

University Of Bourgogne

Digital Signal Processing

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# DSP SYSTEM

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**vibot**



Membre de  
**UBFC**



## What is a DSP?

Digital Signal Processors (DSP) take real-world signals like voice, audio, video, temperature, pressure, or position that have been digitized and then mathematically manipulate them. A DSP is designed for performing mathematical functions like "add", "subtract", "multiply" and "divide" very quickly.

Signals need to be processed so that the information that they contain can be displayed, analyzed, or converted to another type of signal that may be of use. In the real-world, analog products detect signals such as sound, light, temperature or pressure and manipulate them. Converters such as an Analog-to-Digital converter then take the real-world signal and turn it into the digital format of 1's and 0's. From here, the DSP takes over by capturing the digitized information and processing it. It then feeds the digitized information back for use in the real world. It does this in one of two ways, either digitally or in an analog format by going through a Digital-to-Analog converter. All of this occurs at very high speeds.

To illustrate this concept, the diagram below shows how a DSP is used in an MP3 audio player. During the recording phase, analog audio is input through a receiver or other source. This analog signal is then converted to a digital signal by an analog-to-digital converter and passed to the DSP. The DSP performs the MP3 encoding and saves the file to memory. During the playback phase, the file is taken from memory, decoded by the DSP and then converted back to an analog signal through the digital-to-analog converter so it can be output through the speaker system. In a more complex example, the DSP would perform other functions such as volume control, equalization and user interface.



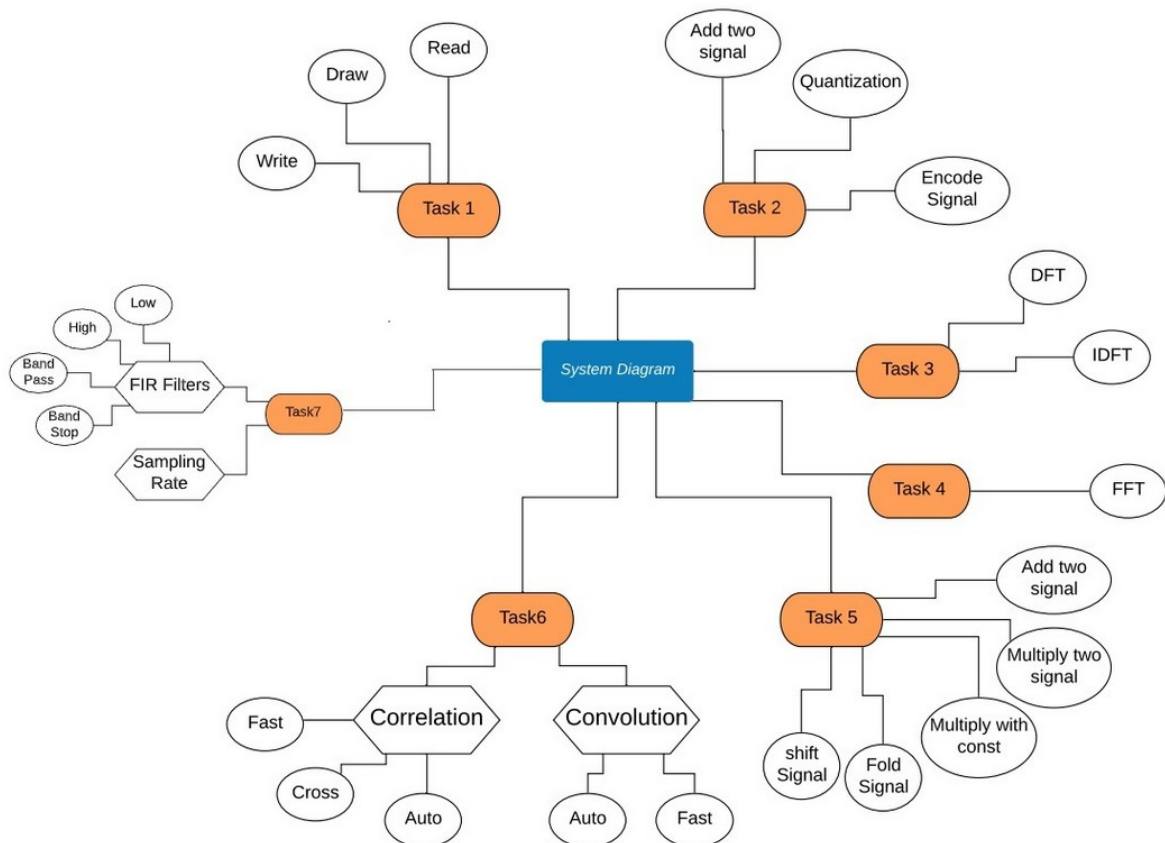
# DSP SYSTEM

## Goal :

This system aim to cover most of part in Signal Processing using Simple GUI to allow to any user to practice and understand the signals and this system divided to several part :

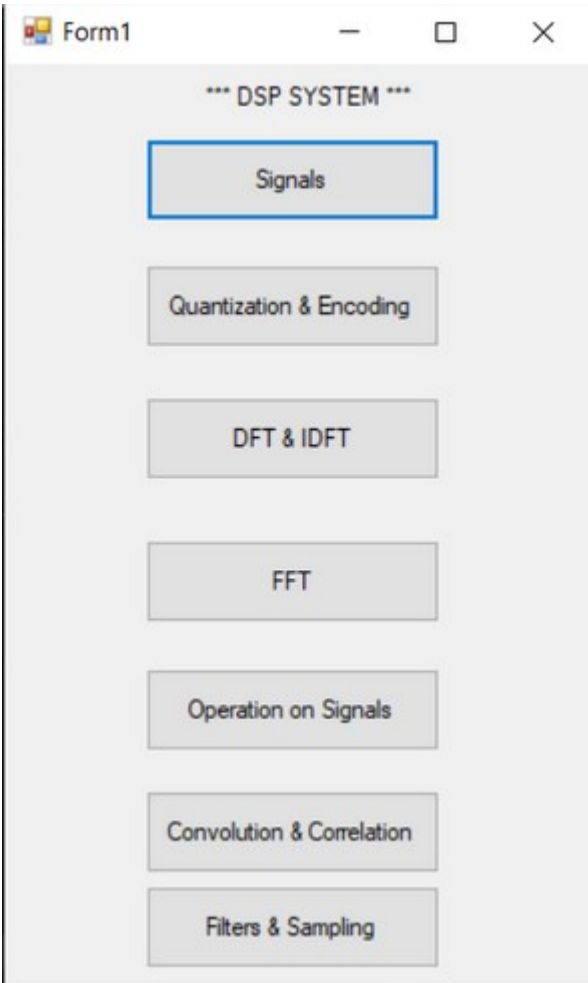
1. Main Window.
2. Signals.
3. Quantization and Encoding.
4. Discrete Fourier Transform (DFT) and (Inverse DFT).
5. Fast Fourier Transform (FFT).
6. Operation On Signals.
7. Convolution and Correlation.
8. Filters and Sampling.

And organized as shown in the digram below of DSP System :

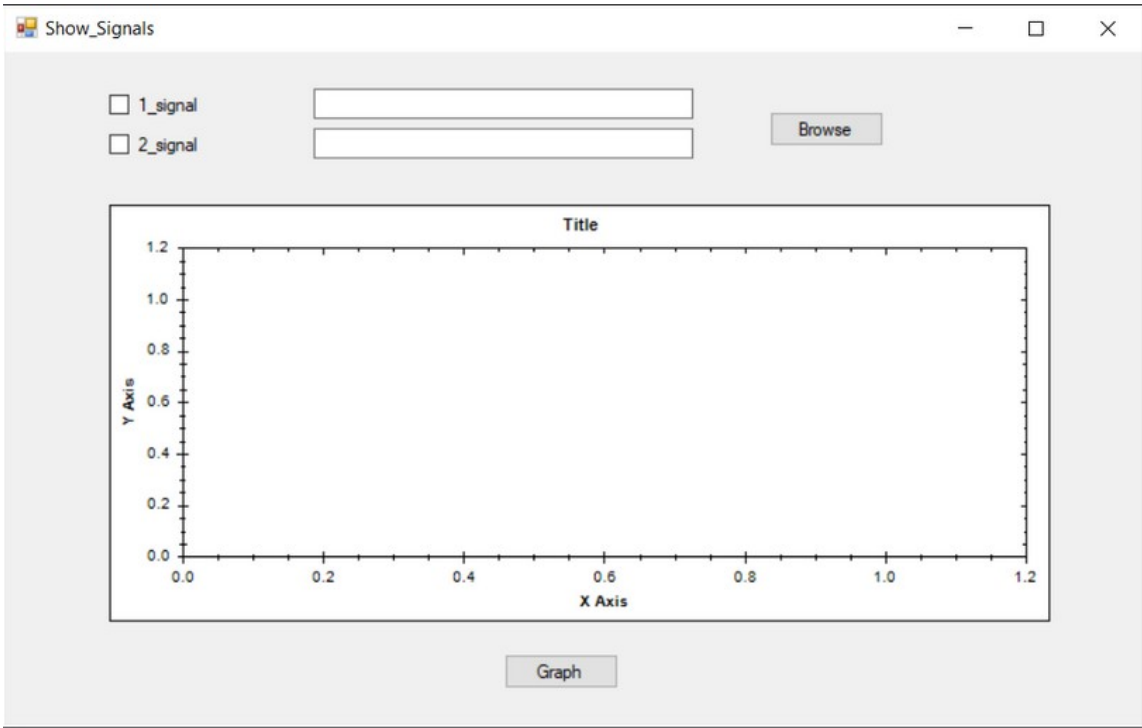


# GUI of System:

## 1. Main Window.



## 2. Signals.



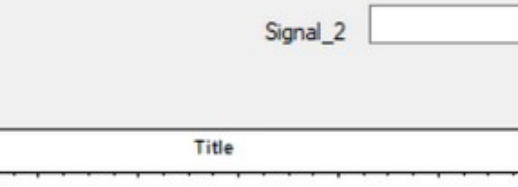
### 3. Quantization and Encoding.

Form3

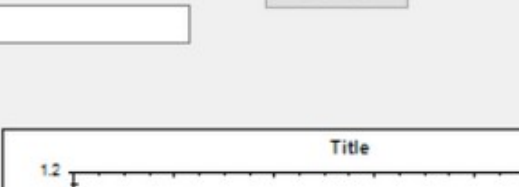
Signal\_1

Signal\_2

2\_Signal



Marge\_Signals



Quantization

☐ #bits

Signal

Number\_Of\_Bits

Quantization\_Level

n	x(n)	Initial_index	xq(n)	enCode	eq(n) = xq(n)
1					

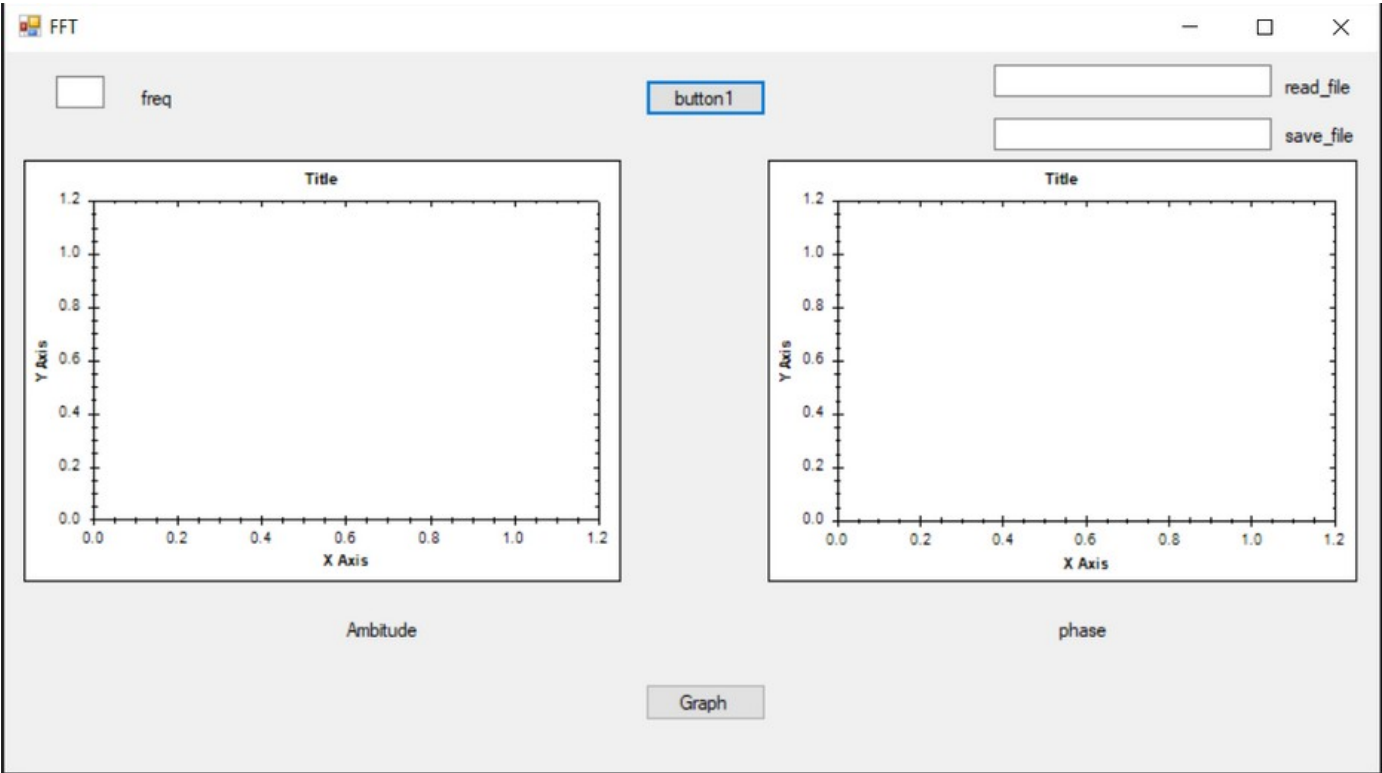
Y Axis

X Axis

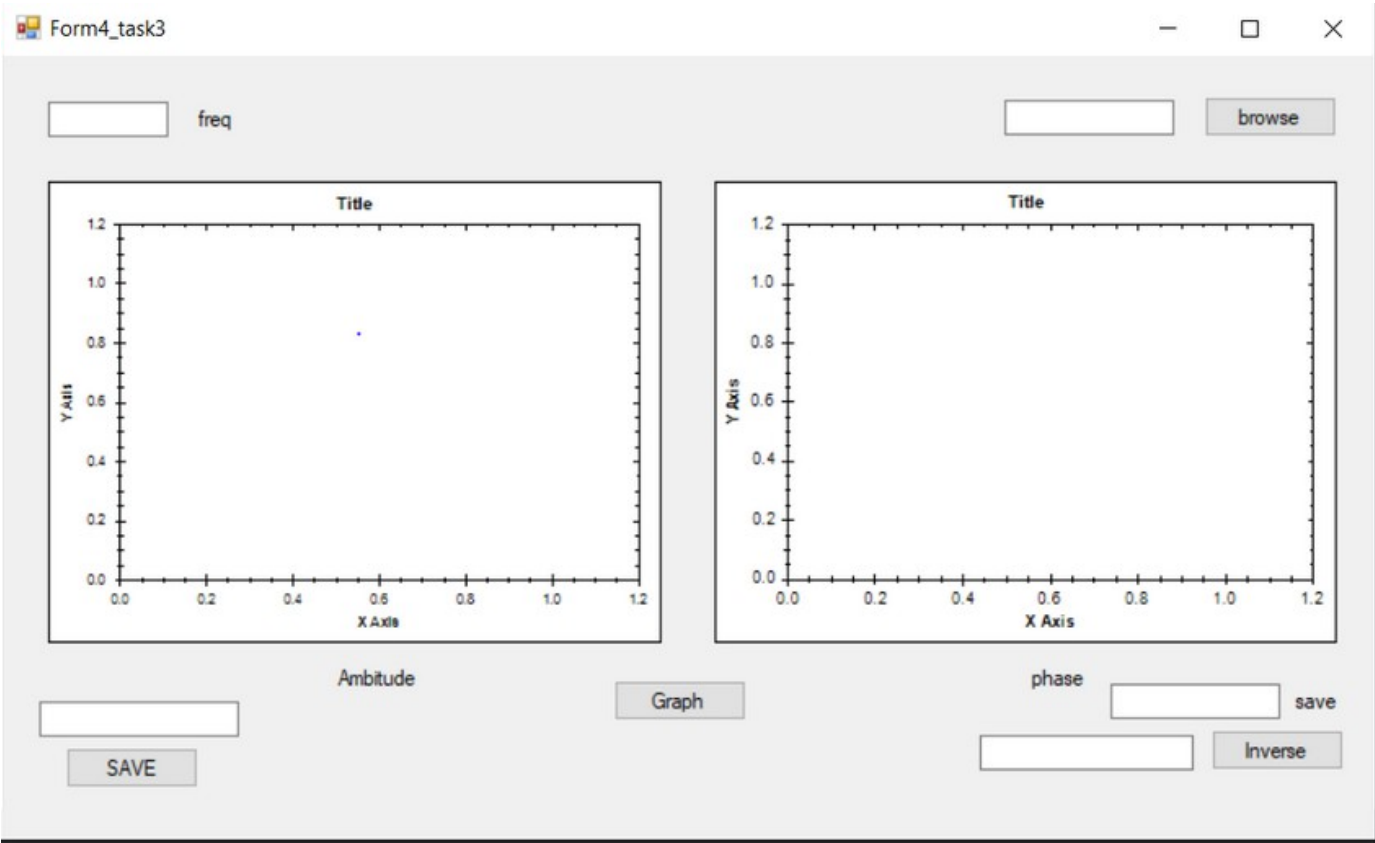
Show\_Data

Graph

4. Discrete Fourier Transform (DFT) and (Inverse DFT).



5. Fast Fourier Transform (FFT).



## 6. Operation On Signals.

Form5

Operation

☐ Signal \* Signal

☐ Const \* Signal

☐ Foalt Signal

☐ Shift Signal

☐ Shift Signal (-n)

Constant

s1

s2

Shift

Title

Y Axis

X Axis

// in final update shift with fold

## 7. Convolution and Correlation.

Form6

Correlation

☐ Auto

☐ Direct

☐ Periodic

☐ Cross

☐ Fast

☐ Non\_periodic

Convolution

☐ Direct

☐ Fast

signal1

signal2

☐ complex

## 8. Filters and Sampling

<i>Filter Type</i>	<i>Sampling</i>
<input type="checkbox"/> LowPass <input type="checkbox"/> HighPass <input type="checkbox"/> BandPass <input type="checkbox"/> BandStop	UP_Sampling N: <input type="text"/>  Down_Sampling N: <input type="text"/>
fc(Cut Of Freq) <input type="text"/>  fc2(Cut Of Freq) <input type="text"/>  <i>Transition Width</i> <input type="text"/>	signal <input type="text"/>
<b>Attinuation</b> <input type="text"/>	
<i>Sampling Freq</i> <input type="text"/>	
<input type="button" value="button1"/>	