1. *FFT* instant spectrum (Documentation - see 01)
1.1. Presentation

FFT instant spectrum represents the spectrum in the frequency domain (Fourier). The proper Fourier discrete spectrum is complex (represented by complex numbers), {a(k) +jb(k)}, k =0,1,…. The power spectrum is of the form {a2(k)+ b2(k)}. The complex amplitude spectrum is the double value row represented by couples a(k); b(k); the absolute amplitude spectrum is the value row represented by the radicals of the previous row values {a2(k)+ b2(k)} in the power spectrum.

The Fourier spectrum is that obtained through method (algorithm) FFT – Fast Fourier Transform. Here the algorithm given in [1] was used. To apply it, a number of samples equal to a power of 2 must be available, for example 16, 32, 64, …, 256, 512, 1024, …. This is the value chosen by the user. If the input value does not equal a power of 2, the superior closest value is automatically computed. For example, if the user specifies 214 values, the number of samples in the analysis window will be 256.

There are differences between the spectrum computed based on a finite number of samples and the real signal spectrum – theoretically computed on an infinite number of samples. The fewer samples, the bigger errors. An additional error is caused by the fact that the using of a finite number of samples is equivalent to the multiplication of the analysed signal by another one called "window signal" of rectangular form. Hence, the found spectrum is in fact that of the product (spectrum convolution) of the two signals.

To minimise errors, the "weighting windows" technique is used. The most common are the Hamming, Hanning, Blackman, Bartlett, Cebyshev, Kaiser, and triangular windows. The first types of windows (including the implicit one – rectangular) were defined in the program.

An experimented user will chose the window he wishes. If no preference, we recommend the Hanning.

The generated spectrum corresponds to the normalised signal (the signal from accessed .wav file is automatically normalised and its continual component is eliminated).

The application generates a text file in which the results are numerically displayed as columns (two columns for the complex representation separated by the *tab*). The user would be able to use these values for a graphic or to process data in another application, for example Excel.

you intend to use the program. In the absence of a written permission, we reserve the right to apply to justice.

1.2. Way of operating

The tool was conceived under the form of an executable called *fft_inst.exe*. This has to be enclosed in the same folder with the sound files (wav) to be analysed. The user selects the following parameters:

- Name of file (it may be selected from the wav file list in the folder). After inputting the sound file name, the information in the header is checked and pieces of information such as sampling frequency, number of channels, number of bits per sample, and the total number of samples are displayed. Only mono-channel sound files are accepted.
- Number of processed samples – dimension of the analysis window (for example *512*)
  The minimum value must be 4 and the maximum one is given by the total number of samples (or the dimension of the memory unit assigned to the vector keeping the read data in the file)
- The place within wav file of the sample on which the analysis window will focus (for example *345* – the program will then select the samples in places 345-512/2 to 345+255)
- Select the type of window
- Select the way in which data will be saved (as complex numbers or in modulus)

References

[1] C Numerical Recipes

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