Assessment (ECE2111)

Lab 05 Result Document

MONASH University

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ECE2111 lab5 results document:

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Section 1:

Question 2: Find the average power of the signal `speech`

The average power of the signal (speech) would be 0.022 W.

Question 3: What value of the amplitude A ensures that the power of the sinusoidal disturbance is one tenth of the power of the signal speech.

The value of the amplitude A that ensures that the power of the sinusoidal disturbance is one tenth of the power of the signal speech is 0.0210.

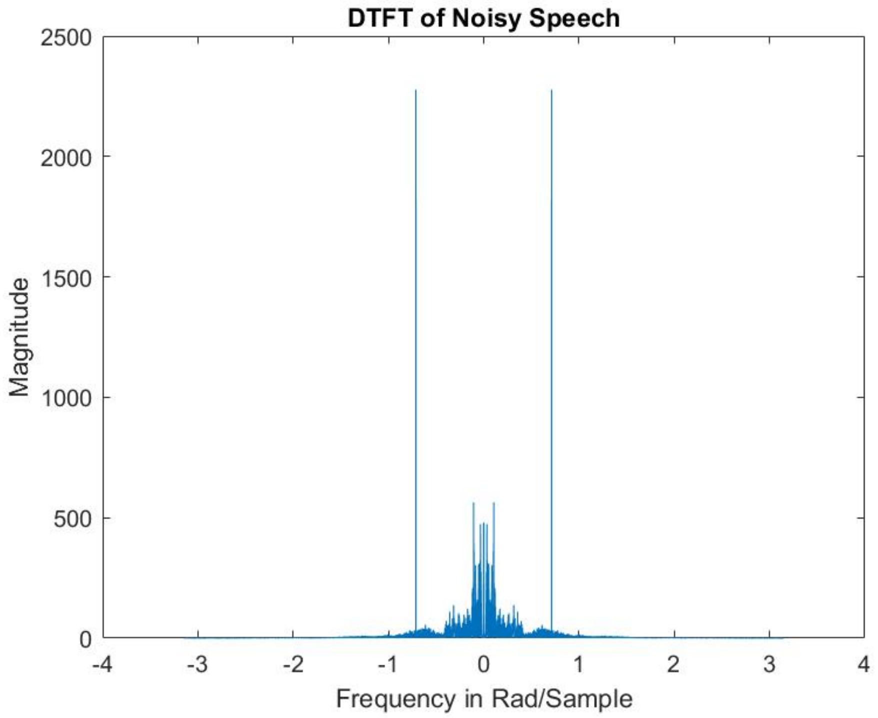
Workings:

The formula for the average power of a cosine signal is A^2/2.

So, A^2/2 = 0.1*0.0022.

A would be equal to approximately 0.0210.

Question 4: Include, below, a plot of the noisy signal in frequency domain (magnitude only) with frequency axis in rad/sample. Use the second type of plot discussed in lab 4. Ensure your plot is labeled.



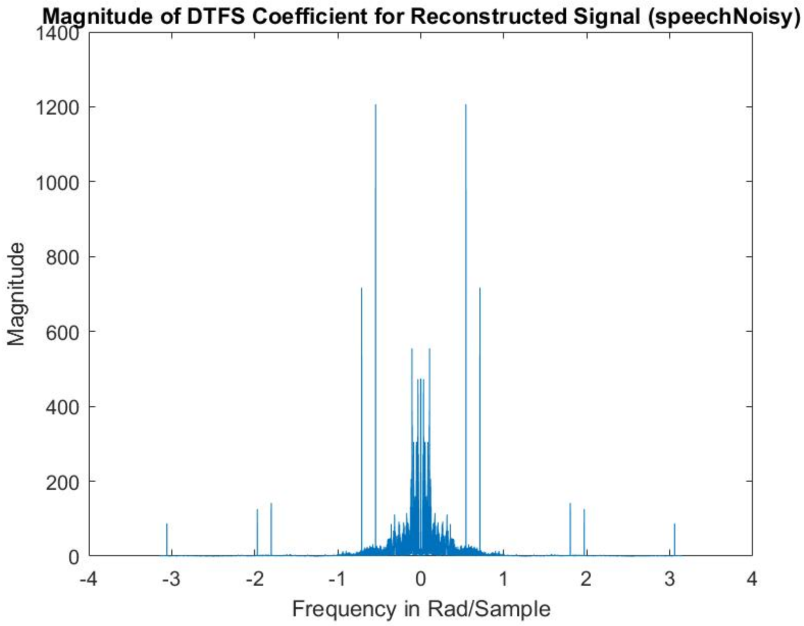
Question 5: Does the sinusoidal noise sound different after downsampling?

Yes, the sinusoidal noise will be distorted. The sinusoidal noise is distorted due to aliasing, leading to the distortion of the signal as the low frequency signal will overlap with the high frequency signal. The quality of the signal itself will also be reduced as downsampling would increase the Nyquist frequency to 8820π rad/s.

Question 7a: Play the reconstructed signal using `soundsc`*. How does it compare with the original signal* `speechNoisy`*?*

The reconstructed signal sounds better but it still has distortion when compared to the original signal and has a lower pitch.

Question 7b: Include, below, a plot of the reconstructed signal in the frequency domain (magnitude only) with frequency axis in rad/sample. Use the second type of plot discussed in lab 4. Ensure your plot is labeled.



Question 7c: Can the reconstruction process undo the effect of aliasing from the down-sampled signal?

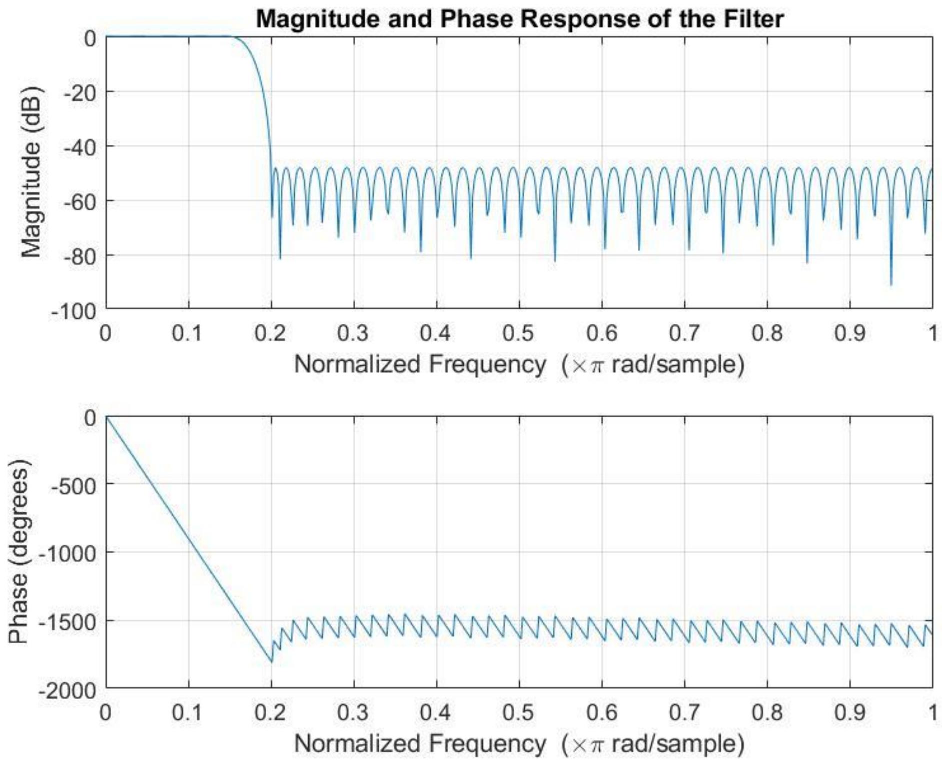
No, the reconstruction process cannot undo the effect of aliasing from the down-sampled signal.

Section 2:

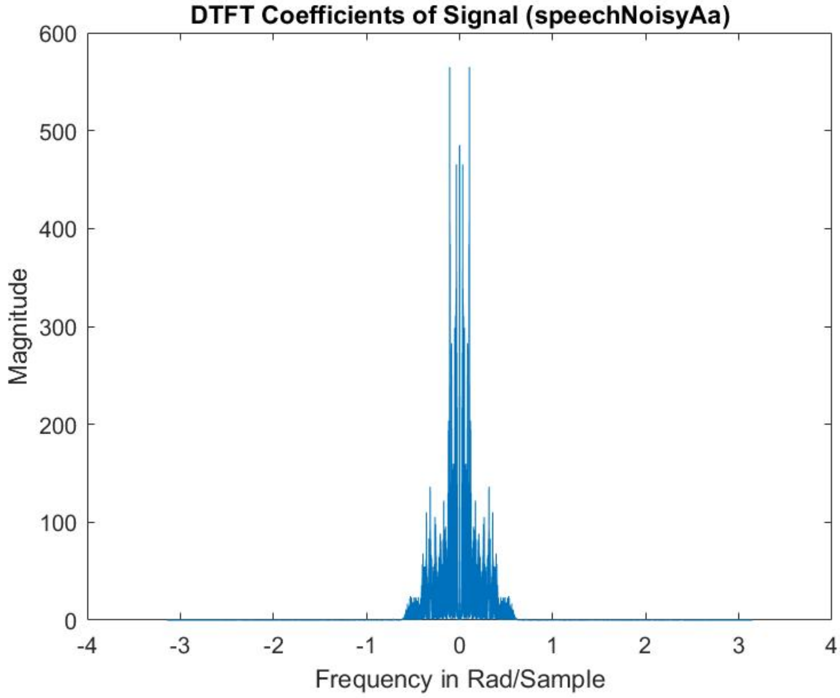
Question 1: Given that we are about to downsample speechNoisy by a factor of 5, determine an appropriate choice of cut-off frequency (in rad/sample) for an anti-aliasing FIR filter.

The appropriate choice for the cut-off frequency would be $\pi/5$ which equates to about 0.628 rad/sample (radian per sample).

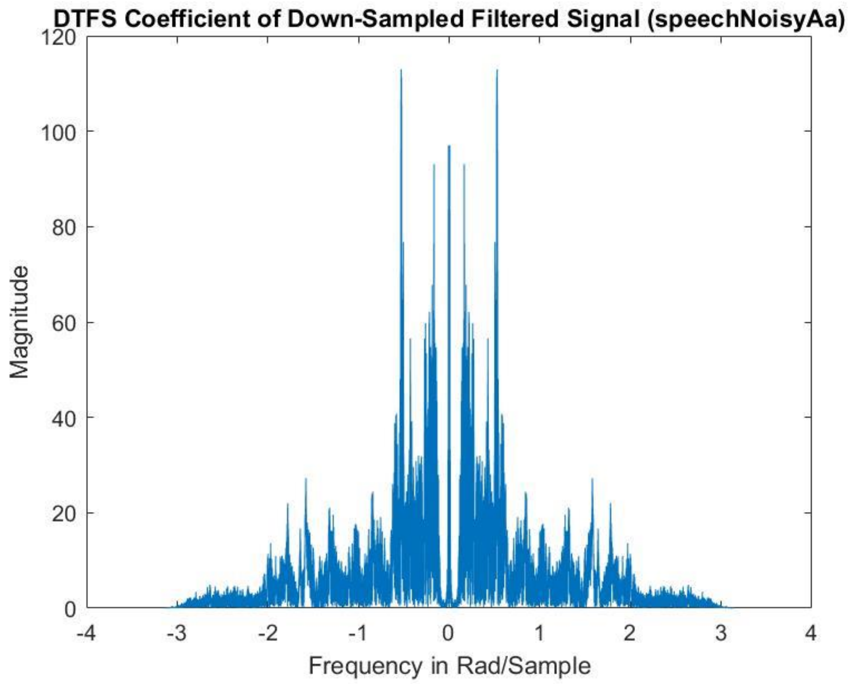
Question 2: Use `freqz(h,1)` *to plot the magnitude and phase response of the filter you have designed and make sure it behaves as you want it to. Include your labeled plot below.*



Question 4: Plot `speechNoisyAa` in the frequency domain (magnitude only) with frequency axis in rad/sample. Use the second type of plot discussed in lab 4. Include your labeled plot below.



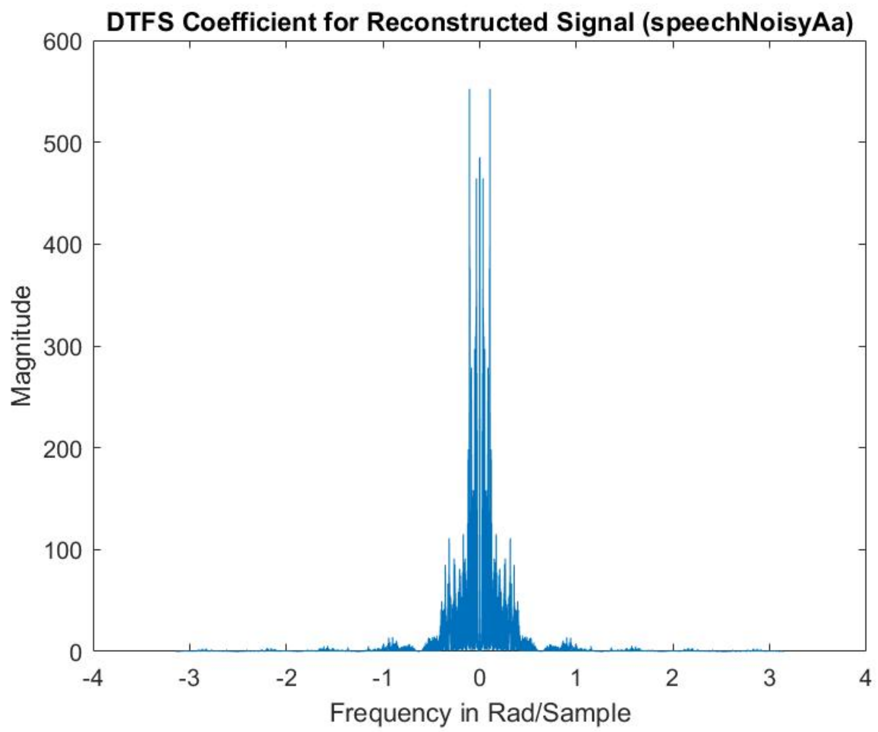
Question 6: Play the down-sampled signal and plot it in the frequency domain (magnitude only) with frequency axis in rad/sample. Use the second type of plot discussed in lab 4. Include your labeled plot below.



Question 7a: How does the reconstructed signal compare with the original signal speech and with the reconstructed signal without anti-aliasing from section 1?

The reconstructed signal compared with the original signal speech sounds slightly distorted and muffled. The reconstructed signal compared with the reconstructed signal without anti-aliasing is that there is a removal of noise due to filtering done by the low-pass filter.

Question 7b: Plot the reconstructed signal in the frequency domain (magnitude only) with frequency axis in rad/sample. Use the second type of plot discussed in lab 4. Include your labeled plot below.



Code for section 1:

Paste your script in here.

```
% Written by Tan Jin Chun
% Last Modified : 17/9/2021
% Lab05T01

clear all;close all;clc

%% Question 1
% Question 1
% Loading in the signal speech into the workspace
load lab5data.mat;

%% Question 2
% The sampling rate
Fs = 44100;

% The length of the signal
N = length(speech);
n = 0:N-1;

% Finding the average power of the signal speech
P = (speech' * speech)/N;

% Transposing speech
speech = speech';

% Creating a new signal called speechNoisy
% Getting the value of A
A = sqrt(0.2 * P);

%% Question 3
% Noise Signal
Hz = 5000;
speechNoisy = speech + A*cos(2 * pi * Hz * n / Fs);
% speechNoisy = conv(speech, A*cos(2 * pi * Hz / Fs));

%% Question 4
% Playing the signal speechNoisy using soundsc
% soundsc(speech);
% pause
% soundsc(speechNoisy);
% pause

% Plotting the noisy signal in the frequency domain (magnitude only)
Nnoisy = length(speechNoisy);
Xnoisy = fft(speechNoisy);
omega = (-floor(Nnoisy/2):(Nnoisy-1-floor(Nnoisy/2)))*(2*pi/Nnoisy);
figure;
plot(omega, fftshift(abs(Xnoisy)));
xlabel("Frequency in Rad/Sample");
ylabel("Magnitude");
title("DTFT of Noisy Speech");
```

```
%% Question 5
% Downsampling the signal
% Factor of 5
L = 5;
xds = speechNoisy(1:L:end);
N = length(xds)*L;

% Playing the downsampled signal
% soundsc(xds);
% pause;

%% Question 6
% Reconstructing the original signal
xrecon = interp1(1:L:N,xds,1:N,"linear",0);

%% Question 7
% Playing the reconstructed signal using soundsc
% soundsc(xrecon);
% pause;

% Plotting the reconstructed signal
Nrecon = length(xrecon);
Xrecon = fft(xrecon);
omega = (-floor(Nrecon/2):(Nrecon-1-floor(Nrecon/2))) *
(2*pi/Nrecon);
figure;
plot(omega, fftshift(abs(Xrecon)));
xlabel("Frequency in Rad/Sample");
ylabel("Magnitude");
title("Magnitude of DTFS Coefficient for Reconstructed Signal
(speechNoisy)");
```

Code for section 2

Paste your script in here.

```
% Written by Tan Jin Chun
% Last Modified : 17/9/2021
% Lab05T02

clear all;close all;clc

%% Question 1
% Loading in the signal
load lab5data.mat;

% Reconstructing back the speechNoisy signal from the previous
section
% Getting the length of the signal
N = length(speech);
n = 0:N-1;
Fs = 44100;

% Finding the average power of the signal speech
P = (speech' * speech)/N;

% Transposing speech
speech = speech';

% Creating a new signal called speechNoisy
% Getting the value of A
A = sqrt(0.2 * P);
Hz = 5000;
speechNoisy = speech + A*cos(2 * pi * Hz * n / Fs);

%% Question 2
% Constructing the Anti-Aliasing Filter (Low-Pass Filter)
Cutoff_Freq = [0 0.15 0.2 1];

% Using the firpm function to create the filter
h = firpm(100, Cutoff_Freq, [1,1,0,0]);

% Plotting the magnitude and the phase response of the filter using
freqz
% function
figure;
freqz(h,1);
title("Magnitude and Phase Response of the Filter");

%% Question 3
% Filter the signal by convolving the signal and the filter
speechNoisyAa = conv(speechNoisy, h);
Na = length(speechNoisyAa);

%% Question 4
% Plotting the speechNoisyAa in the frequency domain (Magnitude
Only)
Xnoisya = fft(speechNoisyAa);
omegaAa = (-floor(Na/2):(Na-1-floor(Na/2)))*(2*pi/Na);
```

```
figure;
plot(omegaAa, fftshift(abs(Xnoisya)));
xlabel("Frequency in Rad/Sample");
ylabel("Magnitude");
title("DTFT of Filtered Noisy Speech");

%% Question 5
% Downsampling the signal by a factor of 5
L = 5;
xdsAa = speechNoisyAa(1:L:end);
NA_Downsampled = length(xdsAa);

%% Question 6
% Playing the downsampled signal
% soundsc(xdsAa);
% pause;
% Plotting the filtered noisy signal in the frequency domain
XnoisyDa = fft(xdsAa);
omegaDAa = (-floor(NA_Downsampled/2):(NA_Downsampled-1-
floor(NA_Downsampled/2)))*(2*pi/NA_Downsampled);
figure;
plot(omegaDAa, fftshift(abs(XnoisyDa)));
xlabel("Frequency in Rad/Sample");
ylabel("Magnitude");
title("DTFT of Down-Sampled Filtered Noisy Speech");

%% Question 7
% Reconstructing the original signal by linear interpolation using
interp1
% Reconstructing the original signal
xrecon = interp1(1:L:Na,xdsAa,1:Na,"linear",0);

% Playing the reconstructed signal using soundsc
% soundsc(xrecon);
% pause;

% Plotting the reconstructed signal in the frequency domain
(magnitude only)
Nrecon = length(xrecon);
Xrecon = fft(xrecon);
omega = (-floor(Nrecon/2):(Nrecon-1-floor(Nrecon/2))) *
(2*pi/Nrecon);
figure;
plot(omega, fftshift(abs(Xrecon)));
xlabel("Frequency in Rad/Sample");
ylabel("Magnitude");
title("DTFS Coefficient for Reconstructed Signal (speechNoisy)");
```