IMTC 2005 – Instrumentation and Measurement
Technology Conference
Ottawa, Canada, 17-19 may 2005

Virtual Laboratory a Key for Teaching Principles of Digital Signal Processing

Marian Jerzy Korczynski
Andrzej Hetman
Artur Hlobaz
Department of Electrical and Electronic Engineering, Technical University of Lodz
Stefanowskiego 18/22, 90-924 LODZ, Poland
Phone: (48 42) 6312518, Fax: (48 42) 6362281, Email: jerzykor@p.lodz.pl, ahetman@p.lodz.pl, ahlobaz@p.lodz.pl

Abstract – *The Digital Signal Processing is a part of data handling so important in today's metrology. Teaching of digital signal processing is carried not only at the electrical and electronic engineering departments as the most traditional interested in this subject but also others not only technical universities are carried lectures and tutorials. The authors are suggesting how to propose understanding the bases of Digital Signal Processing. The virtual harmonic analyzer, which operates on simulated and real data is a base. The presented virtual analyzer is a tool for analyzing of quality of electrical power from the point of view of harmonic constants.*

The proposal delivered here, tells not only how to tackle Digital Signal Processing, but is a tool to identify problems while developing harmonic analyzers.

Keywords - *DSP, Virtual laboratory, Virtual Instruments.*

I. INTRODUCTION

Digital Signal Processing have two main aspects: (a) data, which must come from the physical, real world, and (b) data processing mainly carried by more or less specialized computer based systems.

The real world is mainly analogue, so before the processing data begins, data must be acquired. Some authors of books on DSP describe that DSP starts when data are converted to suitable convenient files for further data processing. It is possible to agree with such an approach, but still connection to physical world is needed. The aspects related to mentioned parts are covered in a virtual laboratory developed by authors.

II. CONFIGURATION OF THE LABORATORY

The Electrical and Electronic Measurement Laboratory use analogue methods in measuring electrical quantities. The aim of this paper is to present the new control set-up, which is making use of the digital processing of the tested signal. The set-up enables students to learn a specificity of digital measurements using sampling of the input signal.

List area covered by projects:

1. Virtual arbitrary signal generator (harmonics, sub- and inter-harmonics included)
2. Harmonic analyzer with signals coming from electrical power supply to be monitored
3. Anti-aliasing filters and their roles. Simulation of anti-aliasing filters
4. Leakage effect as a result of not completeness of required data in a period
5. Over-sampling and filtering for improving amplitude recognition
6. FIR filters and their influence on amplitudes of harmonics of analyzed signals
7. IIR filters and their influence on amplitudes of harmonics of analyzed signals
8. Square signals and their signal analysis
9. Noise reduction special effects and elimination of interferences in sound files of wave type

The laboratory presenting different data acquisition equipment for further Digital Signal Processing is shown in Fig. 1.

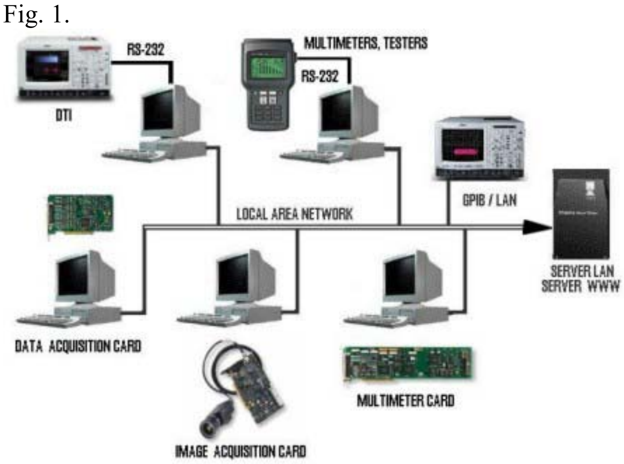

Fig. 1. Configuration of laboratory for acquisition and processing data of real process, mainly laboratory electrical power net is under consideration

Each individual stand include of a PC computer, equipped with different input PCI card, USB based devices, or just instruments and LabView software platform.

III. FACILITIES OF ONE OF THE SET-UP

We shall examine the measurement of RMS value in the most difficult case when neither the value of the first harmonic frequency f_1 nor number of the highest harmonic n_{max} is known. Authors worked out their own method that is presented below. The method requires repeating of sampling 4-times with different values of sampling frequency and number of samples. We start with the greatest sampling frequency, f_{Smax}, that is permissible for the measurement card, DAQC, and with maximal number of samples, M_{max}, which FFT algorithm can transform in a reasonable time. In that way we achieve the maximum width and the best resolution of spectrum. For example we take a measurement of $M_1 = M_{max}$=2048 and $f_{S1} = f_{Smax}$=100 kHz. First, we check on the virtual monitor if there is at least one period of the tested signal in the measurement window otherwise we reduce the sampling frequency. Next, we apply FFT and take notes of the frequency f_1 and the amplitude A_1 of the lowest harmonic as well as the frequency f_{max} and amplitude A_{max} of the highest harmonic as shown in the Table 1. We also note the frequencies f_{1-}, f_{1+} and amplitudes A_{1-}, A_{1+} of adjacent lines in spectrum.

Table 1. Harmonic components of signal applying FFT method

No	-	1	2	3	4
M	-	2048	2048	2048	32
f_S	Hz	100000.00	3030.3015	62.8846	1601.6384
f_{1-}	Hz	0	48.8281	12.8041	0
f_1	Hz	48.8281	50.3077	12.8348	50.0512
f_{1+}	Hz	97.6562	51.7874	12.8656	100.1020
A_{1-}	V	0.3702	2.1050	0.4264	0.0001
A_1	V	10.5783	9.4108	9.9706	9.9999
A_{1+}	V	9.5770	1.3133	0.3887	8.9999
f_{max}	Hz	1513.6700	449.8100	-	450.4610
A_{max}	V	0.3089	1.3868	-	2.0003

Now we calculate the number of the highest harmonic, n_{max1}, number of periods, p_2, and a new sampling frequency, f_{S2}, applying the formulae given below:

$$n_{max1} = \frac{f_{max1}}{f_{1,1}} = \frac{1513.67}{48.8281} = 31.0 \tag{1}$$

$$p_2 = Ent\left(\frac{M_2 - 2}{2n_{max1}}\right) = Ent\left(\frac{2048-2}{2 \cdot 31}\right) = 33 \tag{2}$$

$$f_{S2} = \frac{M_2 f_{1,1}}{p_2} = \frac{2048 \cdot 48.8281}{33} = 3030.3015\,\text{Hz} \tag{3}$$

Then we do FFT again. It was found that, the amplitudes of adjacent lines differ more than 1‰ from that of the amplitude of the first harmonic. It can be argued that measurement should be continued. In order to determine precise the value of frequency f_1 we must increase spectral concentration by applying undersampling. We adjust the sampling frequency only 25% more than frequency $f_{1,2}$

$$f_{S3} = 2\lambda f_{1,2} = 2 \cdot \tfrac{5}{8} \cdot 50.3077 = 1.25 \cdot 50.3077 = 62.8846\,\text{Hz} \tag{4}$$

Undersampling reduces all harmonics in the low frequency range. The value of sampling factor $\lambda = 5/8$ has been chosen by the authors to be the most suitable because, due to aliasing phenomenon, in the neighborhood of the first harmonic 9th, 11th, 19th, 21st et cetera harmonics appear and the amplitudes of these harmonics are small in proportion to the amplitude of the first harmonic. We then apply FFT once more. The measured amplitudes of adjacent lines we utilize to interpolation calculation of the frequency f_{1U}

$$\begin{aligned} f_{1U} &= f_{1-} + \frac{A_{1+}}{A_{1-} + A_{1+}}\left(f_{1+} - f_{1-}\right) = \\ &= 12.8041 + \frac{0.3887}{0.4264 + 0.3887}\left(12.8656 - 12.8041\right) \\ &= 12.8334\,\text{Hz} \end{aligned} \tag{5}$$

The real value of the frequency $f_{1,3}$ we achieve from formula

$$f_{1,3} = f_{S3} - f_{1U} = 62.8846 - 12.8334 = 50.0512\,\text{Hz} \tag{6}$$

To the last FFT we apply values according to formulae as follows

$$n_{max\,3} = \frac{f_{max\,2}}{f_{1,3}} = \frac{449.810}{50.0512} = 8.987 \approx 9 \tag{7}$$

$$M_4 \geq 2\left(n_{max\,3} + 1\right) = 2(9+1) = 20 \Rightarrow M_4 = 32 = 2^5 \tag{8}$$

$$p_4 = Ent\left(\frac{M_4 - 2}{2n_{max\,3}}\right) = Ent\left(\frac{32-2}{2 \cdot 9}\right) = 1 \tag{9}$$

$$f_{S4} = \frac{M_4 f_{1,3}}{p_4} = \frac{32 \cdot 50.0512}{1} = 1601.6384\,\text{Hz} \tag{10}$$

The results of the last FFT show reaching sufficient accuracy of the measurement: amplitude A_{1-} is less that 10^{-3} A_1.

The real parameters of the examined signal had been $A_1 = 10.0000$ V, $f_1 = 50.0510$ Hz, $n_{max} = 9$, $A_9 = 2.0000$ V, $f_9 = 450.459$ Hz. The measured values are the same with 5-digit accuracy.

Finally, on the base of the obtained exact amplitudes of individual harmonics, we can get the RMS value of voltage

$$V = \sqrt{A_0^2 + \tfrac{1}{2}\sum_{n=1}^{N} A_n^2} = 13.89\,\text{V} \tag{11}$$

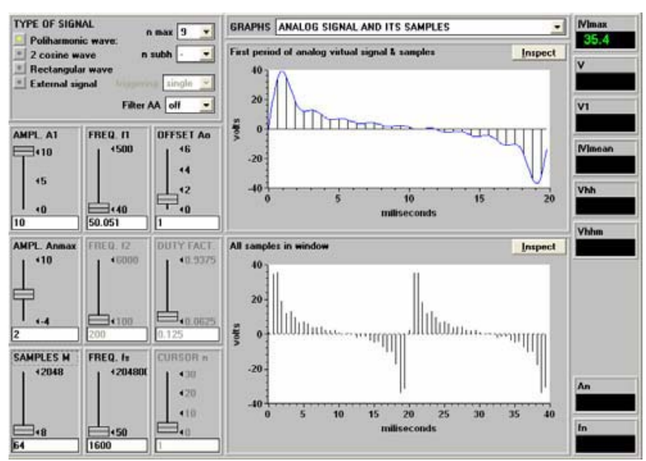

Fig. 2. Front panel of the virtual spectrum analyzer: 9 harmonics generated, basic harmonic of 50.051 Hz, number of samples 64 and sampling frequency of 1600 Hz, which is not adequate for proper analysis of the spectrum

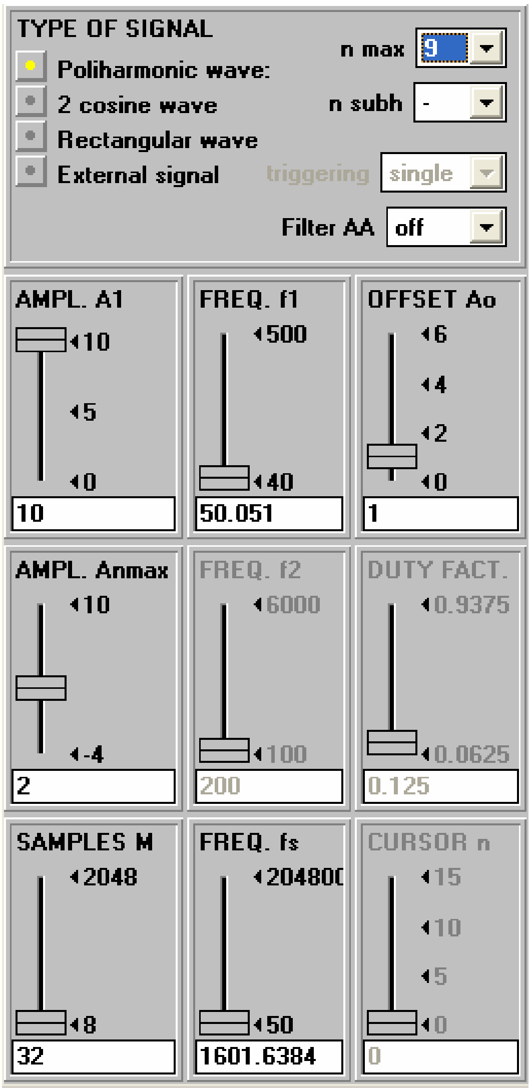

Fig. 3. Part of front panel of virtual spectrum analyzer – communication interface for setting up parameters. Properly adjusted sampling frequency.

The formulae depicted below were applied to analysis of sampling parameters as follows:

A. Number of samples

Formula: $M = 2^{Int} \geq M_{\min} = 2(n_{\max} + 1)$

minimum value $2(n_{\max} + 1)$

maximum value $M_{\max}$

B. Number of spectrum components

Formula: $N = \frac{1}{2} M - 1 \geq n_{\max}$

minimum value $n_{\max}$

maximum value $\frac{1}{2} M_{\max} - 1$

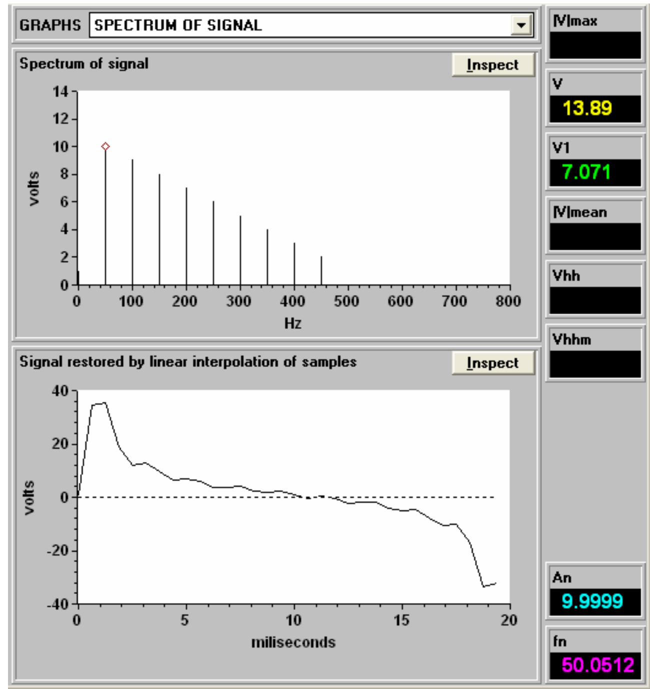

Fig. 4. Part of front panel of virtual spectrum analyzer – displaying spectrum and restored signal by linear interpolation of samples. Properly adjusted sampling frequency.

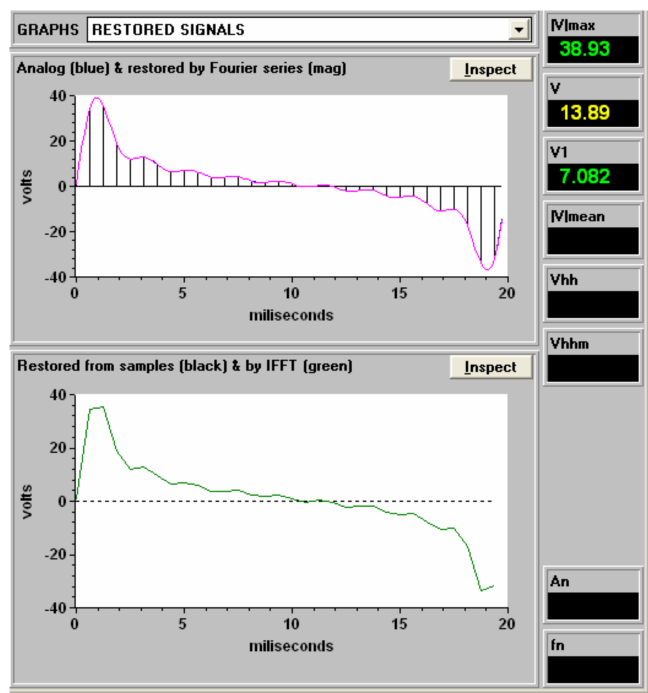

Fig. 5. Part of front panel of virtual spectrum analyzer – displaying analogue signal (in blue) and restored by Fourier series on the top, but both curves are identical in this case, bottom - restored signal by linear interpolation of samples and by Inverse Fast Fourier Transform also identical in this case. Sampling frequency adjusted properly.

C. Number of periods in window

Formula: $p = Ent\left(\frac{N}{n_{max}}\right) = \frac{f_1}{f_w} = \frac{M f_1}{f_s}$

minimum value – equals 1

maximum value $Ent\left(\frac{M-2}{2n_{max}}\right)$

D. Sampling frequency

Formula: $f_S = \frac{M f_1}{p} = M f_W \ge \frac{M}{N} f_{max}$

minimum value $\xrightarrow[M>>M_{min}]{} \frac{2M}{M-2} f_{max}$

maximum value $M f_1$

E. Resolution of spectrum

Formula: $f_W = \frac{f_s}{M} = \frac{f_1}{p}$

minimum value $\xrightarrow[M>>M_{min}]{} \frac{2 f_{max}}{M-2}$

maximum value f_1

Tab. 2 Influence of sampling frequency f_S and number of samples M on resolution of spectrum f_W

$f_W = f_S/M$							
f_S	M=8	M=16	M=32	M=64	M=128	M=256	M=512
400	50	25	12.5	6.25	3.125	1.5625	0.78125
800	100	50	25	12.5	6.25	3.125	1.5625
1600	200	100	50	25	12.5	6.25	3.125
3200	400	200	100	50	25	12.5	6.25
6400	800	400	200	100	50	25	12.5
12800	1600	800	400	200	100	50	25
25600	3200	1600	800	400	200	100	50
51200	6400	3200	1600	800	400	200	100

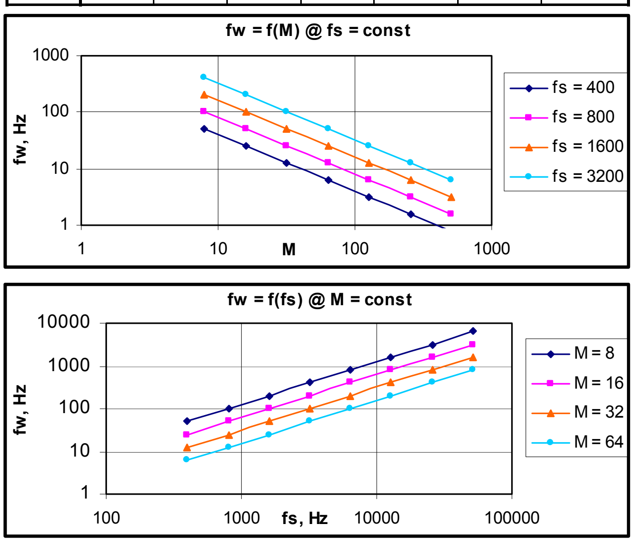

Fig. 6. Resolution of spectrum vs. number of samples M (upper) and sampling frequency (lower)

F. Width of spectrum

Formula: $f_B = N f_W = \frac{N}{M} f_S = \frac{N}{p} f_1 \ge f_{max}$

minimum value f_{max}

maximum value $\frac{M-2}{2} f_1$

Tab. 3 Influence of sampling frequency fS and number of samples M on width of spectrum f_B

$f_B = (1-2/M)*f_S/2$							
f_S	M=8	M=16	M=32	M=64	M=128	M=256	M=512
400	150	175	187.5	193.75	196.875	198.438	199.219
800	300	350	375	387.5	393.75	396.875	398.438
1600	600	700	750	775	787.5	793.75	796.875
3200	1200	1400	1500	1550	1575	1587.5	1593.75
6400	2400	2800	3000	3100	3150	3175	3187.5
12800	4800	5600	6000	6200	6300	6350	6375
25600	9600	11200	12000	12400	12600	12700	12750
51200	19200	22400	24000	24800	25200	25400	25500

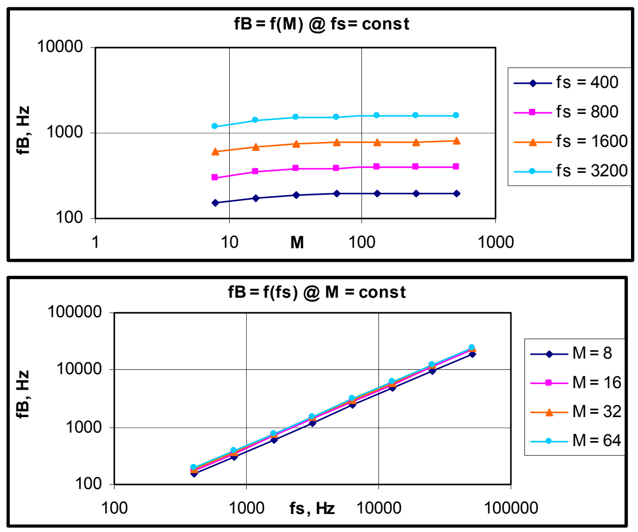

Fig. 7. With of spectrum vs. number of samples M (upper) and sampling frequency (lower)

G. Number of measurable harmonics

Formula: $n_{MH} = Ent\left(\frac{f_B}{f_1}\right) = Ent\left(\frac{N f_s}{M f_1}\right)$

minimum value n_{max}

maximum value $\frac{M-2}{2}$

H. Oversampling factor

Formula: $\lambda = \frac{f_s}{2 f_{max}} = \frac{M}{2 p n_{max}}$

minimum value $\xrightarrow[M>>M_{min}]{} \frac{M}{M-2}$

maximum value $\frac{M}{2n_{max}}$

Tab. 4 Influence of sampling frequency f_S and number of samples M on number of spectrum components N.

$N = M/2$-1							
f_S	M=8	M=16	M=32	M=64	M=128	M=256	M=512
400	3	7	15	31	63	127	255
800	3	7	15	31	63	127	255
1600	3	7	15	31	63	127	255
3200	3	7	15	31	63	127	255
6400	3	7	15	31	63	127	255
12800	3	7	15	31	63	127	255
25600	3	7	15	31	63	127	255
51200	3	7	15	31	63	127	255

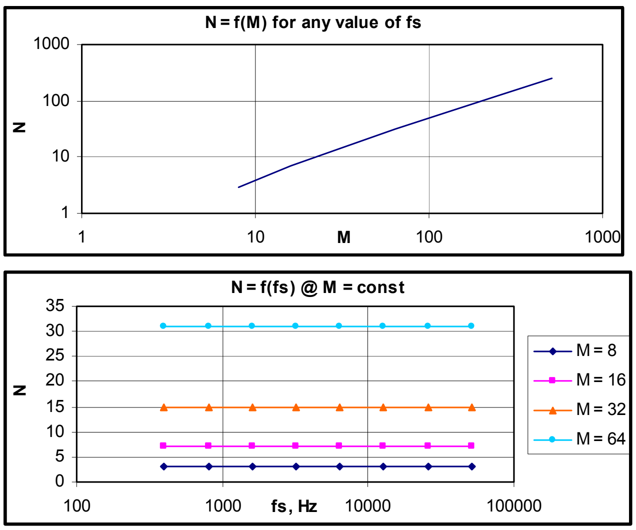

Fig.8. Number of spectrum components vs. number of samples M (upper) and sampling frequency (lower)

The virtual harmonic analyzer is used also to present how to avoid aliasing effect by over-sampling, anti-aliasing filter (simulated) application and both methods together. The two Figures below are presenting: the first - spectrum of rectangular wave and the second - anti-aliasing filter's characteristic.

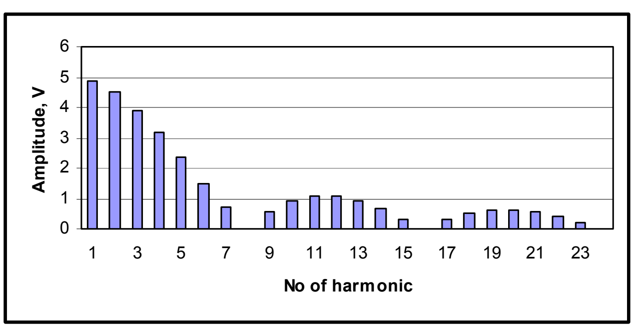

Fig.9. Spectrum of rectangular wave of duty factor 1/8, the first 24 harmonics presented here

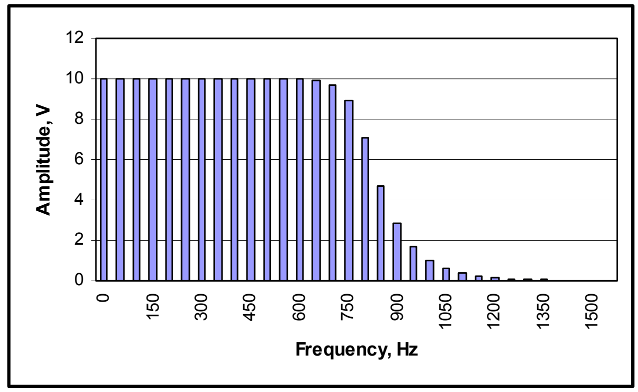

Fig. 10. Anti-aliasing filter characteristic used for observation of 8 spectrum components of the whole spectrum in Fig 9.

IV. CONCLUSIONS

The set-up is used to teach students DSP bases and as a regular spectrum analyzer, which accepts simulated and real signals. It is a nice tool for engineers and harmonic developers of harmonic analyzer

The very important feature of the elaborated set-up is the friendly used operation. The students familiarize with the operation very fast and presented examples well present the theory of DSP.

REFERENCES

[1] Fuerst W., Lynn P. A., Digital Signal Processing with Computer Applications, John Willey & Sons, 1999, ISBN 0-471-97631-8

[2] McGhee J., Kulesza W., Korczynski M. J., Henderson I., Teaching Measurement and Instrumentation in English at the International Faculty of Engineering, 9th International Symposium on Electrical measurement in industry, IMEKO TC-4 Technical Committee on Measurement of Electrical Quantities, Glasgow, Scotland, September 8-9, 1997

[3] McGhee J., Kulesza W., Korczynski M. J., Henderson I., Scientific Metrology, ACGM LODART, Lodz, Poland, first addition, September, 1996, reprint, July, 1998, ISBN83 904299 9 3

[4] McGhee J., Kulesza W., Korczynski. M. J., Henderson I., Measurement Data Handling, ACGM LODART, Lodz, Poland, first addition, 2001, ISBN83-7283-007-x

[5] Mulgrew B., Grant P., Thompson J., Digital Signal Processing, Macmillian Press Ltd, London, 1999, ISBN 0-333-74531-0

[6] Oppenheim A. V., Schafer R. W., Discrete-time Signal Processing, Prentice-Hall, Englewood Cliffs, NJ, 1989

[7] Penarrocha V. M. R., Bataller M. F., Virtual Instrumentation: First Step Towards a Virtual Laboratory, International Workshop on Virtual and Intelligent Measurement Systems, Annapolis, 29-30 April 2000, Annapolis, MD, USA, p. 52-55