Digital Signal Processing - Lab 4
Signal Generation and Compression

1 Generating signals

Assignment 1 Program the DSP as a signal generator. Generate a 5 kHz sine frequency with a sampling rate set to 8 kHz. Rather than using the C *sin* function, use a table containing the samples of your sine signal and loop over it. This technique is more efficient than calling the *sin* function for each input sample. Analyze the produced signal. What do you observe ? Interpret and explain the observed result.

Assignment 2 Program the DSP to generate a *sweep* signal, that is a signal that swipes all the frequencies between a *freq_ start* and a *freq_ stop* frequency. To create a sweep signal, use a table containing a single period sine signal of 8000 points. Generate that sine signal using Matlab with the following function :

$$x = 1000 \times sin(2\pi \times [0 : 7999]/8000)$$

Next, use the table to generate all the frequencies between the start and stop frequencies. That kind of signal generation method is used in many real lige signal generators.

2 Reducing the sampling rate

Assignment 3 Write a program that reduces the sampling rate of the signal by only keeping one sample over six, while keeping the DSP sampling rate set to 48 kHz. Discard the remaining incoming samples. Test your program on a pure 1 kHz sine frequency. Would do you hear ? Analyze the spectrum on the analyzer. What do you observe ? Interpret the observations and repeat the experiment with 2 kHz and 4 kHz frequencies.

Assignment 4 Modify the sampling rate of the DSP so that it samples the input at 8 kHz rather than 48 kHz. Repeat the previous experiment. What do you observe ? Interpret.

3 Signal compression

Assignment 5 The goal of this experiment is to reduce the information stored in a speech signal sampled at 44.1 kHz and quantized at 16 bits. Reduce progressively the sampling rate and listen to the speech signal. According to you, what is the minimum sampling frequency that can be achieved without significantly changing the quality of the speech signal ?

Assignment 6 Reset the DSP to a 48 kHz sampling rate. This time, modify the quantization of the signal by applying binary masks to reduce the number of bits used to store the samples. What is the lowest quantization level that you can achieve without significantly changing the the quality of the speech signal ?