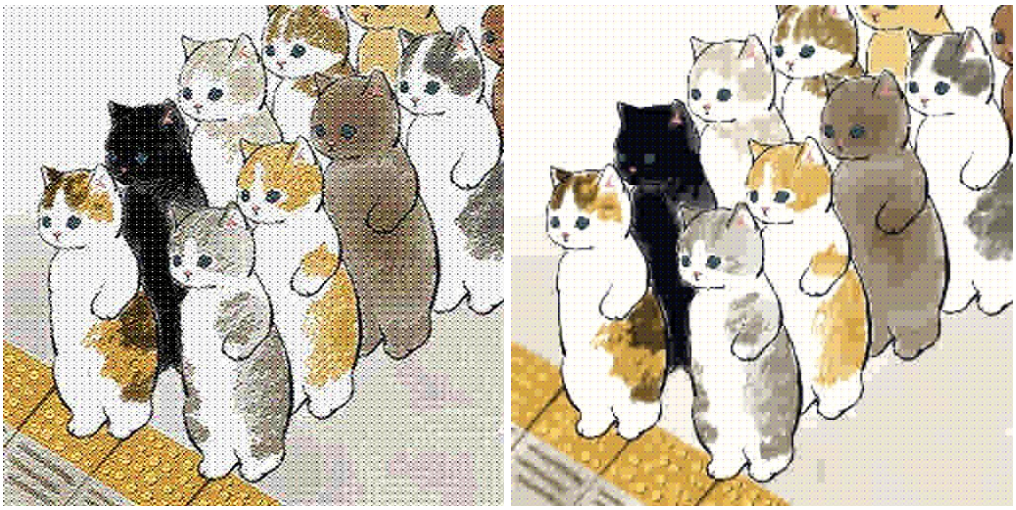


這裡很明顯的可以看到(a)作法使得圖片中每個8*8的方格明顯的左上較亮，我想這是在2D-DCT過程中產生的亮度失真，而在(b)的結果中這個問題並不嚴重，可知透過先把圖片轉至YCbCr空間確實能夠在壓縮過程中有更好的結果。

2. bar_n4m8 (a) / (b)

因為壓縮過程中的失真主要是由quantization造成，在m提高後可以有很好的改善，這兩個結果對比下肉眼就看不出什麼區別了。

3. cat_n4m4 (a) / (b)



在m=4的時候明顯看得出圖片中的方格感，但是(a)的作法在亮度的失真上明顯比(b)嚴重。

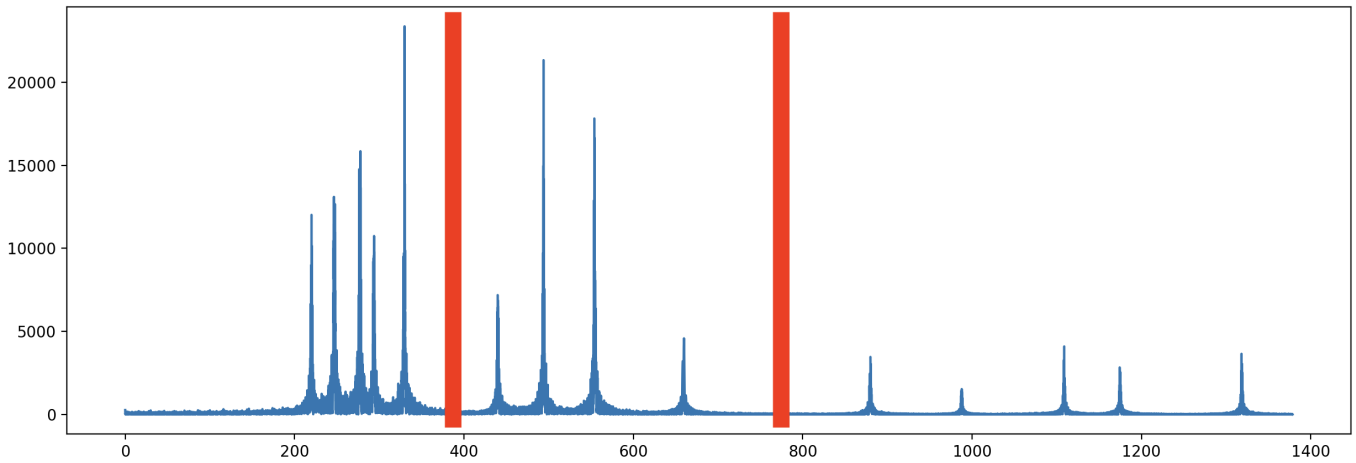
Compression rate and PSNR values

```
cat_N2_M4_A -- compression ratio: 32, PSNR: 29.9718
cat_N2_M8_A -- compression ratio: 16, PSNR: 32.9167
cat_N4_M4_A -- compression ratio: 08, PSNR: 29.0216
cat_N4_M8_A -- compression ratio: 04, PSNR: 33.9120
bar_N2_M4_A -- compression ratio: 32, PSNR: 30.2938
bar_N2_M8_A -- compression ratio: 16, PSNR: 31.3996
bar_N4_M4_A -- compression ratio: 08, PSNR: 29.7751
bar_N4_M8_A -- compression ratio: 04, PSNR: 33.0103
cat_N2_M4_B -- compression ratio: 64, PSNR: 29.9664
cat_N2_M8_B -- compression ratio: 32, PSNR: 29.6289
cat_N4_M4_B -- compression ratio: 16, PSNR: 29.7525
cat_N4_M8_B -- compression ratio: 08, PSNR: 29.7852
bar_N2_M4_B -- compression ratio: 64, PSNR: 29.1305
bar_N2_M8_B -- compression ratio: 32, PSNR: 29.5969
bar_N4_M4_B -- compression ratio: 16, PSNR: 28.3580
bar_N4_M8_B -- compression ratio: 08, PSNR: 29.7928
```

在PSNR的部分全部的圖片都差不多是在30左右，應該算是還ok。

2. Create your own FIR filters to filter audio signal

1. Discuss how you determine the filters.




```
# build filter
filter1 = constructFilter(freq_sample, 10001, 'low-pass', 400)
filter2 = constructFilter(freq_sample, 10001, 'high-pass', 800)
filter3 = constructFilter(freq_sample, 10001, 'band-pass', 400, 800)
```

我在觀察input的spectrum後分別使用low pass, high pass, band pass分離出頻率中的三個區間，由上圖的spectrum可以看到三個區間的分隔點。

2. How you implement the filter and convolutions to separate the mixed song and one/multiple fold echo?

filter

基本上是參考講義中的作法:

`constructFilter(freq_sample, windowSize, mode, *freq_cut)`

```
if mode == 'band-pass':
    (freq_cut1, freq_cut2) = freq_cut
    freq_cut1, freq_cut2 = freq_cut1 / freq_sample, freq_cut2 /
freq_sample
else:
    (freq_cut,) = freq_cut
    freq_cut = freq_cut / freq_sample
middle = windowSize // 2
fltr = np.zeros(windowSize)
for n in range(- windowSize // 2, windowSize // 2 + 1):
    if n == 0:
        fltr[middle] = 1
    else:
        if mode == 'low-pass':
            fltr[n + middle] = np.sin(2 * np.pi * freq_cut * n) / (np.pi *
n)
        elif mode == 'high-pass':
            fltr[n + middle] = -np.sin(2 * np.pi * freq_cut * n) / (np.pi
* n)
        else:
            fltr[n + middle] = np.sin(2 * np.pi * freq_cut2 * n) / (np.pi
```

```
* n) - np.sin(2 * np.pi * freq_cut1 * n) / (np.pi * n)
if mode == 'low-pass':
    fltr[middle] = 2 * freq_cut
elif mode == 'high-pass':
    fltr[middle] = 1 - 2 * freq_cut
else:
    fltr[middle] = 2 * (freq_cut2 - freq_cut1)
for i in range(windowSize):
    fltr[i] *= (0.42 - 0.5 * np.cos((2 * np.pi * i) / (windowSize - 1)) +
0.08 * np.cos((4 * np.pi * i) / (windowSize - 1)))
return fltr
```

這裡的windowSize我都是使用10001，再用更高的order也聽不太出區別了。

convolution

我使用fft後相乘再ifft的做法，算的比較快

```
def convolution(audio, filter)
```

```
filter_pad = np.zeros(len(audio))
filter_pad[:len(filter)] = filter[::-1]
filtered_audio = np.fft.ifft(np.fft.fft(audio) *
np.fft.fft(filter_pad)).real
return filtered_audio
```

其中必須先將filter加上padding使他和音頻的長度相同。

echo

同樣是講義的方法

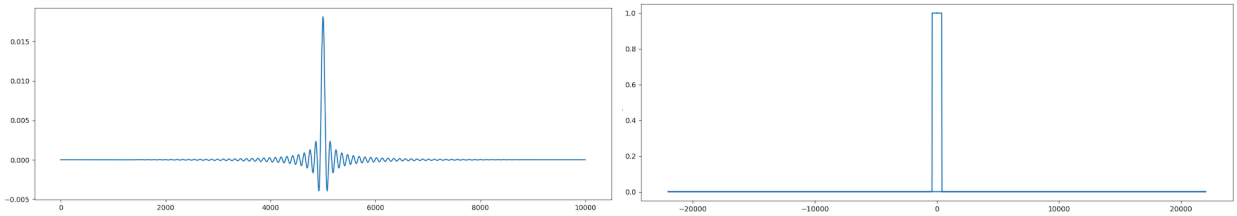
```
def echo(signal, mode='one')
```

```
delay = 12800
newSignal = np.zeros_like(signal)
newSignal[:delay] = signal[:delay]
for i in range(delay, len(signal)):
    if mode == 'one':
        newSignal[i] = signal[i] + 0.8 * signal[i - delay]
    else:
        newSignal[i] = signal[i] + 0.8 * newSignal[i - delay]
return newSignal
```

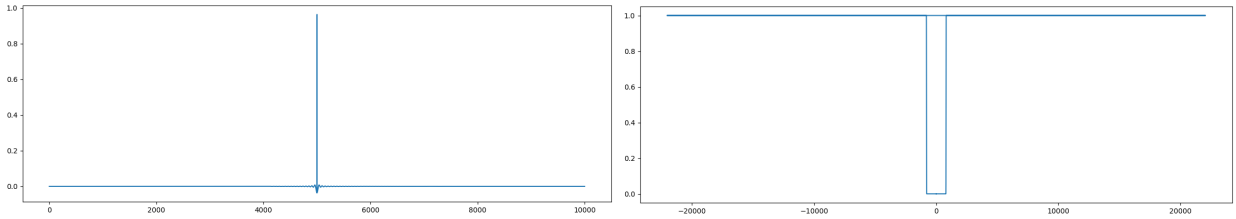
就是將音頻加上去疊起來，這裡delay我用12800比較聽得出回音。

3. Compare spectrum and shape of the filters.

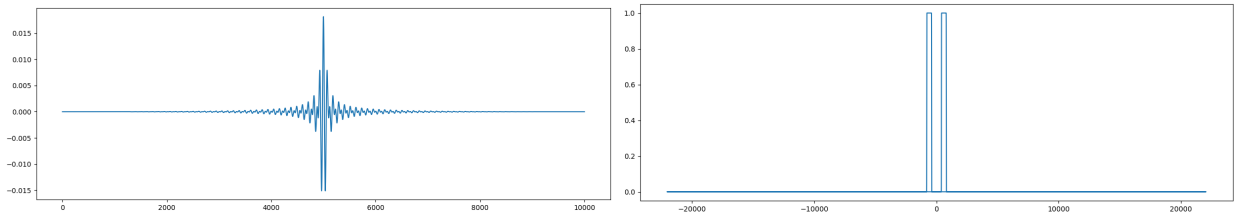
Low pass shape / spectrum



High pass shape / spectrum



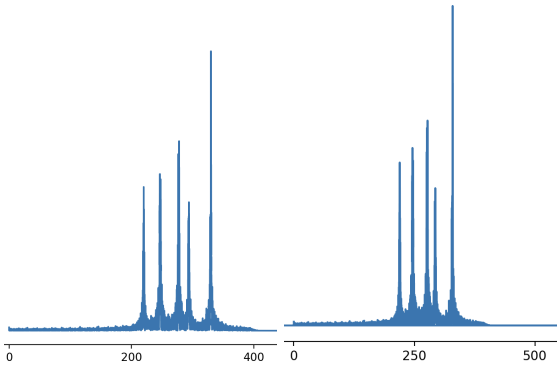
Band pass shape / spectrum



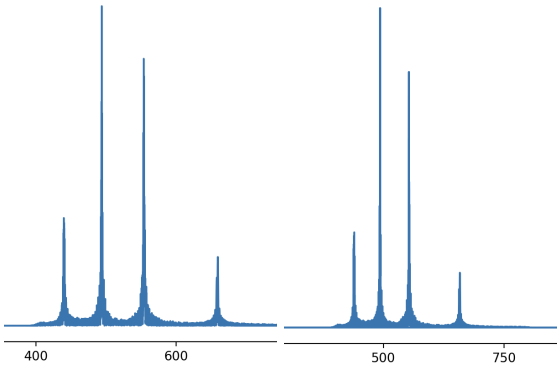
可以看到這些filter的shape和spectrum符合各自的特徵，spectrum更可以清楚看出是讓哪部分的頻率通過，如low pass只讓threshold以下的頻率通過，high pass則相反，band pass則是只讓一個區間的通過。

4. Briefly compare the difference between signals before and after reducing the sampling rates.

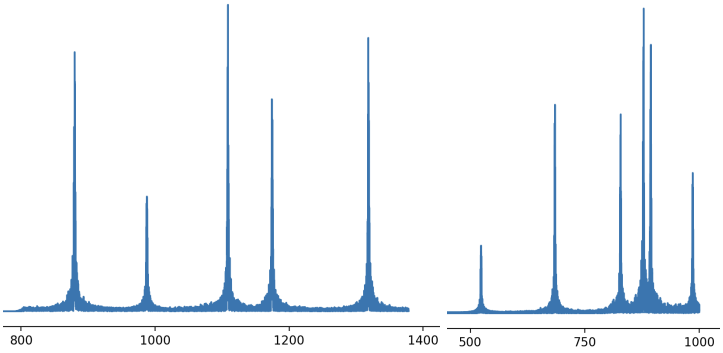
Low pass before / after



Band pass before / after



High pass before / after



根據Nyquist Theorem，用於sample的頻率至少要為取樣音頻中最高頻率的兩倍，而在low pass和band pass中取出的音頻最高頻率分別是350Hz和670Hz左右，因此Nyquist frequency分別是700Hz和1340Hz，以題目要求的2000Hz降取樣出的音頻跟原本的並無太大差別。但是在high pass取出的音頻中最高頻率約為1330Hz，Nyquist frequency為2660Hz，因此無法以2000Hz取樣出和原本相同的音頻，聽起來就和原本的音頻差非常多，整體的頻率降低非常多，這是sample rate太低造成的aliasing。