Digital Hearing Aid

Submitted to
Asst Prof Piyush Tiwari
Dept of Electronics and Communication
Delhi Technological University

Gaurav Saxena(2K19/EC/059)
Himanshu Verma(2K19/EC/075)

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1. ABSTRACT

In this project we developed a noise reduction filter and frequency shaper using MATLAB programming language. We also developed a GUI which takes into account the input from the user and proceeds further which shall be clear as we move on to further slides.

This digital hearing aid system is design to adapt for normal and moderate hearing loss patient.

2. Introduction

Why use a digital hearing aid?

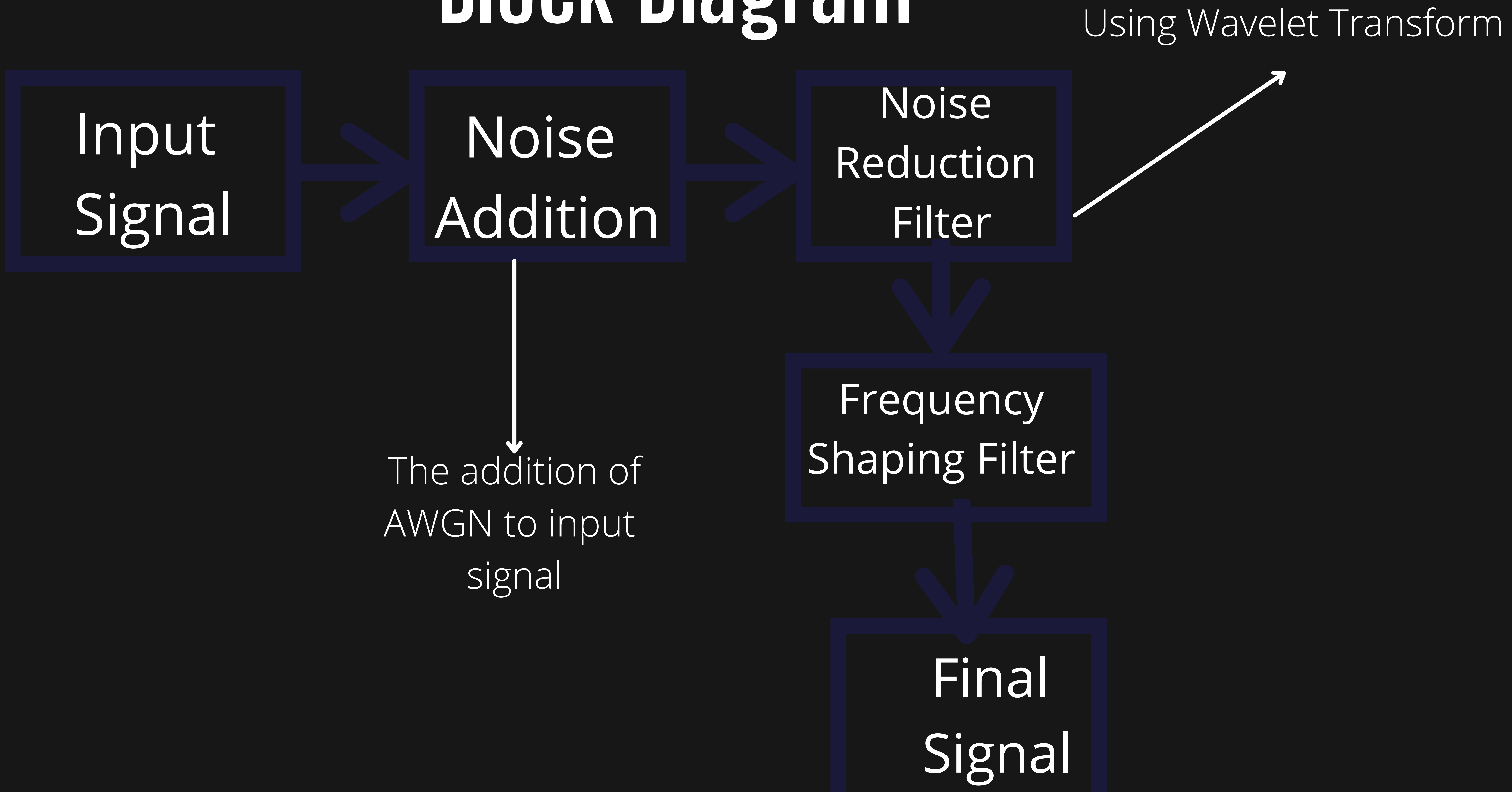
Roughly 10% of the world population bears from some hearing loss. However, only a portion uses hearing aid. This is due several factors which include the stigma associated with wearing a hearing aid, customer dissatisfaction with the devices not meeting their expectations, and the cost associated with the new digital versions of hearing aids. Hearing loss is typically measured as the shift in auditory threshold relative to that of a normal ear for detection of a pure tone

. Improvements have been made by using the development of digital sound treatment for the efficiency of hearing aids[1]. Nowadays, the digital hearing aids are small, which can be hidden inside the ear and have an almost perfect sound reproduction. The research of Digital hearing aids has been growth and now a small programmable computer that are capable in amplifying millions of different sound signals had been constructed in the devices, thus improving the hearing ability of hearing-impaired people. The first digital hearing aids were launched in the mid 80’s, but these early models were slightly unpractical. After ten years later, the digital hearing aids really became successful, with small digital devices placed either inside or discreetly behind the ear.

Different degrees of Hearing Losses

Degree of hearing loss	Hearing loss range (dB)
Normal	–10 to 15
Minimal	16 to 25
Mild	26 to 40
Moderate	41 to 55
Moderately severe	56 to 70
Severe	71 to 90
Profound	91+

Block Diagram



3. Design Details

1. *Noise Addition*

The program that we have built gives the user an option either to record his own voice as input or use some inbuilt audio data which can be used. MATLAB has several inbuilt audio data which can be seen from the fig below

```
Example audio data (MAT files).
  chirp         - Frequency sweeps          (1.6 sec, 8192 Hz)
  gong          - Gong                      (5.1 sec, 8192 Hz)
  handel        - Hallelujah chorus         (8.9 sec, 8192 Hz)
  laughter      - Laughter from a crowd     (6.4 sec, 8192 Hz)
  splat         - Chirp followed by a splat (1.2 sec, 8192 Hz)
  train         - Train whistle             (1.5 sec, 8192 Hz)
```

Code snippet of Noise addition

```
load laughter.mat
filename = 'laughter.wav';
audiowrite(filename,y,Fs);
xn = awgn(y,24,'measured');
subplot(2,1,1)
plot(y);
subplot(2,1,2)
plot(xn);
```

awgn(in,snr,signalpower) accepts an input signal power value in dBW. To have the function measure the power of in before adding noise, specify signalpower as 'measured'.

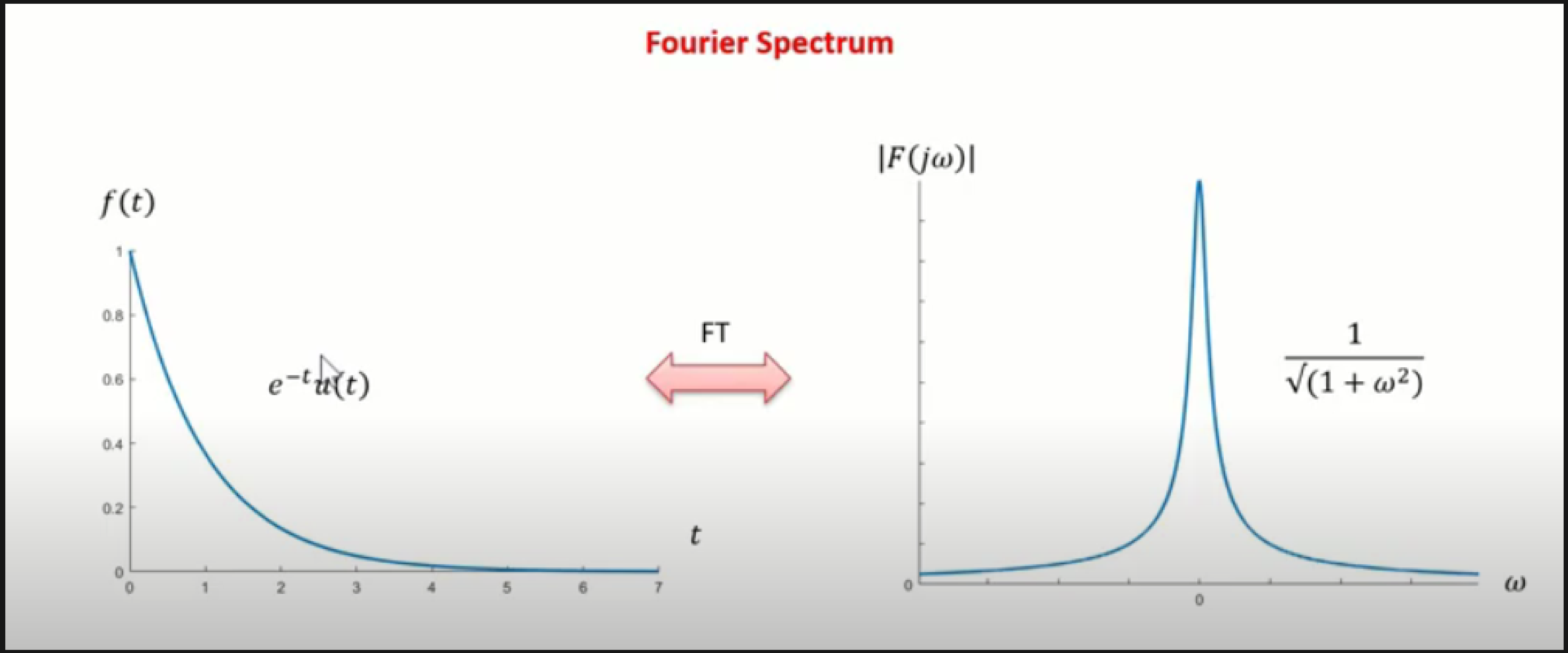
Add a little bit of body text

2. Noise Reduction using Wavelet Transform

Drawbacks that are faced with fourier transform and led to use Wavelet Transform

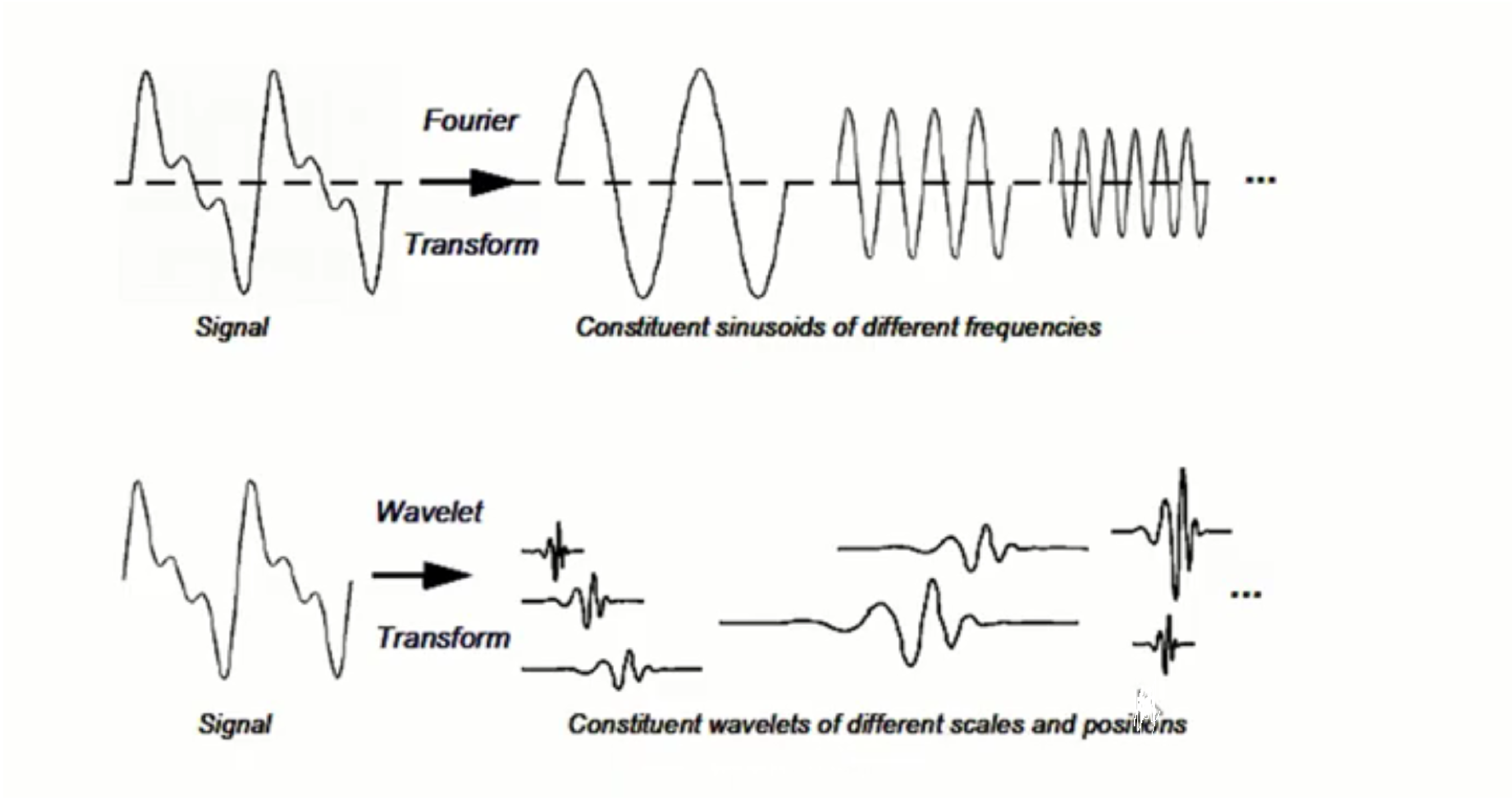
To put simply a fourier transform (FT) will tell you what frequencies are present in your signal. A wavelet transform (WT) will tell you what frequencies are present and where (or at what scale). If you had a signal that was changing in time, the FT wouldn't tell you when (time) this has occurred.

Consider the example : Suppose we have a exponential function and its fourier tansform taken as given below



The above spectrum is between the magnitude and its frequency.If something happens we cannot tell at which time it occurs. There are many other examples which helps us to understand the drawbacks of fourier transform.

Comparasion of Wavelet and Fourier Transform



Wavelet Transform

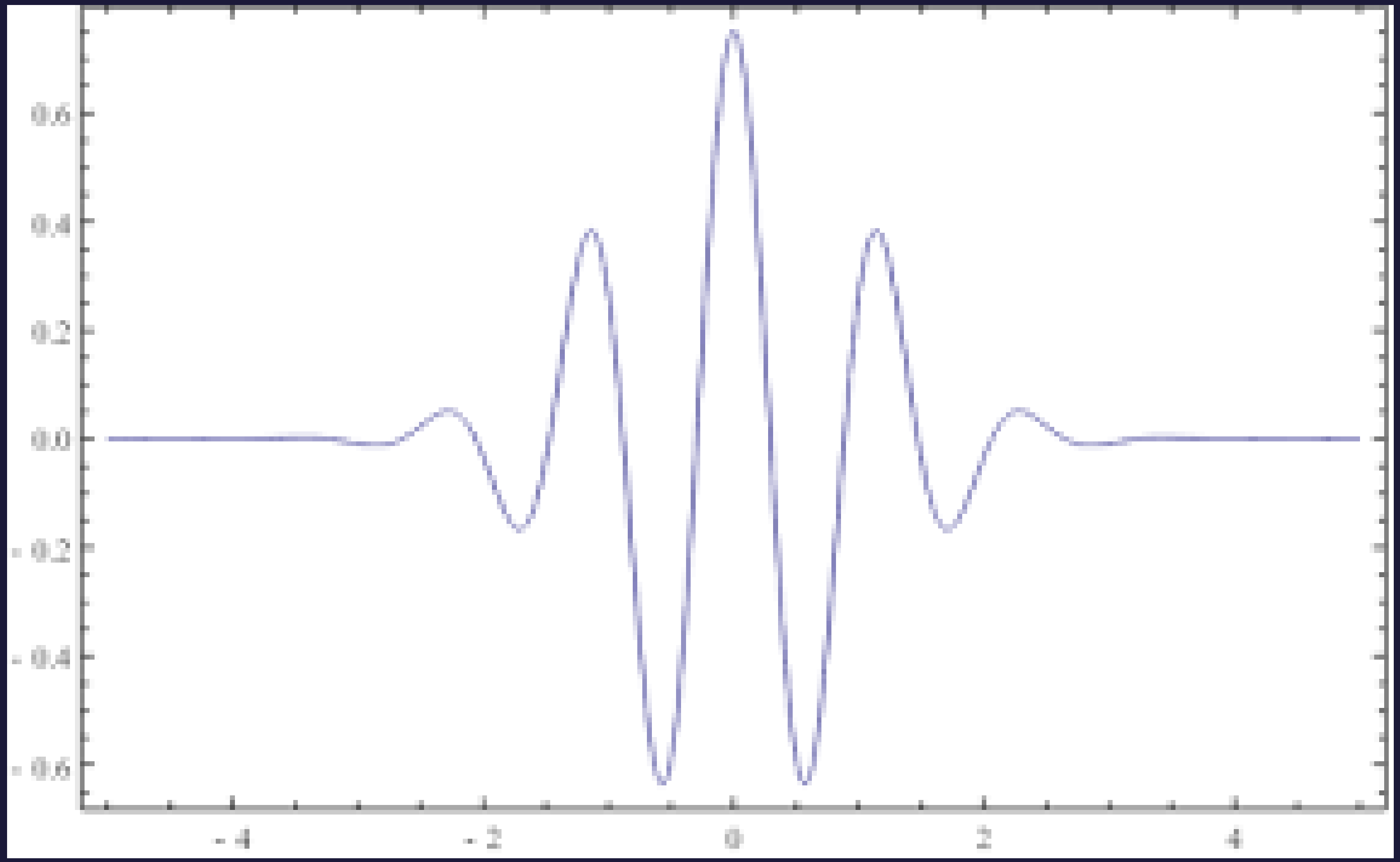
A wavelet is a waveform of effectively limited duration that has an average value of zero.

$$\varphi_{\tau,f}(t)dt = \frac{1}{\sqrt{a}}\,\varphi^{*}\left(\frac{t-b}{a}\right)$$

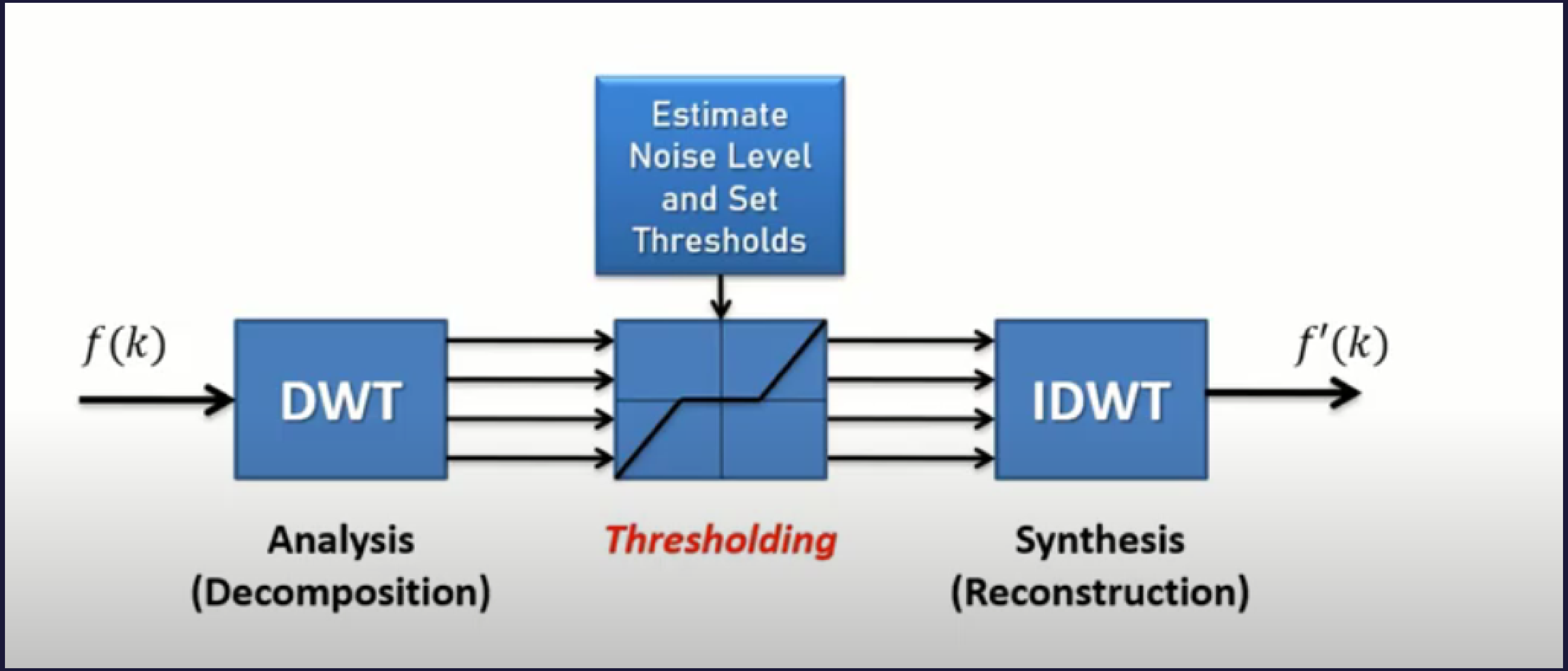
where a is Dilation(Scale) Parameter and b is the Translation(position) parameter

The wavelet transform is one of the operative transforms for processing one-dimensional audio signals, where it provides outstanding results on audio signals even if wavelets transform is operating on multiresolution frequencies.

Example: Morlet Wavelet



Wavelet Based Desnoising Scheme



Here the input signal is first decomposed using the discrete wavelet transform in various labels and we obtain coefficients after that,Approximated and Detailed Coffecients. All the detailed coffecients are shrinked and then these modified wavelet cofficient are collected togethe and we take the inverse discrete wavelet transform to achieve the signal back.

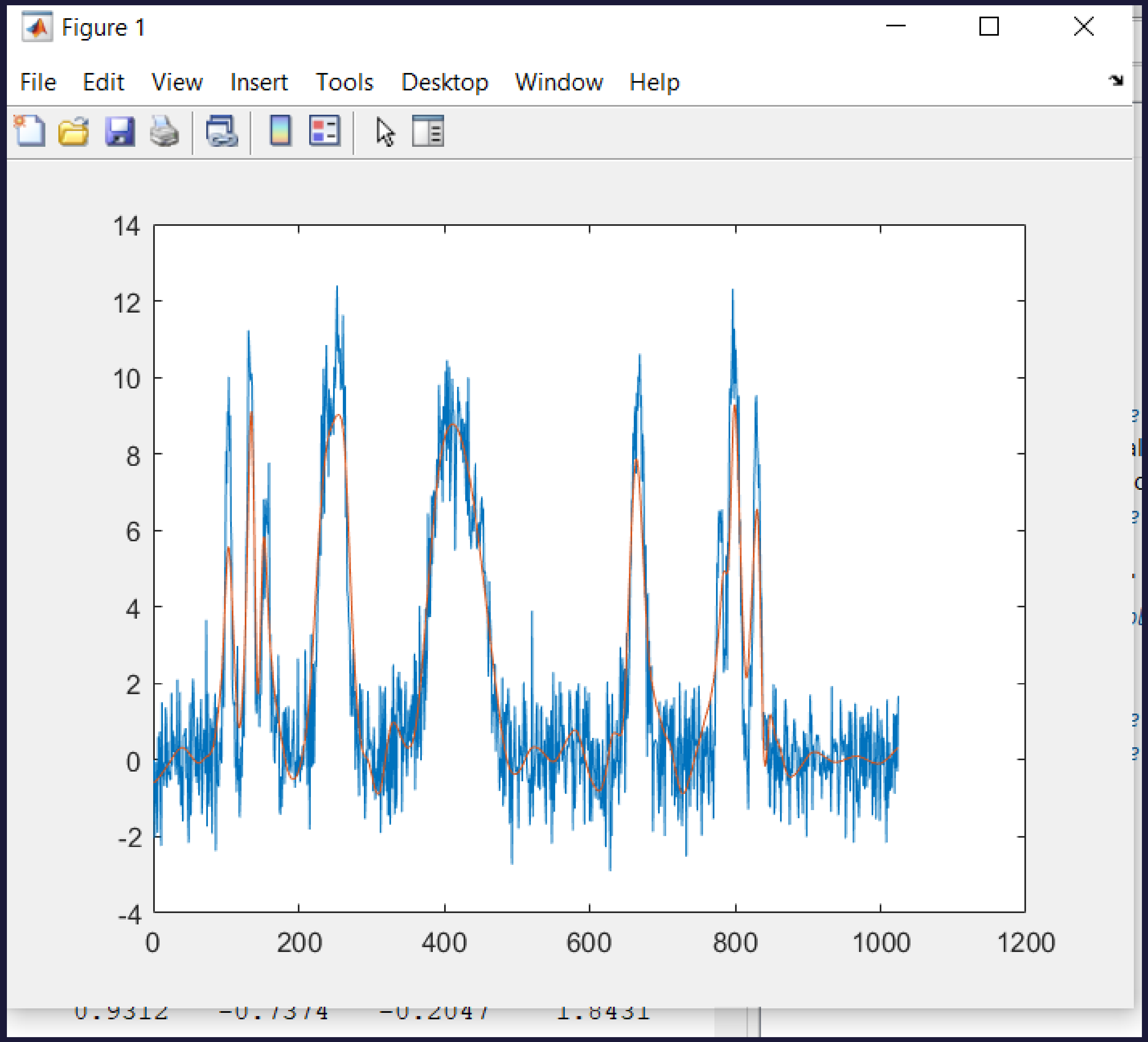
The next challenge is to actually decide the threshold and to decide noise estimation methods there are certain methods by which we find appropiate threshold values.

Noise Estimation Methods	MATLAB Functions	
MiniMax Criterion	wden	wdenoise
SqtwoLog Criterion (UniversalThreshold)	wden	wdenoise
RigrSURE (Rigorous Stein's Unbiased Risk Estimate)	wden	wdenoise
HeurSURE (Heuristic Stein's Unbiased Risk Estimate)	wden	
Bayes (Empirical Bayes)	-	wdenoise
BlockJS (Block James-Stein)	-	wdenoise

Wden(one dimesional denoising function)

Code snippet

```
load wnoisydata
xnn = table2array(wnoisydata)
xn = xnn(:,1);
xden = wden(xn,'sqtwolog','s','mln',5,'sym8');
h1 = plot([xn xden]);
h1(1).Linewidth = 3;
```



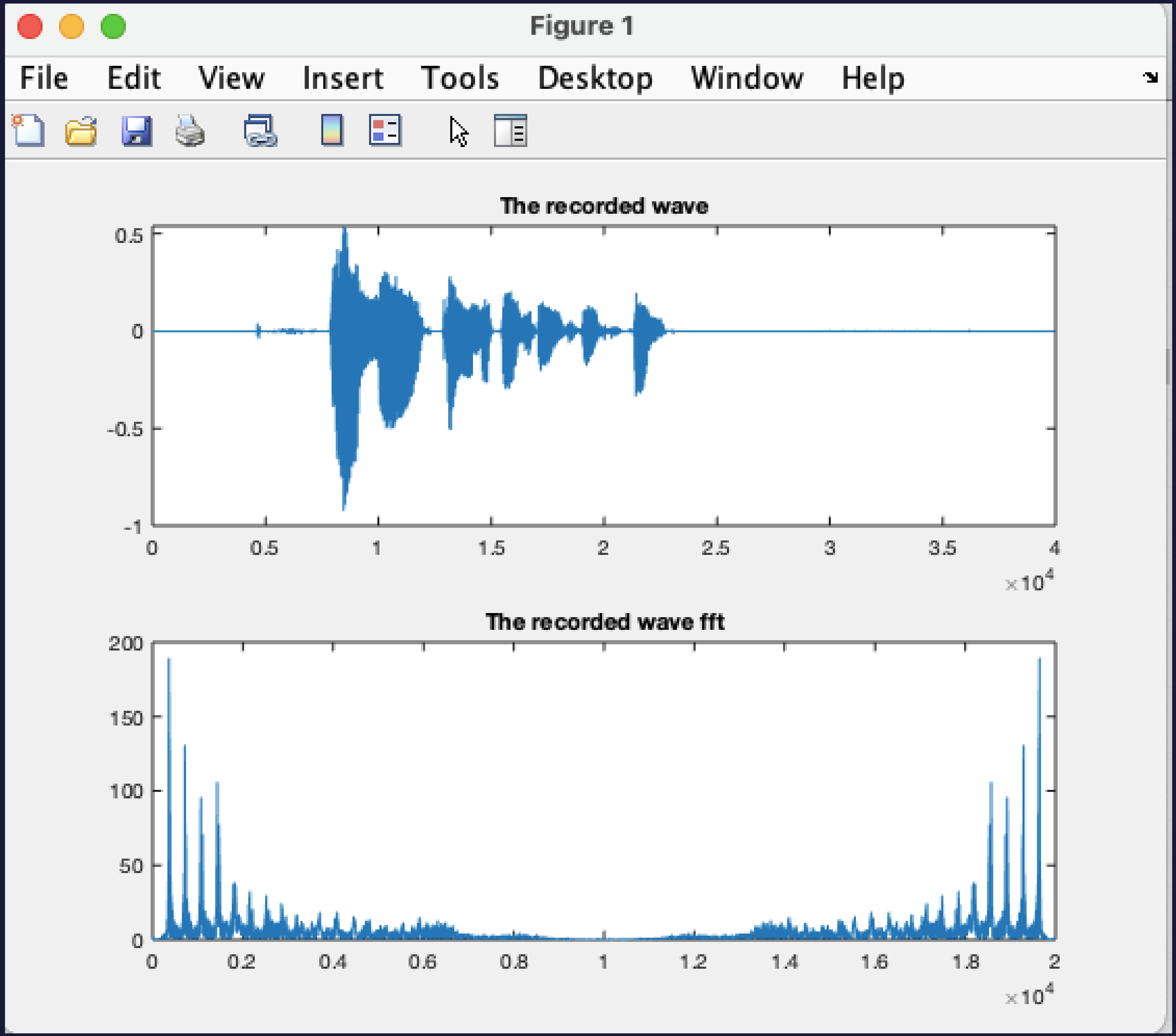
Frequency Shaping Filter

Frequency-selective filters attempt to exactly pass some bands of frequencies and exactly reject others. Frequency-shaping filters more generally attempt to reshape the signal spectrum by multiplying the input spectrum by some specified shaping.

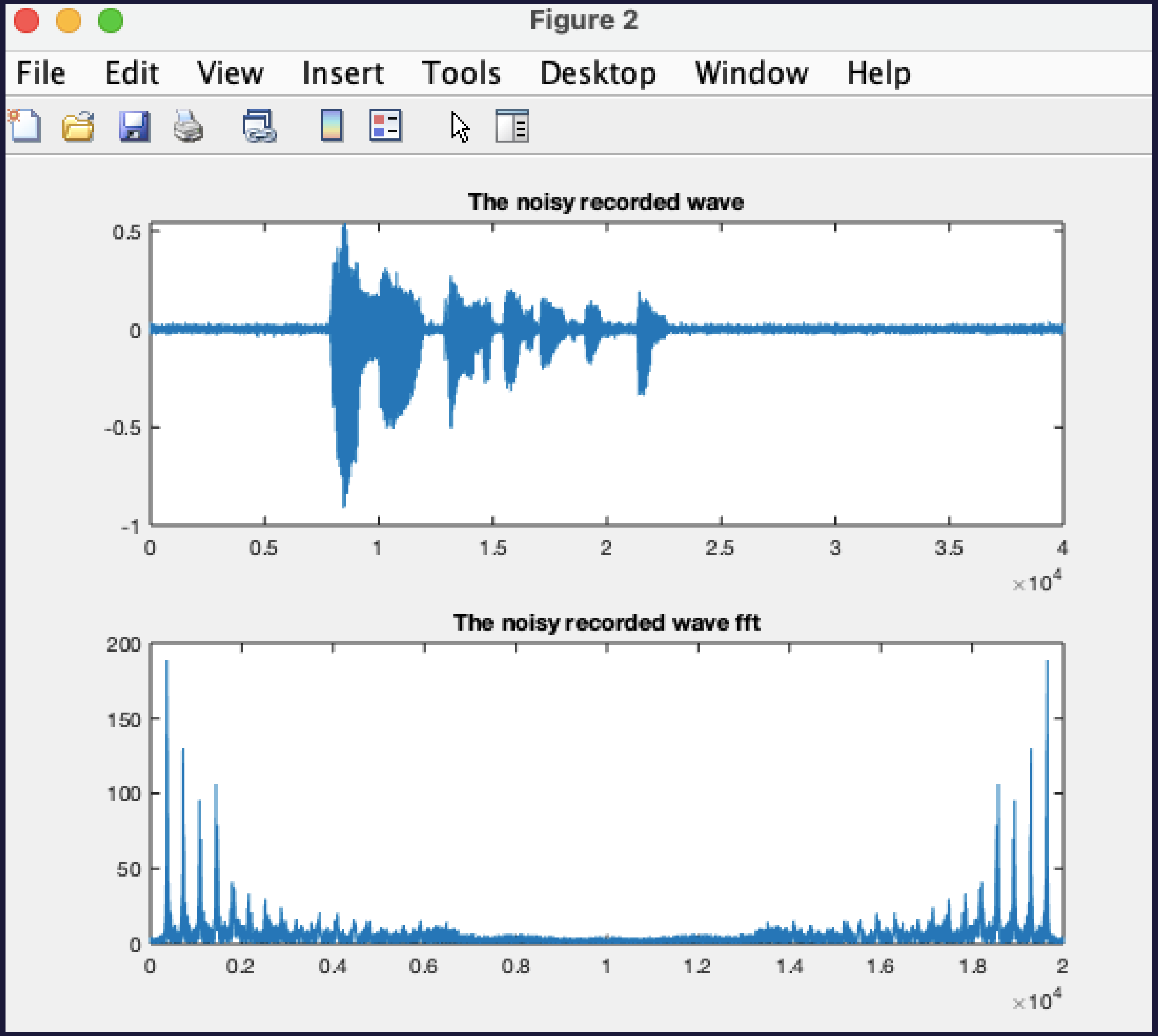
The filter applies a gain greater than one to the frequencies that the user has difficulty hearing.As one of its parameters, the filter takes in a vector of frequencies, determined by an audiologist, that define the user's hearing characteristics. For each range, the frequency shaper applies a certain gain based on the user's specific hearing loss. Thus, our frequency shaper iscompletely configurable to any user. We implemented our frequency shaper in MATLAB.

Results

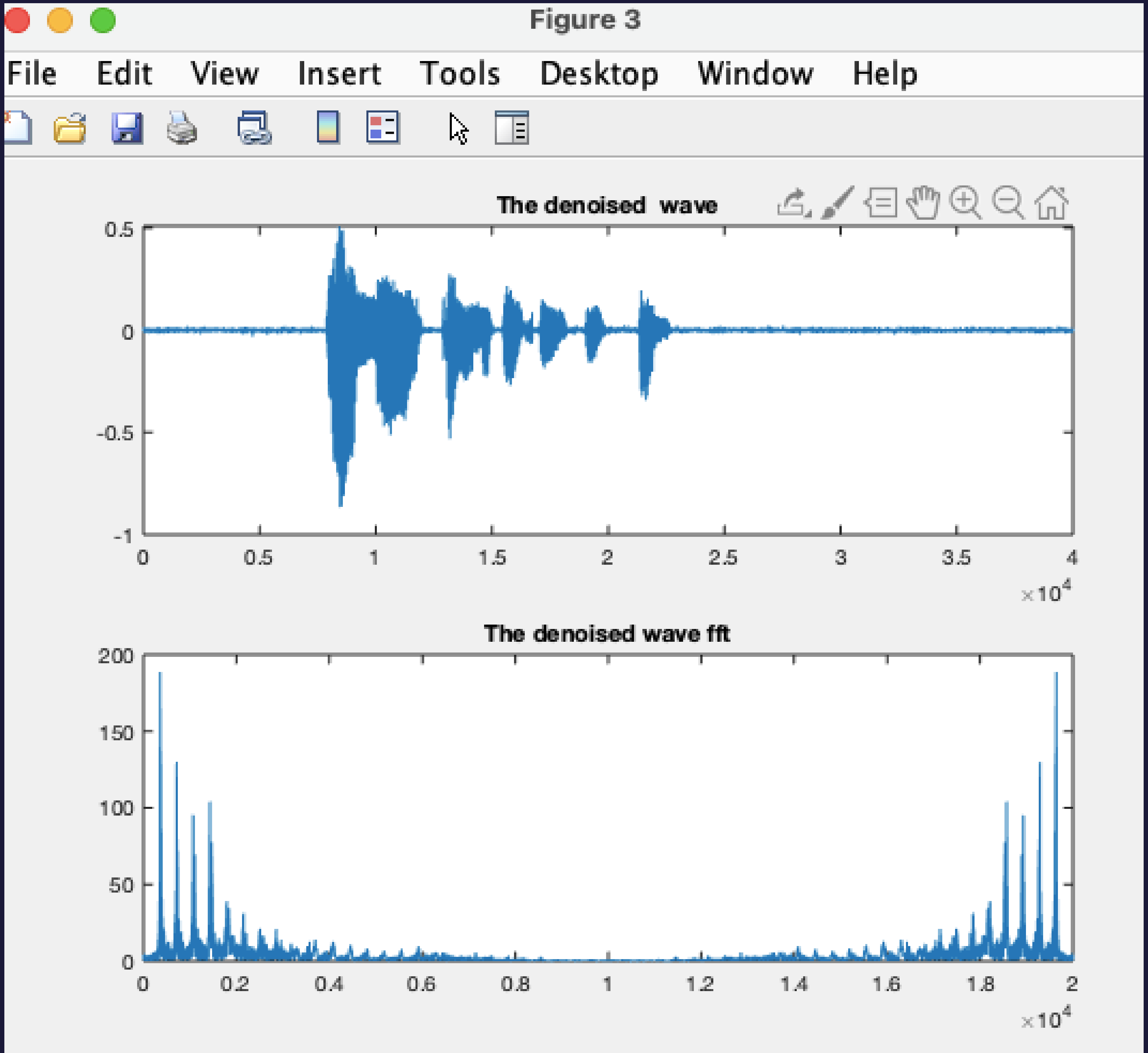
Figure 1 below is the original speech signal



This signal is added by Adaptive White Gaussian Noise (AWGN) and random noise.
Next, Adaptive White Gaussian Noise is added to the original wave signal. The purpose of this addition just to simulate noises in the real life situation.
Figure 2 shows the signal after noise addition.



Afterward, the denoising process takes place which removes most of the noise in the signal as shown in figure 3



Afterward , the filter shaping takes place which provide certain gain based on the user’s specific loss

Our Subject has moderate hearing loss at medium frequency range of 3000 - 6000Hz.The required gain is 45db on this frequency range. So the frequency shaping or modified function is show below

By using frequency shaping function , gain of speech signal has been modified on specific frequency range, and the processed signal has increased the gain within limits, Fig 4.2.1 ,4.2.2 shows the relative magnitude of

input speech signal and processed output signal/adjusted signal with no noise

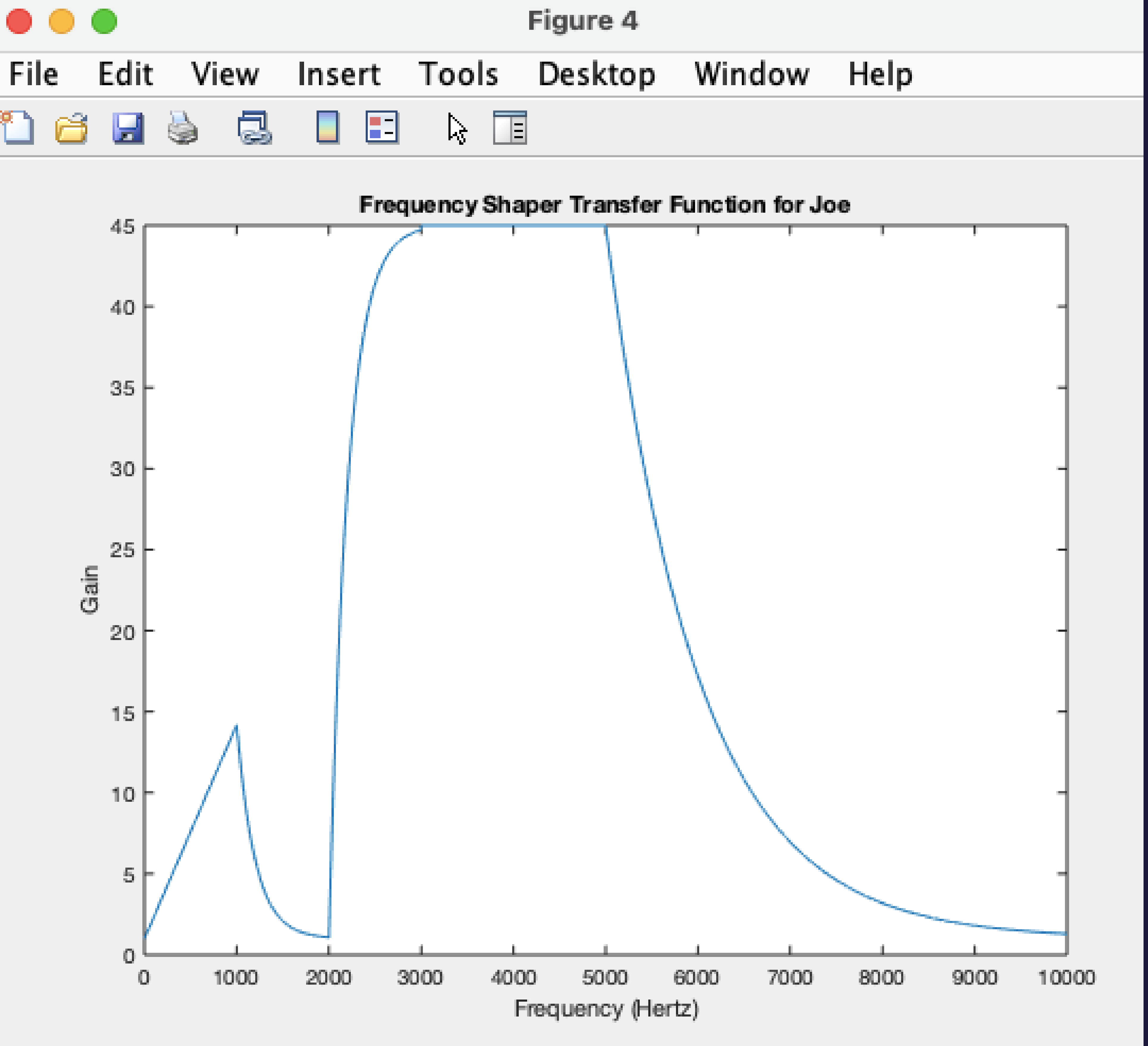


Figure 4
File Edit View Insert Tools Desktop Window Help
Frequency Shaper Transfer Function for Joe
Gain
Frequency (Hertz)
0 5 10 15 20 25 30 35 40 45
0 1000 2000 3000 4000 5000 6000 7000 8000 9000 10000

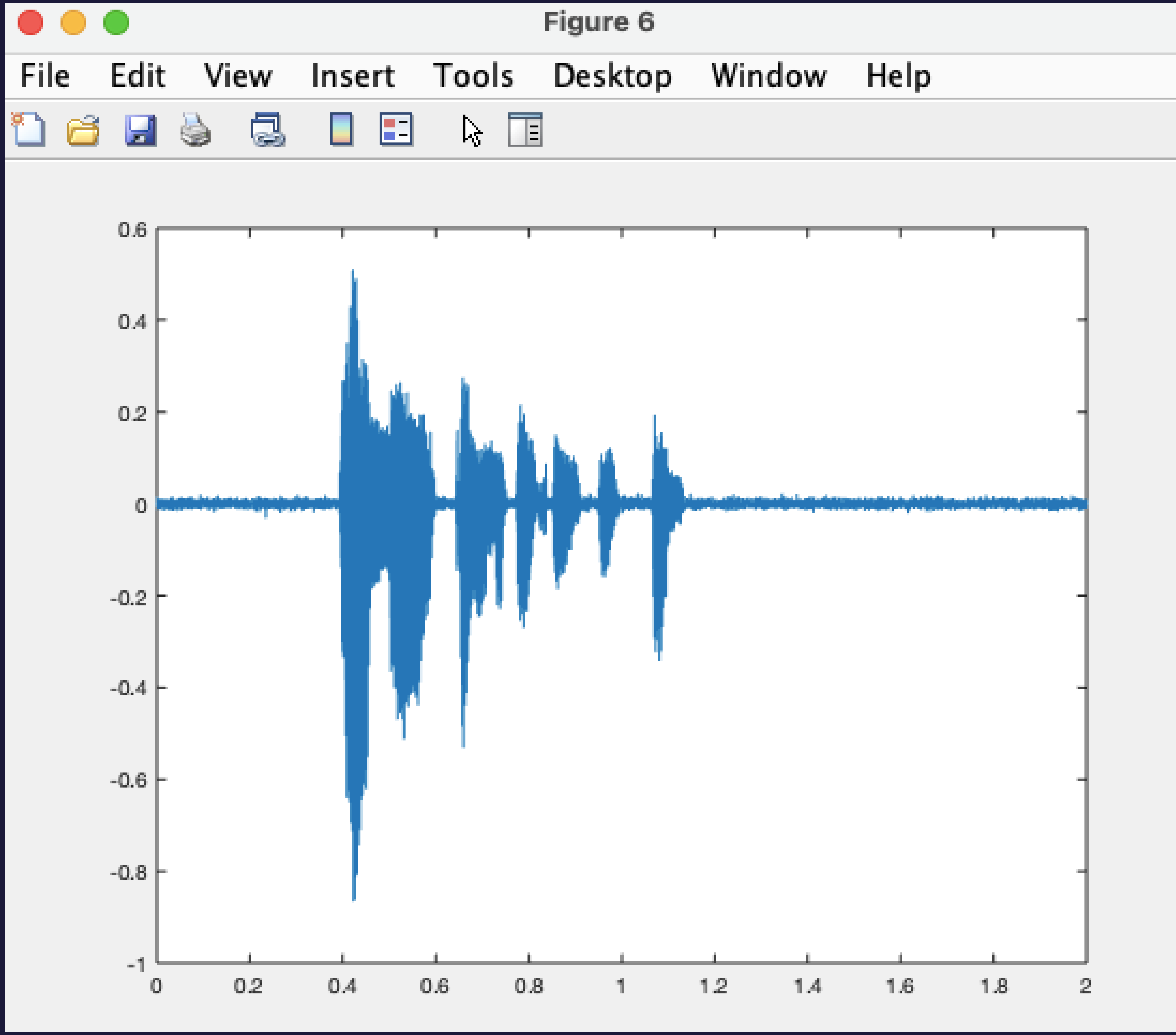


Figure 6
File Edit View Insert Tools Desktop Window Help
0.6 0.4 0.2 0 -0.2 -0.4 -0.6 -0.8 -1
0 0.2 0.4 0.6 0.8 1 1.2 1.4 1.6 1.8 2

figure 4.2.1

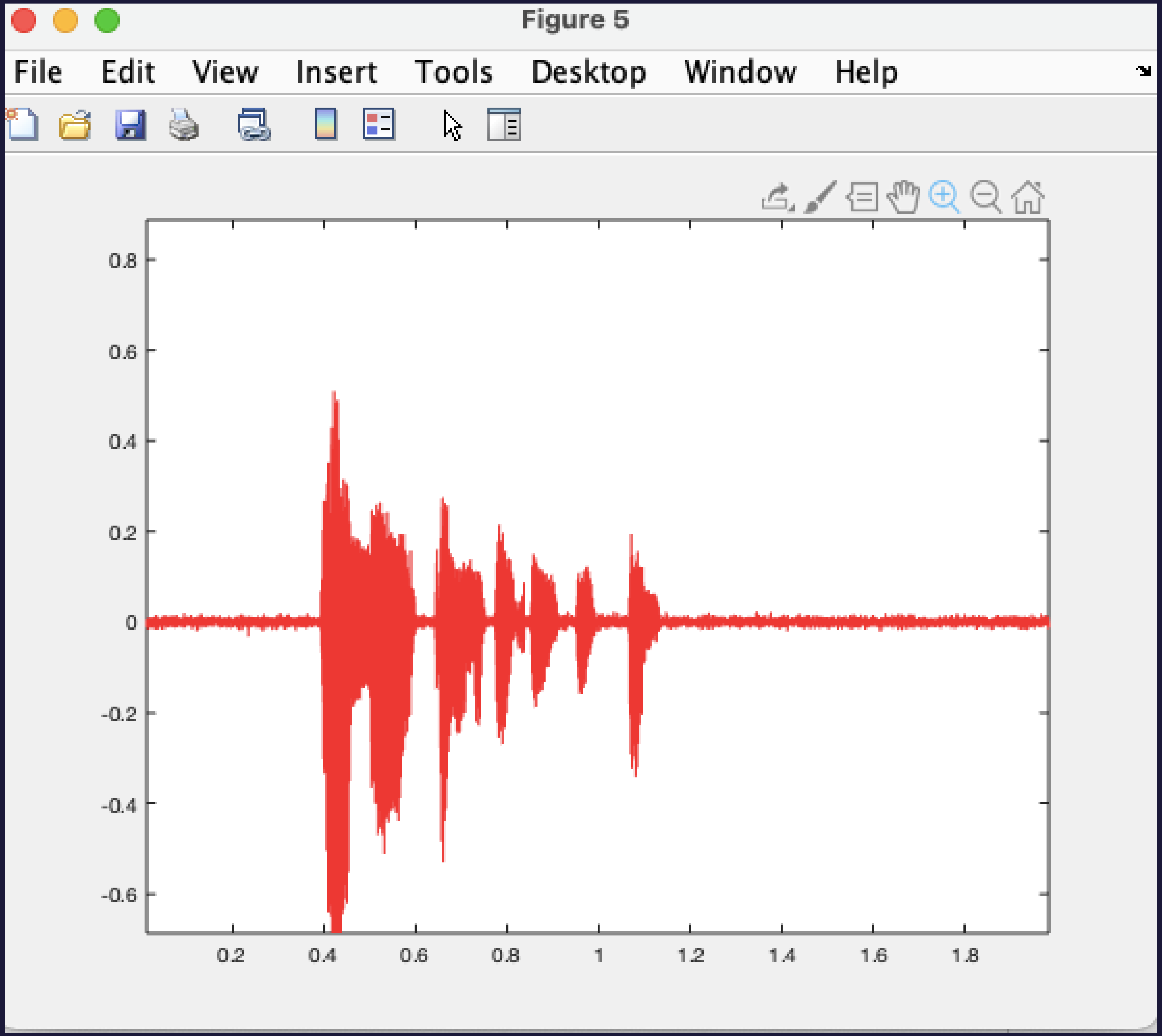
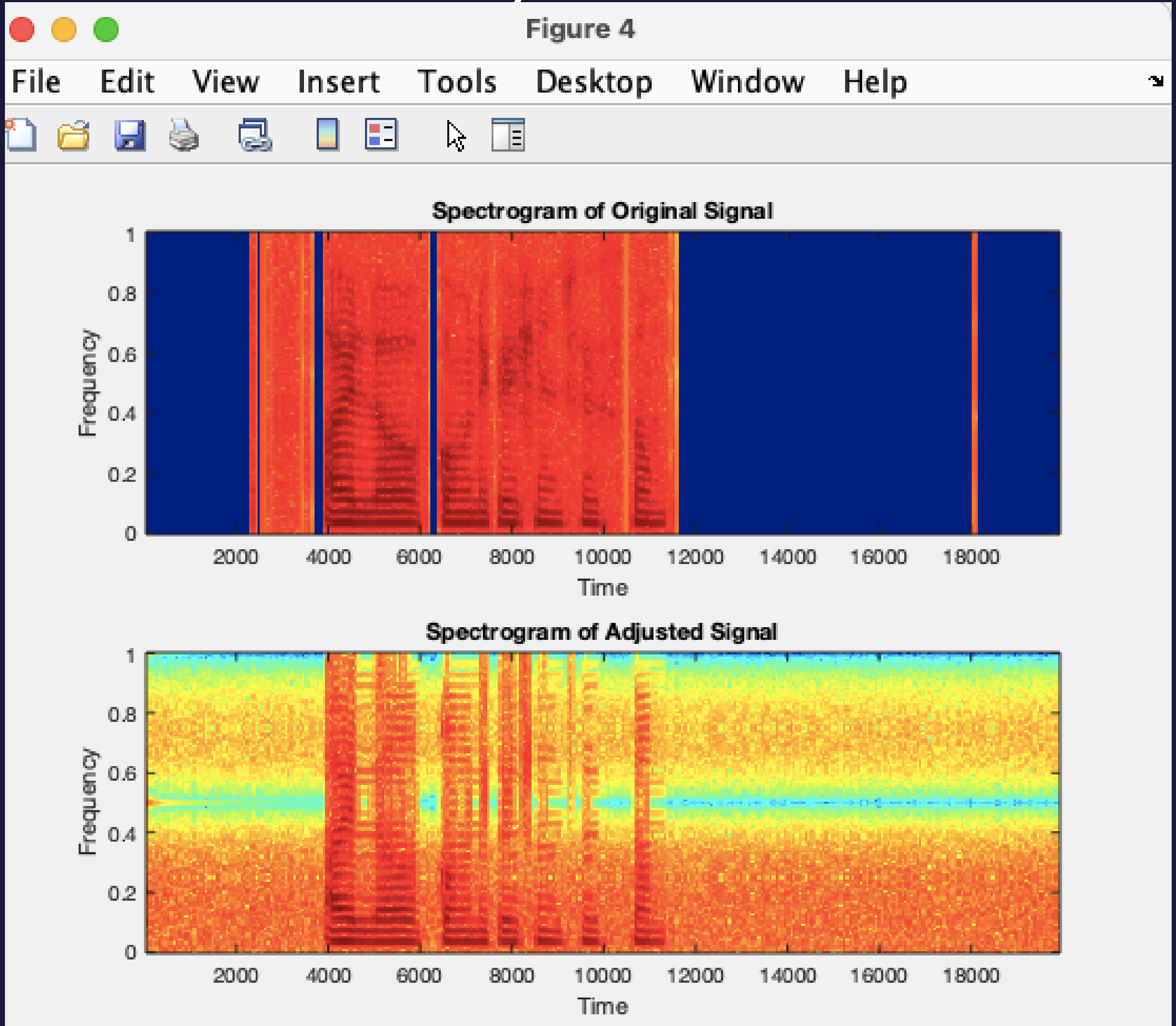


Figure 5
File Edit View Insert Tools Desktop Window Help
0.8
0.6
0.4
0.2
0
-0.2
-0.4
-0.6
0.2 0.4 0.6 0.8 1 1.2 1.4 1.6 1.8

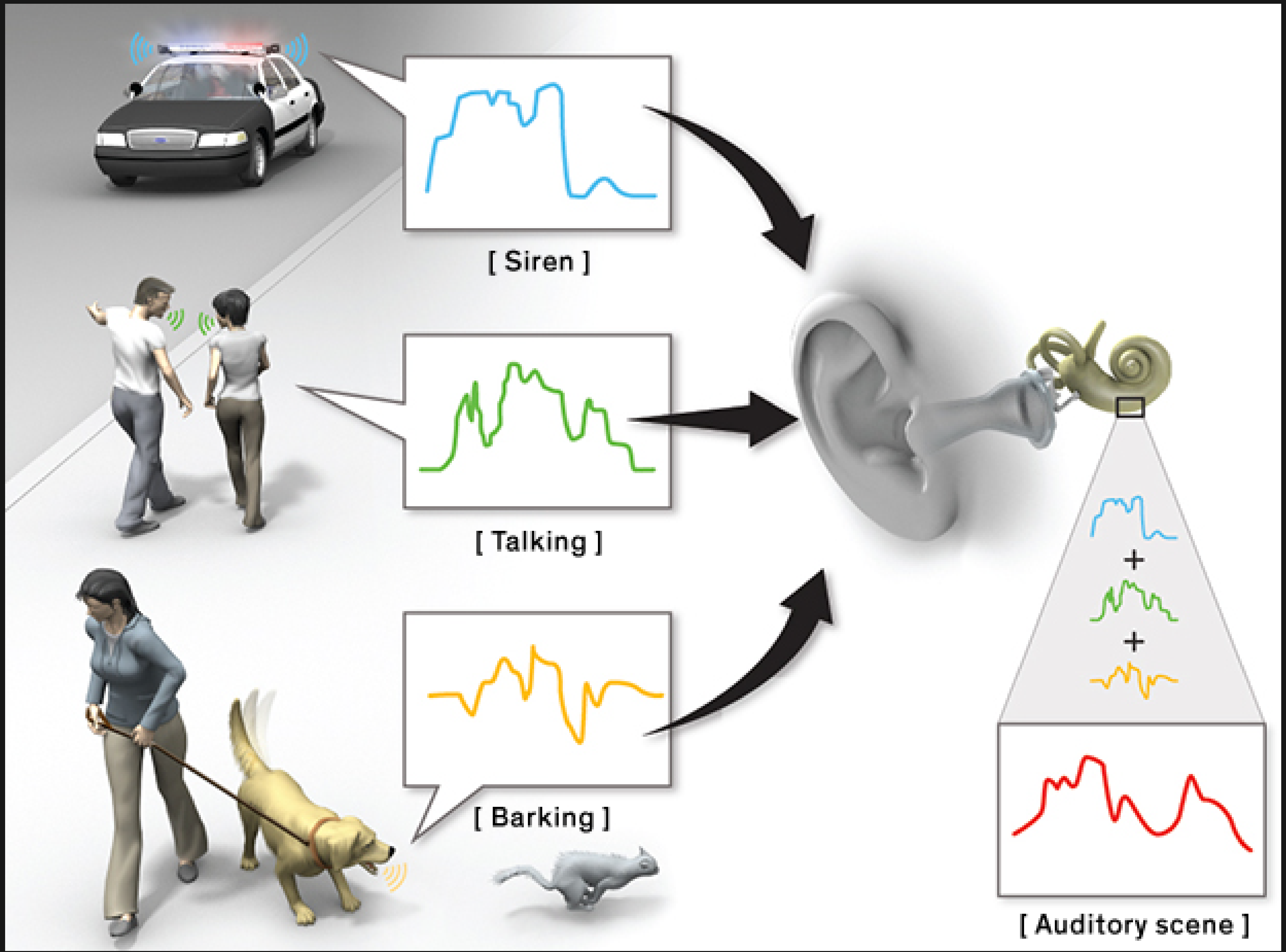
figure 4.2.2

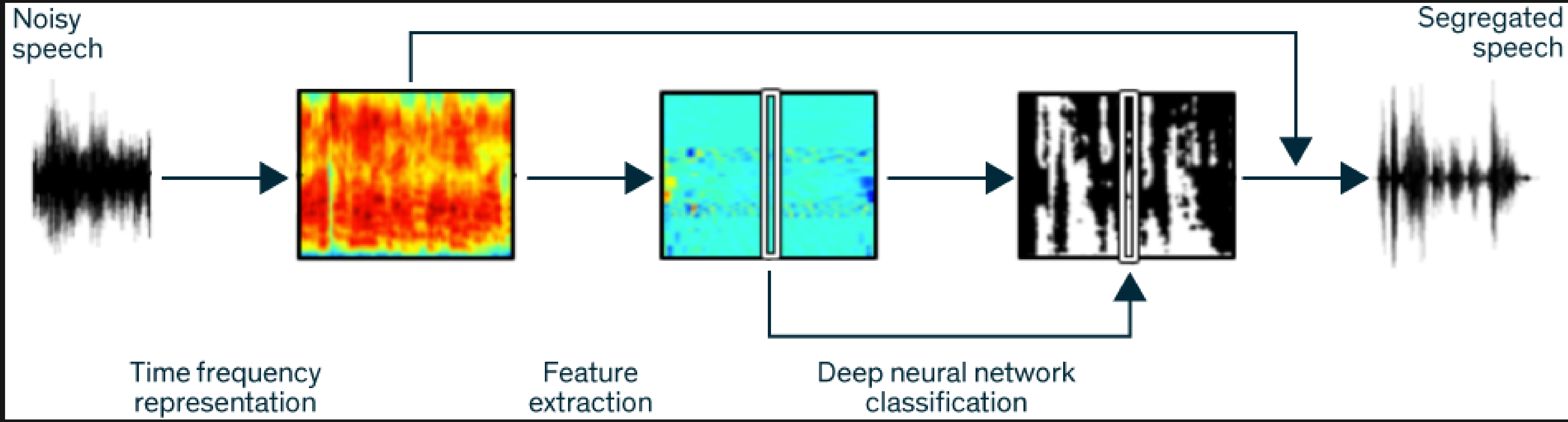
Comparing the spectrograms of the original signal and the filtered signal, we can see that the amplitude of the noise in the signal was noticeably reduced as shown in figure 4 below.



Future Vision

Machine learning based on deep neural networks for segregating of sounds





To separate speech from noise, a machine learning program breaks a noisy speech sample into a collection of elements called time-frequency units. Next, it analyzes these units to extract 85 features known to distinguish speech from other sounds. Then, the program feeds the features into a deep neural network trained to classify the units as speech or not based on past experience with similar samples. Lastly, the program applies a digital filter that tosses out all the nonspeech units to leave only separated speech.

References

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