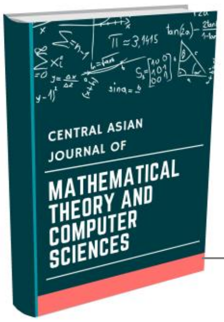

CENTRAL ASIAN JOURNAL OF MATHEMATICAL THEORY AND COMPUTER SCIENCES

http://cajmtcs.centralasianstudies.org/index.php/CAJMTCS

Volume: 03 Issue: 06 | Jun 2022 ISSN: 2660-5309

Assessment of Signal Distortion Removal in the Noisy Environments

Parveen Kumari
Student, Sat Kabir Institute of Technology and Management, Haryana, India,
dahiyaparveen22@gmail.com

Shalini Bhadola, Kirti Bhatia
Assistant Professor, Sat Kabir Institute of Technology and Management, Haryana, India,
shalini77info@gmail.com, bhatia.kirti.it@gmail.com

Rohini Sharma
Assistant Professor and corresponding Author, GPGCW, Rohtak, India
rohinisharmaohlan@gmail.com

Annotation:
The study of digital signal processing (DSP) has become increasingly significant over the past few decades, both academically and operationally. The creation and application of affordable software and hardware is a key factor in its commercial success. DSP algorithms are now being used in new technologies and applications across many industries. Electronics and communication engineers with experience in DSP will be in more demand as a result of this. In this article, we have sampled various analog signals with different frequencies. We have study Uniform Quantization, A-Law and Mu-Law Transformation based quantization as well as Quantization after adding noise. We have applied different filtering techniques to remove noise. We have used Wiener filter for noise cancellation. A bandpass signal with a bandwidth of 25 KHz is input. It is enough to sample bandpass signals at a rate twice the bandwidth of the signal in order to enable accurate reconstruction. We discovered that depending on the sampling rate, we either get a positive or a negative (flipped) representation of the original spectrum.

ARTICLEINFO

Article history:
Received 18 Apr 2022
Revised form 15 May 2022
Accepted 30 Jun 2022

Key words: Signal, Sampling, Quantization and Filtering.

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INTRODUCTION:

In order to improve and magnify the sound that the user hears when using the device, signal processing is essential for recording and processing ambient sound. With the least amount of delay feasible, the sound is changed from analogue to digital in this process, then returned to analogue before reaching the ear.

Prior to performing operations like noise cancellation, extracting crucial and necessary elements from the signal, or packaging their information, the signal is first verified in several areas, including time, frequency, etc. After the pre-processing procedure on the signal processing procedures, the technique is prepared for its characteristics to be put into categorization and detection techniques for detection and prediction. Consequently, any noises in the sound signals are removed to improve the telecommunication network' efficiency [1].

The microphones, microprocessor, receiver, and power source are the four coordinated components at the heart of auditory rehabilitation technology. Typically, in DSP, error in amplitude is not "created" through the conceptual sampling procedure, hence time-discrete signal models are used for theoretical analyses and inferences. Quantized signals, like those produced by an ADC, are necessary for numerical approaches. A spectrum of frequency or a group of statistic gages could be the processed outcome. However, a digital-to-analog converter transforms it into another quantized signal (DAC). Fig 1 shows a signal processing workflow.

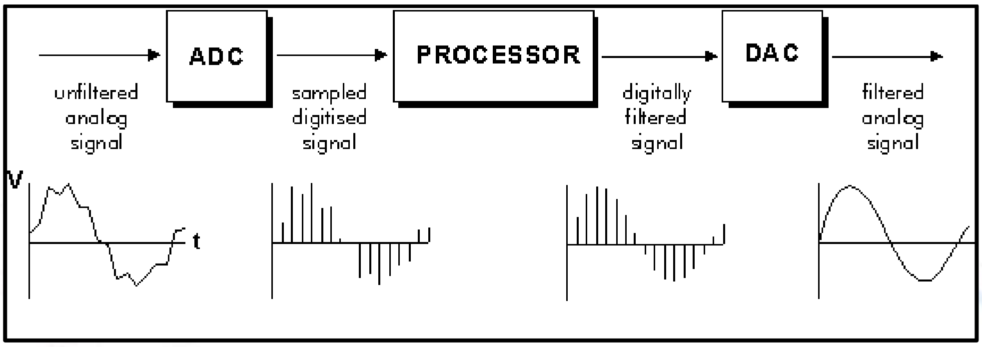

Fig 1: Signal Processing Workflow

An analogue-to-digita converter must digitalize the analogue (continuous) signal before it can be analysed and altered digitally. Typically, sampling is accomplished in 2 steps: quantification and discretization which is the method of splitting a signal into identical temporal segments and assigning a measured amplitude to each segment [4].

SAMPLING

Sampling is the conversion of a continuous-time signal into a discrete-time signal in signal processing. The transformation of a continuous signal into a series of samples is a typical example. A quantity or combination of values at a particular point in time or location constitute a sample. A sampler is a component or process that takes discrete samples out of a continuous signal. A theoretically perfect sampler creates samples at the specified points that are equal to the instantaneous value of the continuous signal, Fig 1 [2].

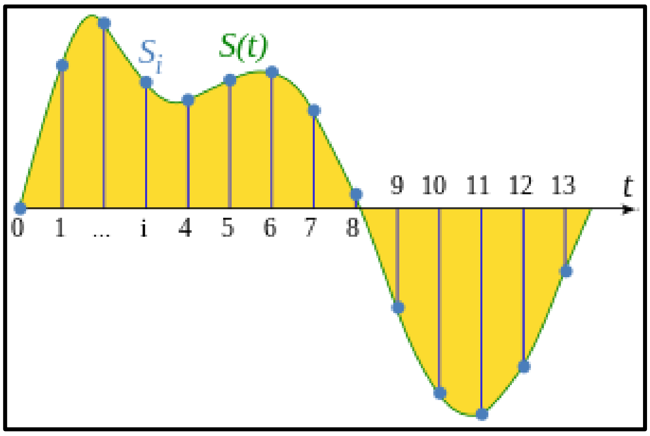

Fig1: Sampling of the Signal

The temporal quality of the continuous signal is replicated in samples at the designated points using a mathematically ideal sampler. The received data can be reproduced from a stream of samples up to the Nyquist limit by passing the stream of samples via a low-pass filter termed a reconstructions filter [5-7].

QUANTIZATION

Since it is carried out on the y as opposed to the x axis, quantization is the reverse of sampling. A sampled image with actual values is converted to one with just a limited set of distinct values by the process of quantization. The image's amplitude values are digitalized during the quantization process. To put it simply, quantizing an image involves splitting a signal into quanta (partitions) [8].

Let's now examine the process of quantization. Here, we categorise the values produced by the sampling process into levels. Despite having been taken, the samples in the image from the sampling explanation were still stretching vertically to a continuous range of grey level values. These vertically varying values have been quantized into 5 distinct levels or partitions in the graphic below. varying between 0 black and 4 white. Depending on the kind of image you want, this level might change [3]. Fig 2 so a scientific process of signal processing. Fig 3 shows conversion of analog signals in to samples and how quantization process converts them into digital signal.

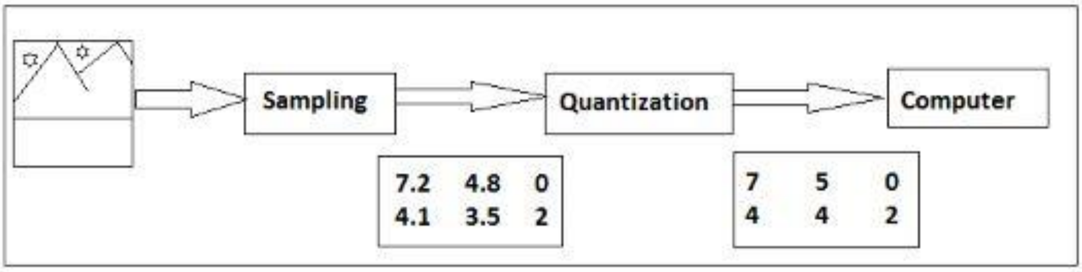

Fig 2: Scientific process of signal processing

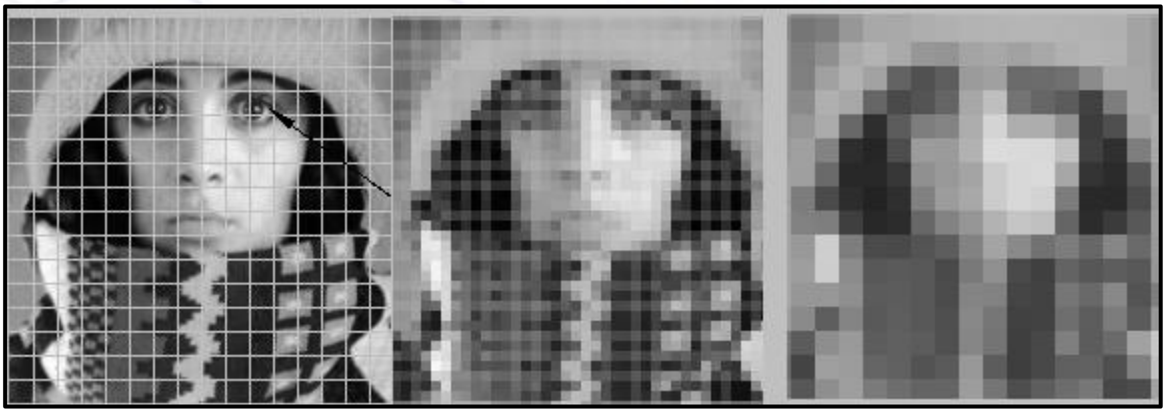

Fig 3: Quantization Process

DISCRETE FOURIER TRANSFORM (DFT)

The (DFT), as its name suggests, transforms discrete-time data sets into discrete-frequency representations. This contrasts with the DTFT, which transforms discrete time to continuous frequency. Computers frequently perform DFT designs when frequency info is required since the generated frequency information is discrete. There is an approach for computing the DFT that is extremely quick on current computers by using a number of mathematical gimmicks and generalisations. This approach, also referred to as the (FFT), takes a far shorter amount of time to compute than standard DFT calculations while still producing the same results.

FILTERING

The very first 3 elements of a concatenation of cosines are used to create the taper known as the Blackman window. It was made with hardly no leaking at all in mind. It is only marginally less ideal than a Kaiser window. An image is processed in the frequency domain via frequency filters. The image is first transformed using Fourier analysis, then multiplied by the filter function, and finally turned once more into the spatial domain (SD). While attenuating low frequencies improves the edges, attenuating high frequencies produces a smoother image in the spatial domain. All filters are also implementable in the spatial domain, and provided the desired filter effect has a straightforward kernel, filtering in the spatial domain is computationally more efficient. If there isn't a simple kernel in the SD, frequency filtering is more proper and might be more effective. A common method for reducing noise in a picture or wave is the median filter, a non-linear digital straining method. A typical pre-processing step to improve the results of future processing is noise reduction [9-14].

WIENER FILTER

By applying LTI straining to an experimental noisy procedure, the Wiener filter can be applied to estimation of an anticipated or target random procedure. Additive noise and established static signal and noise spectra are assumed. Between the intended process and the estimated random process, the Wiener filter reduces the MS error [16].

PROPOSED METHOD

Stage 1: Initially, we have determined the DFT of the signal with different frequencies. Draw the spectra with Fs = 12KHz and note the variation with N after entering an analogue signal waveform to determine the DFT of the waveform for N = 12, 64, 128, 256. Results of the sampling is shown in Fig 1.

Detect aliasing by repeating the technique above with other sampling frequencies, such as Fs=4 KHz, 5 KHz, and 8 KHz. Results in Fig 2,3, 4 and 5.

The Fast Fourier Transform of the signal was computed using the built-in function and the normalised DFT of the provided analogue waveform was shown. The signal's spectrum was made symmetrical about the origin. The difference between the theoretical magnitude and the measured spectrum is shown to decrease as the number of sample points increases.

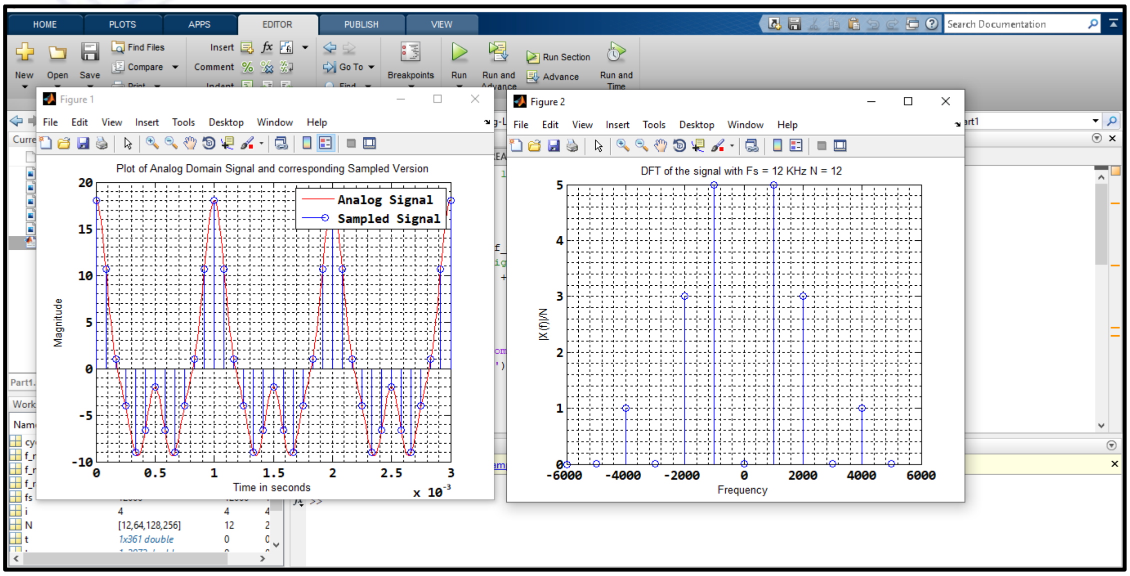

Fig 1: Sampling of original Analog signal and its Sampled Signal

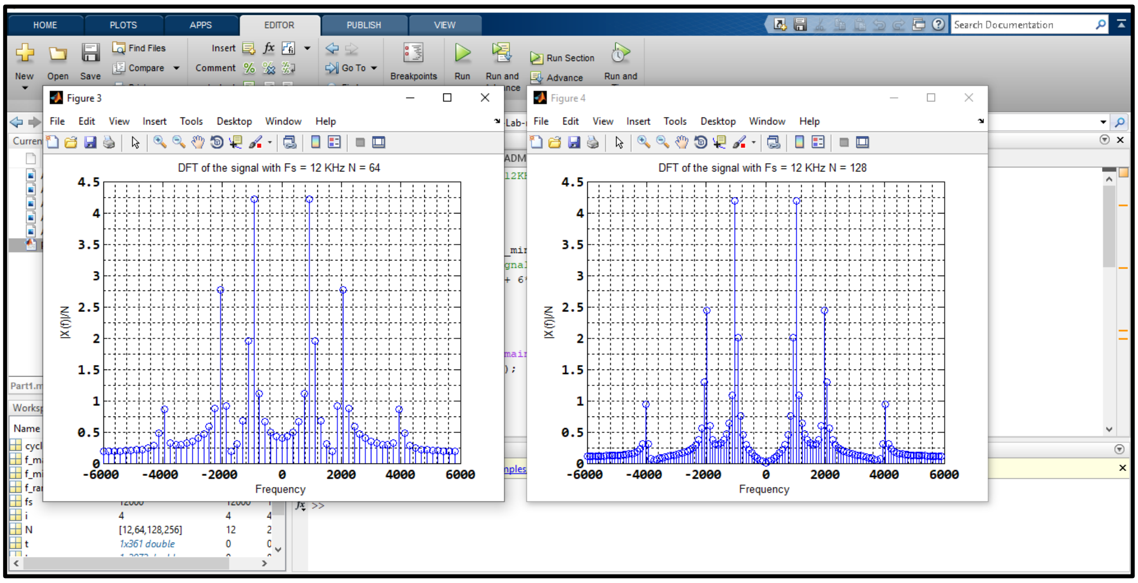

Fig 2: Sampling of Analog signal of different Frequencies

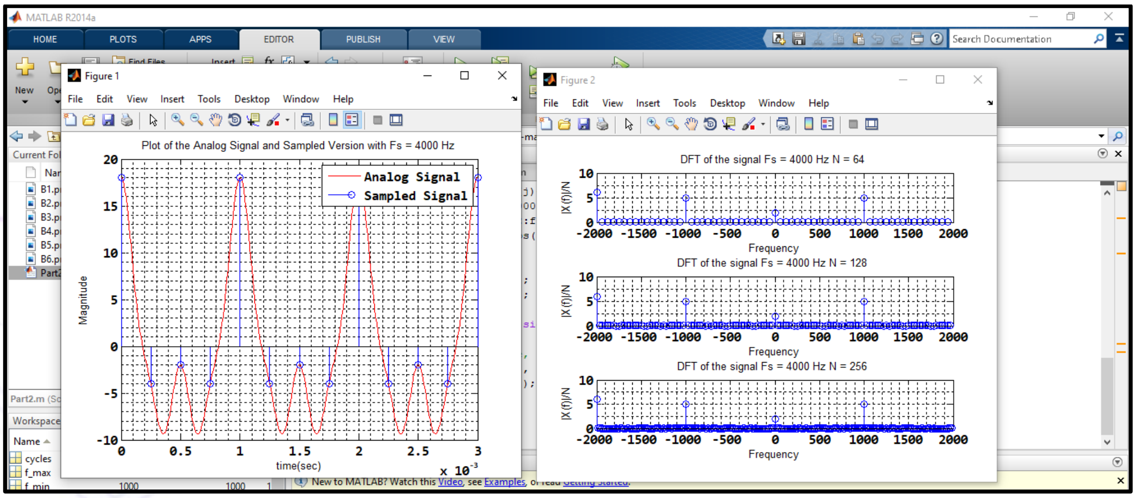

Fig 3: Sampling of original Analog signal and its Sampled Signal for frequency 4K

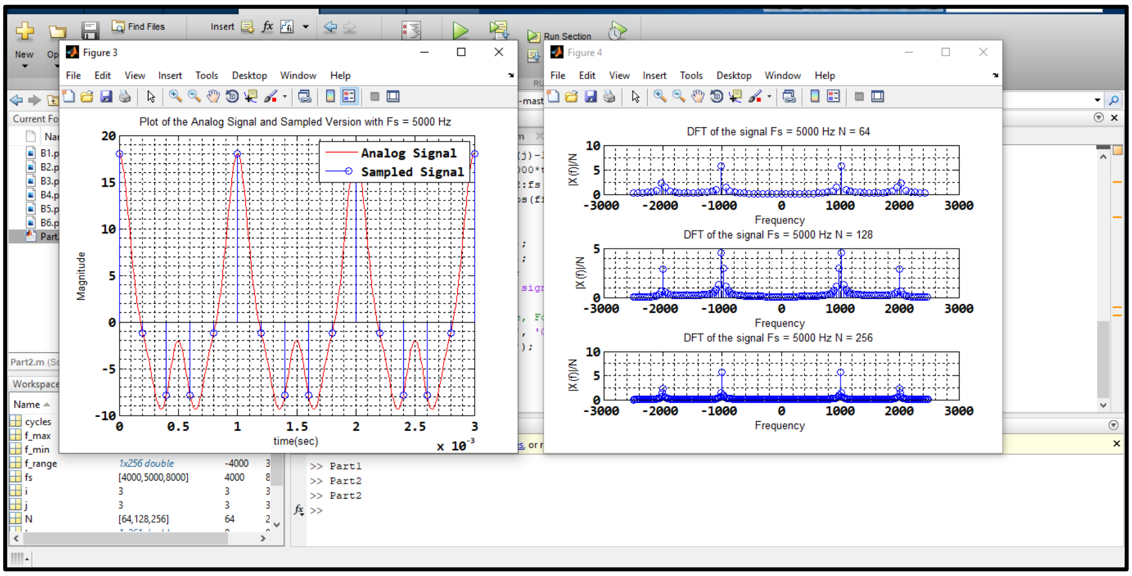

Fig 4: Sampling of original Analog signal and its Sampled Signal for frequency 5K

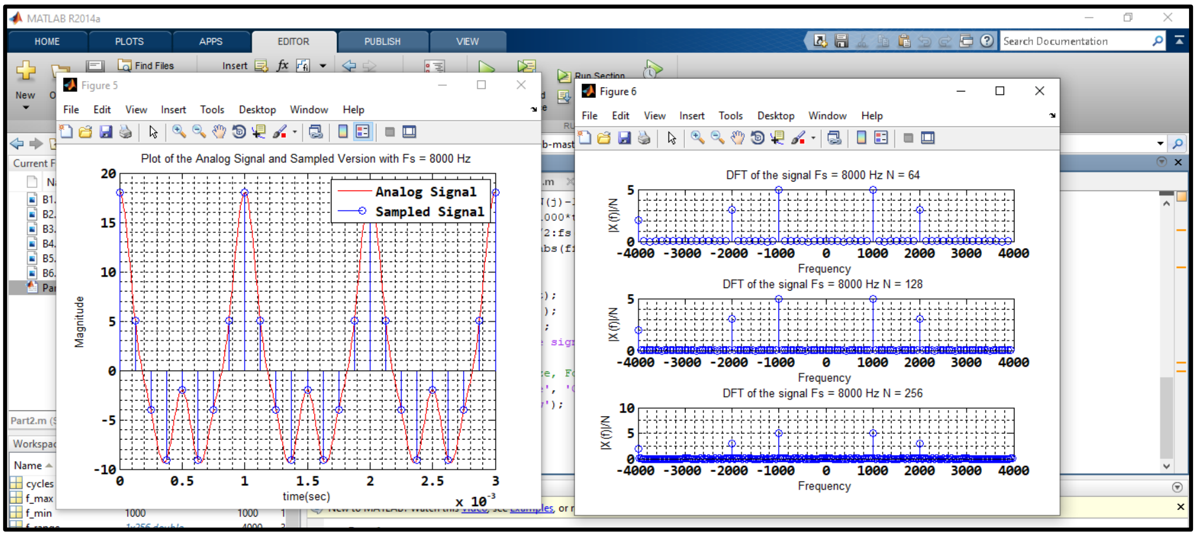

Fig 5: Sampling of original Analog signal and its Sampled Signal for frequency 8K

Stage 2: Quantization Results

Quantization discrepancy between an input value and its quantized value, like round-off error. A quantizer is a component or algorithm that conducts quantization. we study Uniform Quantization of Grayscale image. By creating a 1-bit and 2-bit uniform quantizer, we quantized an 8-bit grayscale image. Thresholding was used to convert pixel values falling within a specific range to a specific pixel quantization value, which allowed for uniform quantization of pixel intensities. The threshold was set at 127 in the 1-bit uniform quantization example, and all pixels below that value were mapped to value 64 while the remaining pixels were translated to value 192. After quantization, the complete image was thus only represented by 2 pixel values. Four stages of quantization were used for 2 Bit quantization. Fig 6 illustrates 1 bit and 2 bit Uniform Quantization.

Stage 3: Filtering using Spatial Domain Filter

Fig 7 shows the image before filtering and Fig 8 shows image after applying spatial domain filter [15].

Stage 4: Noise cancellation using Wiener Filter

There are two sensors in this issue: a primary sensor and a secondary sensor. The signal s[n], which is tainted with noise v1[n], is received by the principal sensor. A signal that is correlated with the noise v1[n] is received by the secondary sensor as v2[n]. Estimating s[n] is the goal.

We have used Wiener filter for noise cancellation. The Wiener filter (WF) can be employed to approximate a desirable or goal randomized process by performing LTI) filtering to a recorded noisy system and providing established static signal and noise characteristics as well as additive noise. The Wiener filter minimises the MS error between the desired process and the predicted random processes. Figure 9 to fig 12 display the desired signal and the signal with disturbance using different order Wiener Filter.

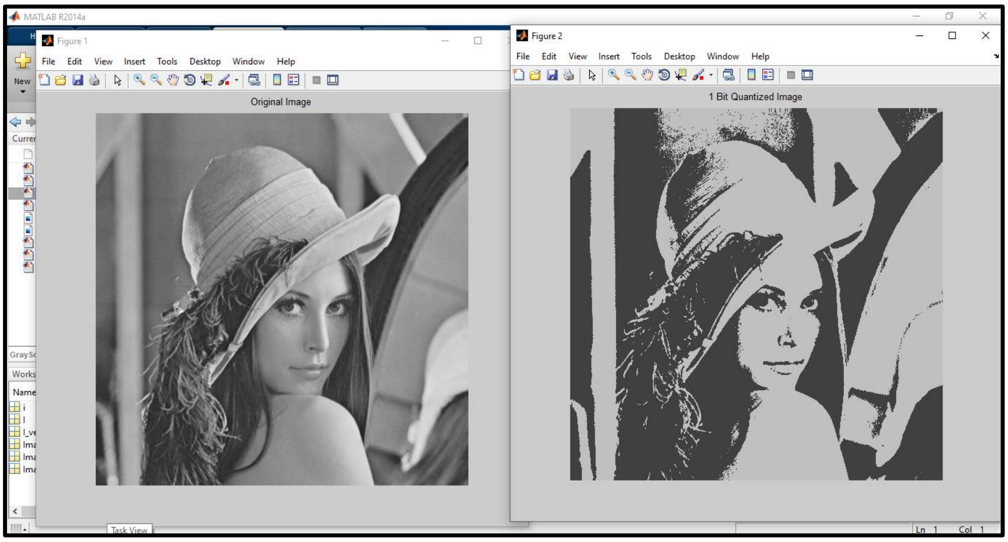

Fig 6: 1-bit and 2-bit Uniform Quantization

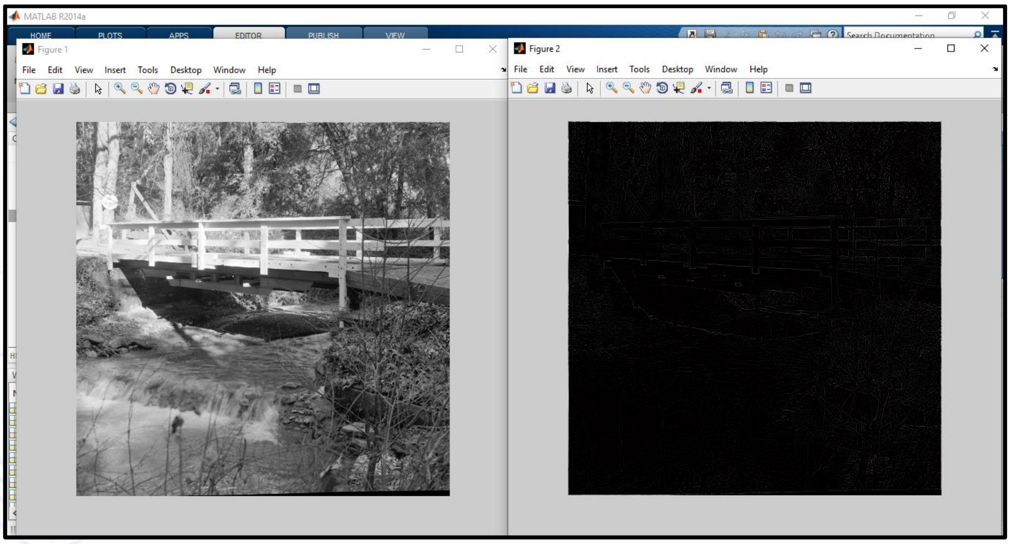

Fig 7: Image before filtering

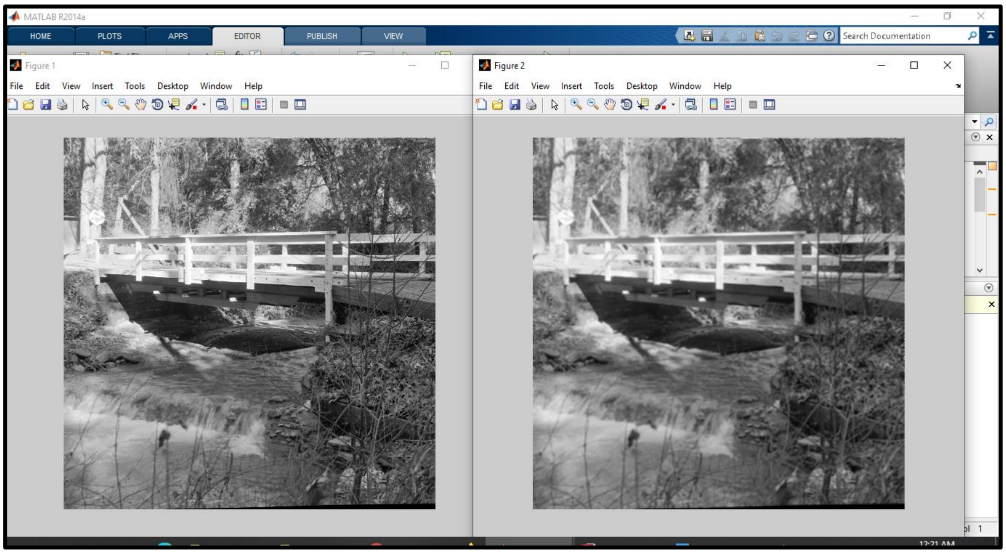

Fig 8: Image After filtering

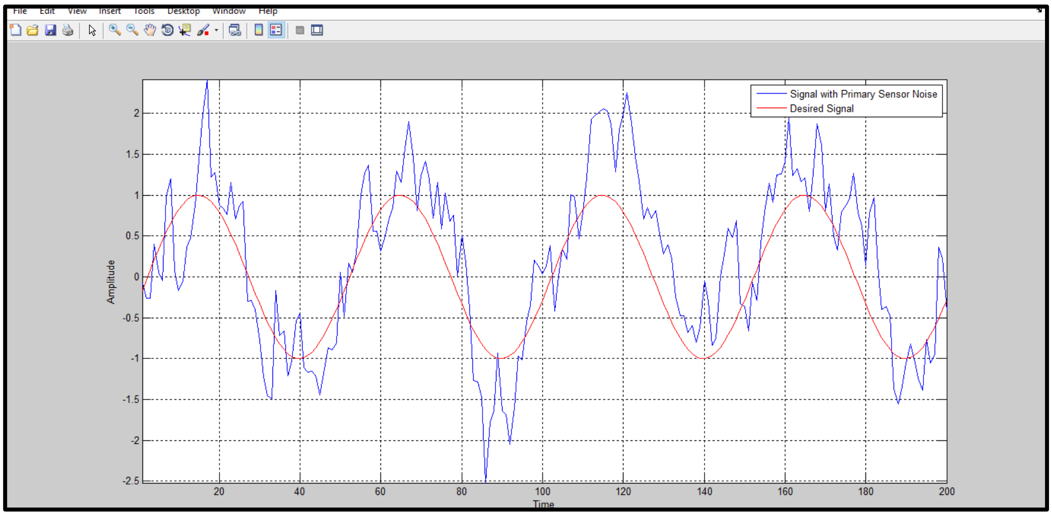

Fig 9: Required signal and Nosy signals

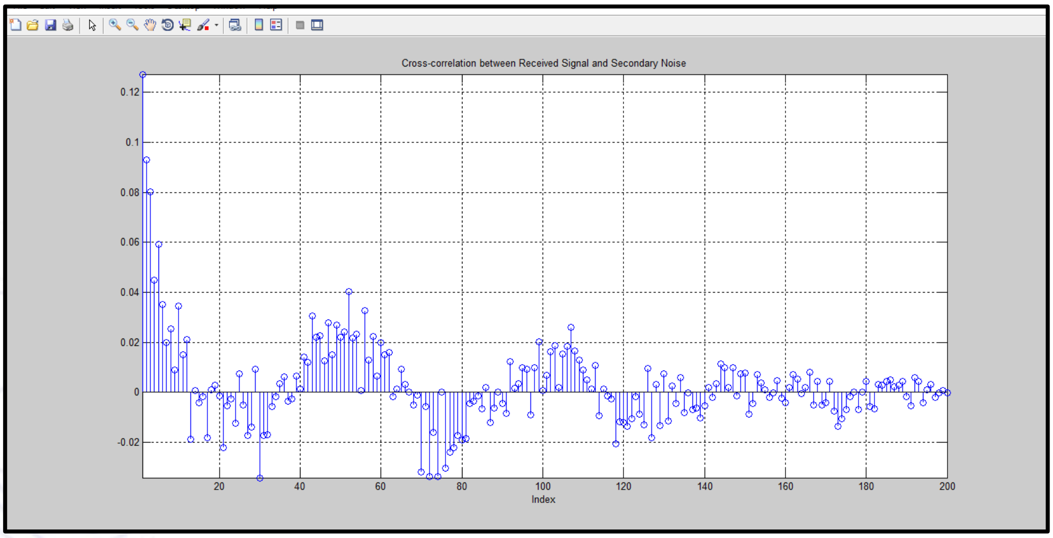

Fig 10: Link between received signal and other noises

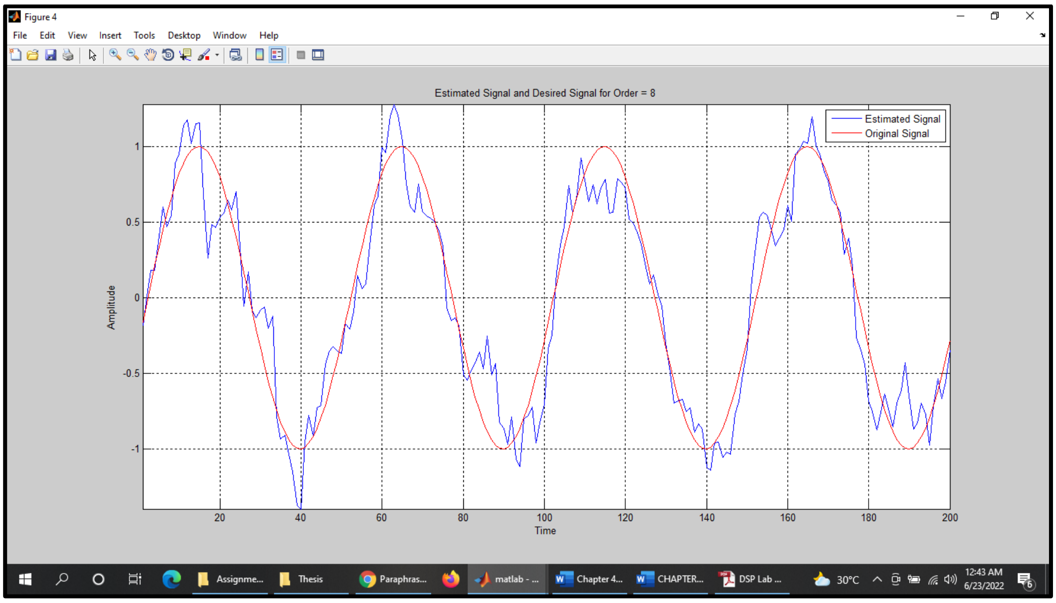

Fig 11: Approximated signal and original signal for 8th order

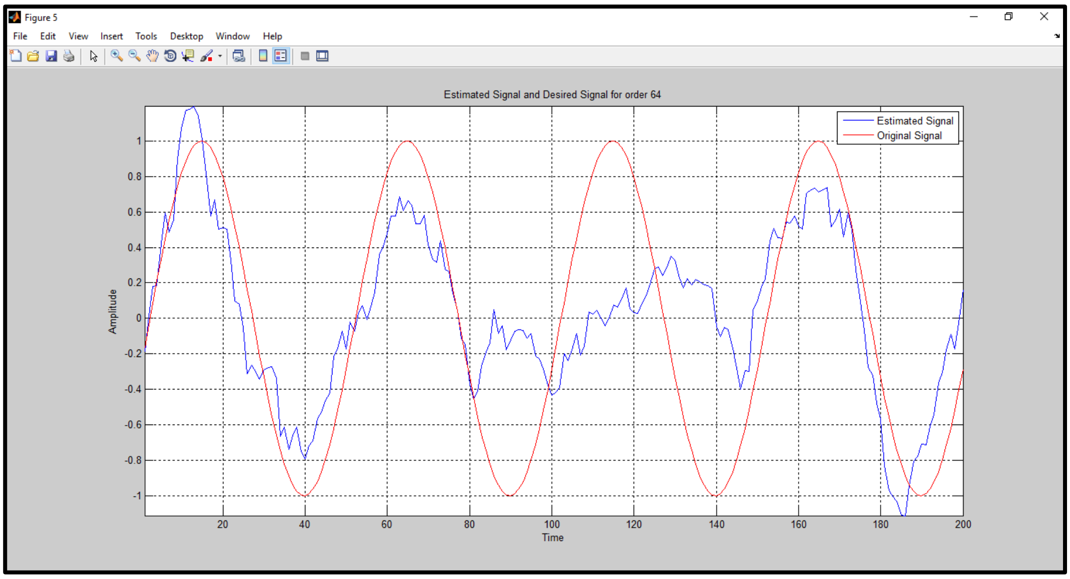

Fig 12: Approximated signal and original signal for 64th order

By employing a connected signal as an input and sifting that identified signal to obtain the estimation as an output, the WF aims to determine an arithmetical approximation of an unidentified signal.

For instance, the known signal might be made up of a potentially valuable uncertain signal that has been tampered with by additive noise. By removing the noise from the distorted signal, the Wiener filter can estimate the underlying signal of interest.

CONCLUSION

Wiener filtering uses the values of sample cross-correlation, hence we need the process to be stationary and ergodic. We have chosen a harmonic process as the appropriate signal for this reason. The primary sensor detects the channel output, while the secondary sensor detects the surrounding environment to provide noise cancellation. We still get a respectable result with the naive approach. This is because there is a significant correlation between v1 and v2, thanks to the constant "a." It might not be the case in a real-world situation.

As compared to the 1 Bit quantized image, the visual contrast in the first scenario significantly improved. There were some vertical lines seen. Although there were more horizontal lines added in the second section, we still saw an improvement in the image contrast. We diffused the fault to all the nearby pixels in the third section to achieve the best image clarity. Overall, diffusing error resulted in less quantized image error than the original image.

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