Experiment 3

Quantization and Phase Shift

1. Purpose

The main purpose of this experiment is to study the quantization of discrete-time signals. Two basic quantizer types are compared with each other. Signal level mismatch and effect of increasing the number of bits of quantizer is investigated. Example for adaptive quantization appraoch is presented for speech signal. Finally, linear and nonlinear phase filters are investigated.

2. Lab Work
   - (a) Generate a discrete-time sinuosidal signal x[n] with the gen function implemented in Experiment 1, with the parameters: $A = 3, f = 1Hz, T = 2sec, F_s = 50Hz$
     - i. Implement $3-bit$ midrise type quantizer. Make the reconstruction levels be spaced so as to span the entire amplitude range of the signal. You may use the maximum amplitude of the signal in designing reconstruction levels. Plot original signal, quantized version and quantization error. Calculate output signal to noise ratio.
     - ii. Implement $3-bit$ midtread type quantizer (This time quantizer will be asymmetric). Plot original signal, quantized version and error. Calculate output signal to noise ratio.
   - (b) In order to see the effect of signal level mismatch in your desgined midrise type quantizer, obtain the signals
     - i. $x_1[n] = 0.5\,x[n]$
     - ii. $x_2[n] = 2\,x[n]$

   but do not modify the quantizer. Calculate output signal to noise ratios. Comment on the results.
   - (c) Load the sound.wav. Encode the data using $3-bit$ uniform midrise type quantizer (again, space reconstruction levels so as to span the entire amplitude range of the signal). Listen the original and encoded versions. Calculate the output signal to noise ratio and plot the error.
   - (d) Do the procedure in question (c) with $4-bit$ uniform midrise type quantizer. Comment on the results by comparing with question (c).
   - (e) Better approach in speech quantization is segmentation. Partition the utternce, into segments of 100 samples. Now, quantize these segments individually as in question (a)(i) (place reconstruction levels in each segment considering the maximum amplitude range in corresponding segment). Listen the original and encoded version. Calculate the output signal to noise ratio and plot the error. Cooment on the results by comparing with question (c)
   - (f) Generate a discrete-time signal $x[n]$ which is sum of three sinusoids using the gen function implemented in preliminary work of experiment 1, with the following parameters

$$A_1 = 15, f_1 = 4Hz, T_1 = 2sec, F_{s_1} = 50Hz$$
$$A_2 = 5, f_2 = 12Hz, T_2 = 2sec, F_{s_2} = 50Hz$$
$$A_3 = 3, f_3 = 20Hz, T_3 = 2sec, F_{s_3} = 50Hz$$

Filter the signal $x[n]$ with:

i. $|H(e^{jw})| = 1$ and $\measuredangle H(e^{jw}) = -w$

ii. $|H(e^{jw})| = 1$ and $\measuredangle H(e^{jw}) = -w^2$

See the phase spectra of original signal and filtered ones. Plot the original signal and the filtered ones on the same page (You may use MATLAB's subplot() function). Comment on the results

(g) Load the file sound1.wav by using wavread() function of MATLAB. Filter the signal with:

i. $|H(e^{jw})| = 1$ and $\measuredangle H(e^{jw}) = -w$

ii. $|H(e^{jw})| = 1$ and $\measuredangle H\left(e^{jw}\right) = -w^2$

See the phase spectra of the original and filtered ones. Plot the original signal and the filtered ones on the same page by using subplot(). Listen filtered signals and comment on the results.