Elec Eng 4TL4
Lab 2 – Resampling, Reconstruction, and Convolution

Instructor: Dr. R. Thamarasa (Thamas)
George Gill – L03 Group 21 –Gillg62- 400327563
Richa Rani – L03 Group 21 –Ranir- 400311161
TA: Zhilli

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As a future member of the engineering profession, the student is responsible for performing the required work in an honest manner, without plagiarism and cheating. Submitting this work with my name and student number is a statement and understanding that this work is my own and adheres to the Academic Integrity Policy of McMaster University and the Code of Conduct of the Professional Engineers of Ontario. Submitted by [George Gill, Gillg62, 400327563]

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Introduction

a.) Create the discrete-time sequence:
$x[n] = u[n] - u[n-10]$
where u[n] is the unit-step function (i.e., $u[n] = 1$ for $n \geq 0$ and $u[n] = 0$ for $n < 0$). You do not have to zero-pad x[n] (i.e., the vector you have should contain no zero elements).
Plot x[n] using stem() function

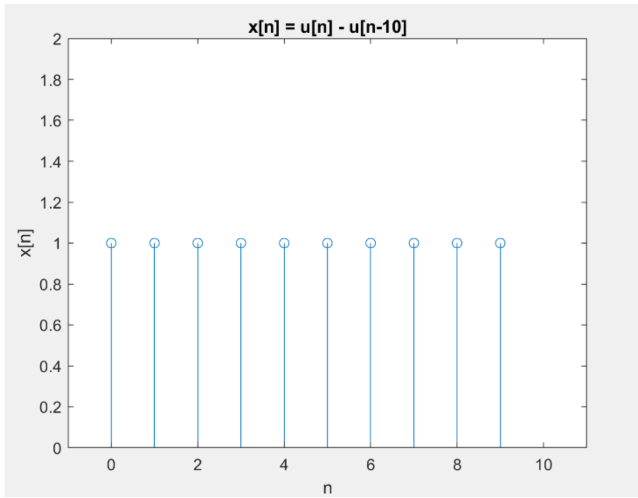


Figure 1

b.) Now, convolve x[n] over and over:
$a[n] = x[n] * x[n]$
$b[n] = a[n] * x[n]$
$c[n] = b[n] * x[n]$
$d[n] = c[n] * x[n]$
You can use the MATLAB function conv() with proper input parameters.

c.) Plot a[n], b[n], c[n], d[n] using stem() function.

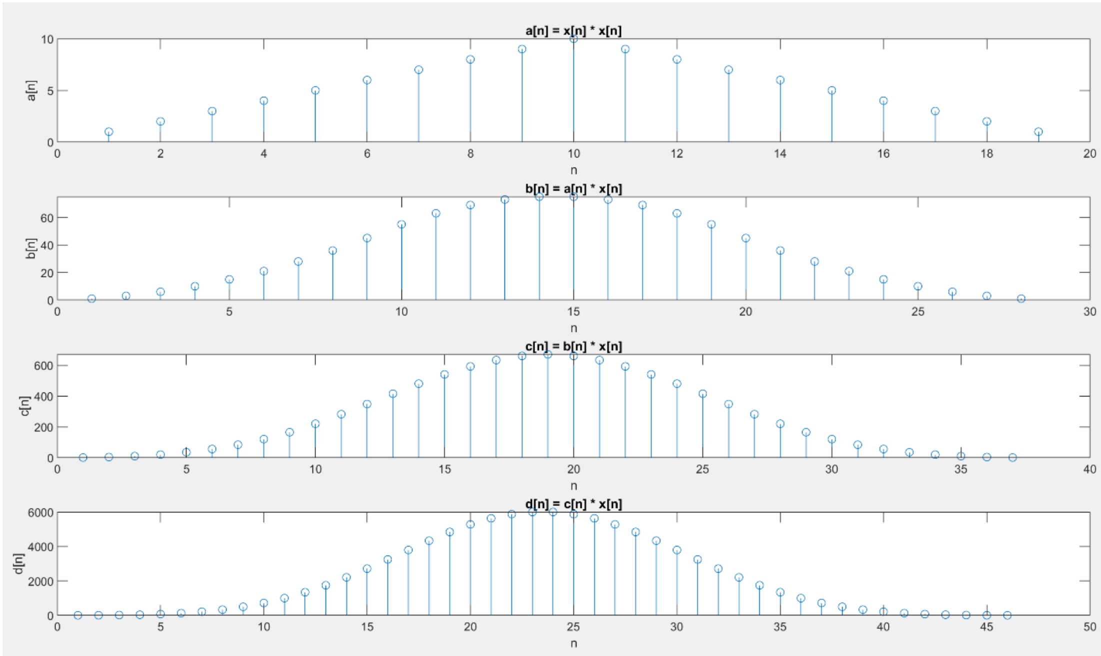


Figure 2

Convolution of Signals and System Impulse Responses

a.) Load the supplied acoustic impulse response of a room into MATLAB using the command:
[impr,fs] = audioread('roomIR.wav');
This impulse response was obtained by creating an impulsive noise at one position in the room and recording (and digitizing) the sounds arriving at another position in the room

b.) Plot the impulse-response waveform impr using the plot() command and listen to it using the soundsc() command. What can you see and hear in the impulse response?

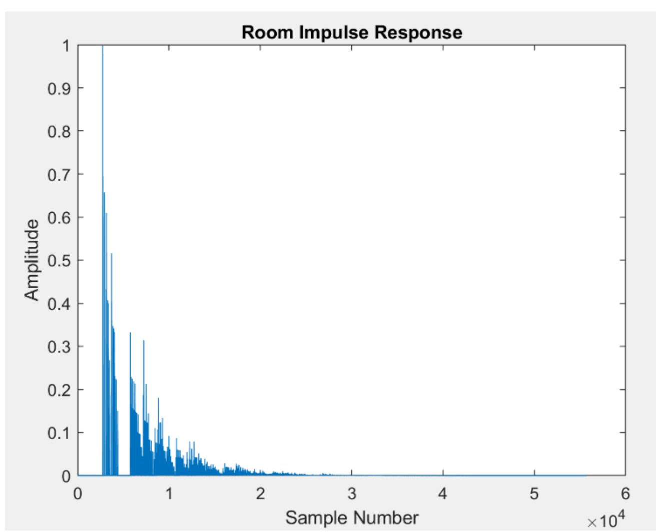
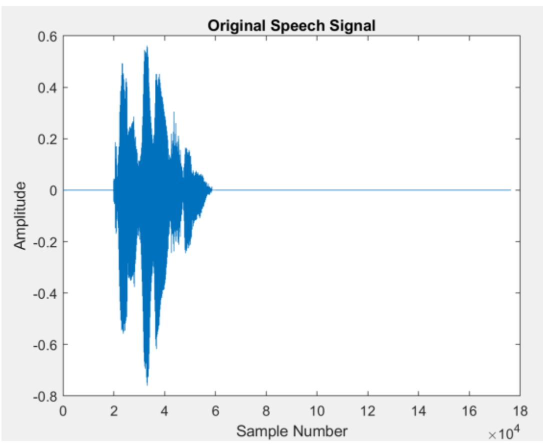


Figure 3

The sound heard in the impulse response of the roomIR.wav file was similar to the sound of an object dropping in a room. After the peak of the impulse response at the beginning, there is a gradual decay of the sound instead of an immediate drop. This represents the signal being reflected and absorbed by the walls in a room.

c.) Load the supplied speech signal into MATLAB using the command:
[y,fs] = audioread('convolution.wav');

d.) Convolve the speech signal with the impulse response, and plot and listen to the resulting signal. Describe what you see and hear, comparing it to the original speech signal y. Explain what the convolved signal is physically equivalent to, according to the impulse-response theory



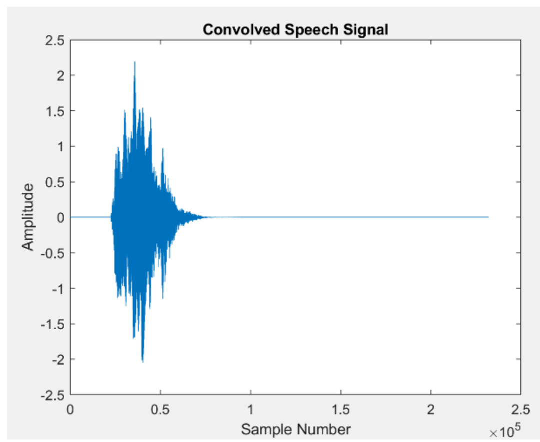


Figure 4

When comparing the original and convolved speech signal, they are quite different. In terms of sound, while the original signal had a clear sound, the convolved signal sounded like it was in a room where the signals were bouncing off the walls, much like the impulse signal. You can see the effect of the impulse signal in the convolution because the number of samples in the convolved signal is much more than the number of samples in the original signal. This is because this includes the gradual decay after the impulse signal from the impulse signal it was convolved with.

Image Resampling and Reconstruction

a.) Import KillarneyPic.png, and report its size and bytes.

b.) Most MATLAB functions require arrays to be of the double class. Convert the imported picture into a double-class variable using im2double() and display it in a MATLAB figure using the imshow() function

c.) ou will be using this image to explore different resampling and reconstruction effects. Produce new images by resampling, and in some cases reconstructing, the original as follows i.) Impulse sampling at 1/5 of the original rate (i.e., set 4 out of every 5 samples to zero)

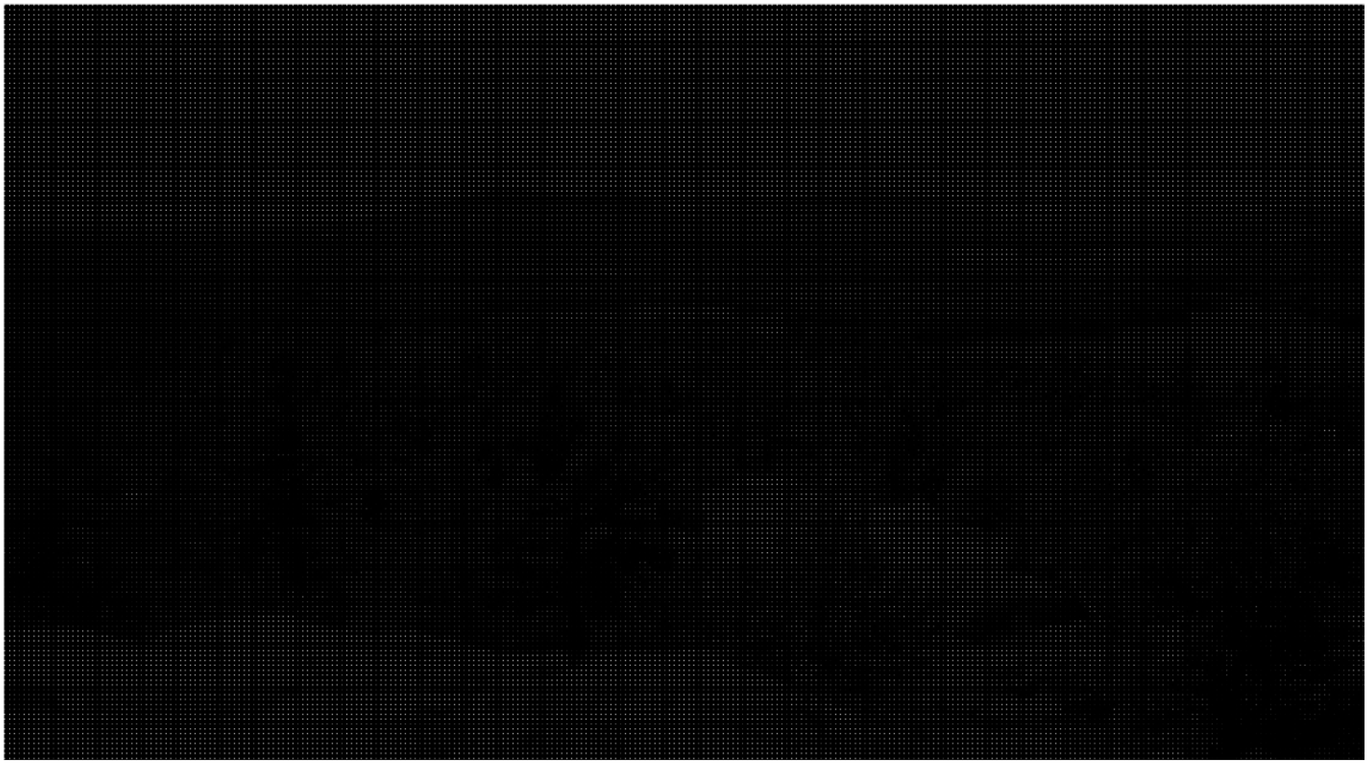


Figure 5

ii.) Downsampling by a factor of 5 (i.e., discard 4 out of every 5 samples)

Figure 6

iii.) Zero-order hold reconstruction (back to the original rate) from the downsampled image created in part (ii) above

Figure 7

iv.) First-order hold reconstruction (linear interpolation—back to the original rate) from the downsampled image created in part (ii) above

Figure 8

d.) Of the 4 resampled/reconstructed images plotted in gray-scale, which one most closely resembles the original picture? Rank the images in order of FIDELITY compared to the original.

1. Most Similar: First Order Hold. This is because the interpolation method preserves the most amount of detail. It does not set the values in between the bits as zero but estimates the values based on the previous and subsequent values and creating a smoother gradient
2. Zero-Hold reconstruction: This method duplicates the previous value all the way to the next value which creates less of a smooth gradient compared to the previous method.
3. Down sampled image: This discards 4 out 5 values creating a large loss in data
4. Impulse Sampling: This is the least similar to the picture because it only retains one piece of data out of 5 and also replaces the other 4 pixels with zero which is just black, making the picture the least similar.

FMCW Radar Signal Processing Project with TMS320C6713 DSK

iv.) Observe the signals on the oscilloscope, and explain why the waveform of the transmitted signal is different from the waveform in Figure 1.

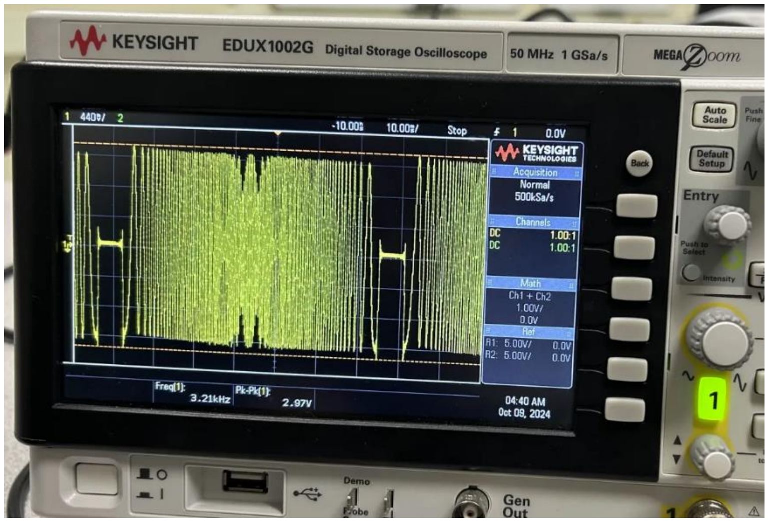


Figure 9

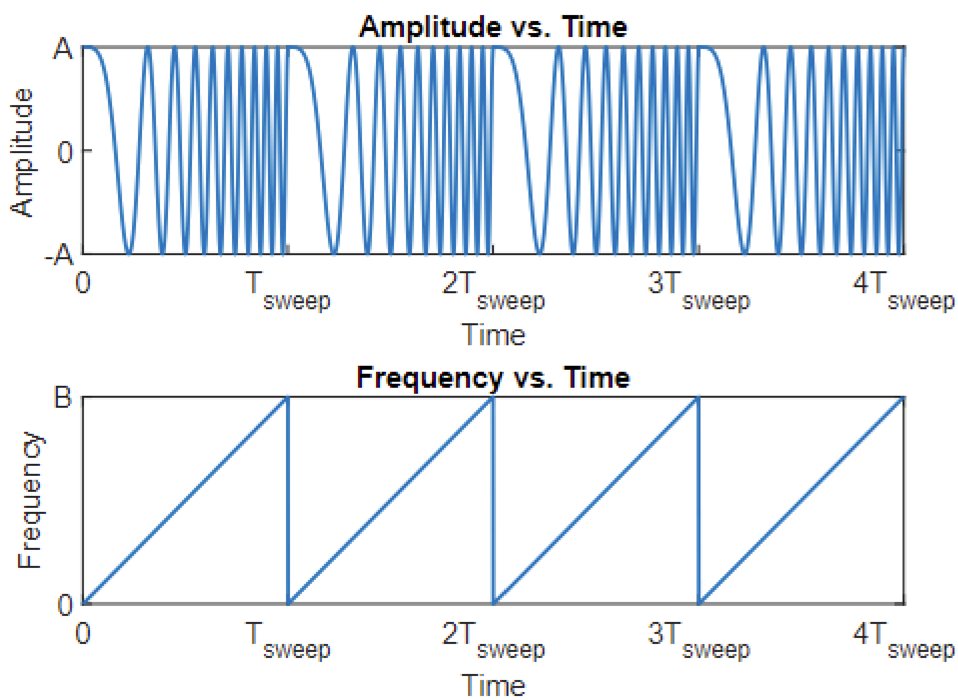


Figure 10

The figure above plots the transmitted signal via the DSP board. This was observed using an oscilloscope. Visually examining the signal, we see it differs from Figure 10 present in the laboratory manual. Firstly, the most distinct difference is the zero-padding present in the transmitted signal. This is seen occur twice in

Figure 9, as the signal inserts zero padding after a certain duration. The point of using zero padding in this application is to increase the size transmitted signal. Furthermore, another clear difference between the two figures is the frequency. In Figure 9, the transmitted signal has a much higher frequency causing a lot more oscillations as compared to the signal in Figure 10. This simply due to the sampling frequency used in Figure 9 is greater than the sampling frequency used in Figure 10.

v.) Down sample the transmitted signal with a factor of 2 by modifying the interrupt function. Describe what you observed on the oscilloscope and explain.

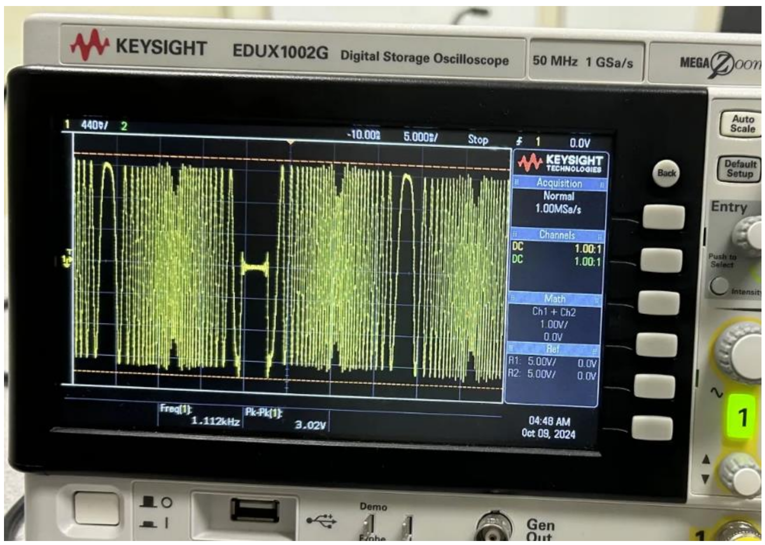
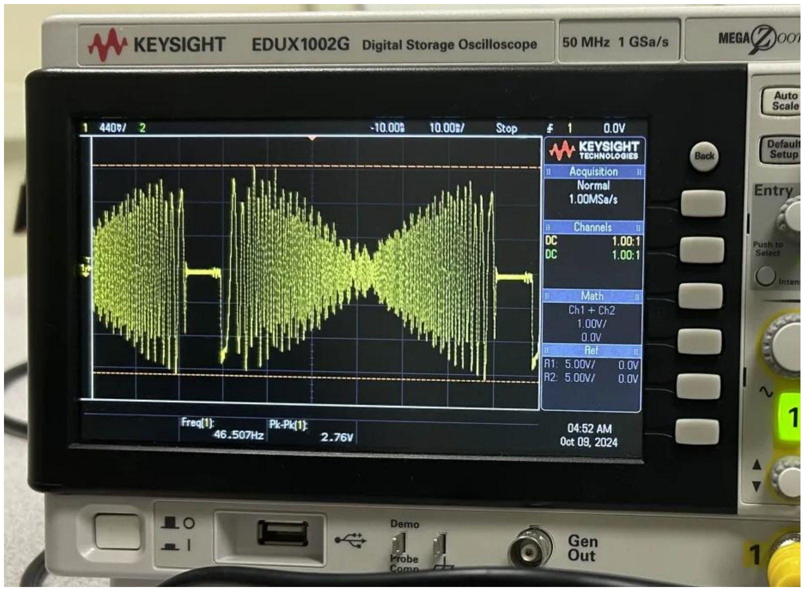


Figure 11

Next, we modified the interrupt function to reduce the number of samples, in other words the sample frequency was decreased as Fs = N/T where N is number of samples. This results in the signal which is shown in Figure 11. By decreasing the number of samples, we see this signal shows similarities to the signal in Figure 10. For example, both signals clearly have a segment where the frequency is much lower than the rest of the signal. However, this same segment was not present in Figure 9 as the frequency was doubled.

vi.) Modify the interrupt function to plot the received signal (without downsampling) instead of the transmitted signal. Observe the received signal and see if you can determine the number of targets in the coverage. You have to look for peaks to find the targets



Next, the interrupt function was modified to instead plot the received signal. This signal looks much different than the previous signals, as the shape represents a bow ties shape however the properties of the frequency modulation are still present as the frequency can be seen to lower at both ends of the signal, and greater at the middle. By counting the number of peaks of this signal, we can estimate the number of targets present as 2.