Real time analysis of spectrum of supply current with utilization of full digital system

Peter DRGONA*, Michal FRIVALDSKY*, Pavol SPANIK*, Jozef SEDO*

and

Jozef KUCHTA**

* University of Zilina, Department of mechatronics and electronics,
Zilina, 01026, Slovakia

** EVPU (Electronic research and testing institute),
Nova Dubnica, 01851, Slovakia

ABSTRACT

This paper deals with construction of device for online, real-time analysis of recuperative current with the use of microprocessor and thus will be representing full digital system for real time analysis. The proposed system should be utilized for railway transport for the measurement of recuperative currents which is being supplied from power converter of traction vehicle. In the first part of paper, the current sensor and digital filter are described. Then the computation algorithm is being presented. Last part of paper shows implementation in microprocessor and experimental verification of proposed solution.

Keywords: real-time measurement, spectrum, full digital, FFT.

1.INTRODUCTION

Nowadays, a high frequency power semiconductor converters are used in all areas including industry and transportation, for control the energy flow. Their main advantage is high efficient power conversion, high frequency and small size. On the other side, converters uses semiconductor devices in high frequency switching mode, so they represent nonlinear load for distribution network and supply current for converters is not harmonic. As a result, the higher harmonics of supply current are generated, which significantly affects the EMC of converter and other electronic devices. Emissions of higher harmonics are strictly set by certain standards. For railway applications, these limits are set by standard UIC 550-3 (European Union).

Without utilization of power factor corrector (PFC) the supply current of converter acts as current with high portion of non-harmonic elements. Even PFC is being used the given amount of these non-harmonic elements are still being taken away from distribution network. This problem can be also registered during recuperative braking of traction vehicles. As a results of described behavior of power semiconductor systems, the higher harmonics of supplying current are significantly affecting EMC of the semiconductor converter and other electronic devices.

This paper is presenting novel full digital control methodology for higher order frequency spectrum identification provided online, in the real time. Core of proposed system is based on modern microprocessor unit from TMS320F Family from Texas Instruments. Other additional measurement equipment and selection of most suitable additional features will be consequently described. Last part of paper is given for implementation and verification through various experiments.

2. CURRENT SENSING

This chapter describes selection of the current sensing device/method which is suited in final system. In principle there exist three general methods for current sensing:

a) utilization of shunt resistor

b) utilization of current transformer

c) measurement through Hall sensor

Shunt resistor is able to be used for various amplitudes and shape of current waveforms. The output signal for further processing is voltage which is proportional to flowing current. The main criterions for shunt resistors are very low resistance value (in order to meet acceptable values of voltage which has to be processed) and power dimensioning according to flowing current. Base on worldwide standards the most of shunts are designed to have voltage drop (sensing voltage) 50mV, 75 mV or 100 mV at the nominal current. Disadvantages related to usage are temperature stresses which in extreme condition can expressively change value of resistance and thus value of sensing voltage, what can lead in measurement failure and thus collapse of total system. Also at the measurement of time variable waveforms of current, the proposed shunt resistor must be made from material which acts as low inductance due to suppression of measurement distortion.

Therefore more suitable methodology is utilization of current transformer. It transforms flowing current to corresponding voltage, which is being measured by sensing device. Requirement according current transformer is that it cannot operate in no load condition. Also there are other disadvantages according magnetic saturation and power dimensioning, what makes this solution for high power application (what traction vehicle introduces) non attractive due to large volume.

The last mentioned proposal according current measurement is base on well - know methodology, which is based on utilization of Hall sensor. In principle there exist two possible ways:

- measurement in open loop
- measurement in closed loop

In our proposed measuring device we have focused on second way, measurement in closed loop, during which importance has to be given mainly for operation of hall sensor in linear region. Closed loop operation is characterized by total compensation of magnetic flux around magnetic circuit. Each deviation from zero offset leads to generation of Hall effect. Electrical circuit then supplies secondary current, which is necessary for compensation of magnetic field. Advantages of closed loop operation are high accuracy, large available bandwidth, fast response, perfect linearity, possibility of overloading.

Digital filters

Due to fact, that hall sensor is equipped with current output, the transformation from current to acceptable voltage level for A/D converter (microprocessor) is necessary. A/D converter is operating with input voltage level between 0 VDC - 3 VDC.

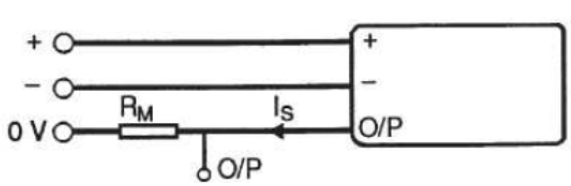

Fig. 1 Principle schematics of Hall sensor connection with measuring resistor

Fig. 2. shows schematics for Hall sensor connection with measuring resistor which is suited for current transformation into voltage. Determination of measuring resistor R_M was done using next formula:

$$R_M = \frac{U_{AD\max}}{I_{pp\_o\max}} = \frac{U_{AD\max}}{\frac{1}{n} I_{pp\_i\max}} = \frac{3}{\frac{1}{1000}300} = 10\Omega \qquad (1)$$

Hall sensor has maximum measuring range ± 150 A, so the maximum amplitude of measured current is 300A. Due to this value the voltage drop on measuring resistor will have to be in the range of ± 1.5 V, but for A/D converter the conversion of negative voltages is not possible. For the purpose of the voltage level adjustment on the measuring resistor without loss of information of measured signal the utilization of next circuit (fig. 3.) was necessary.

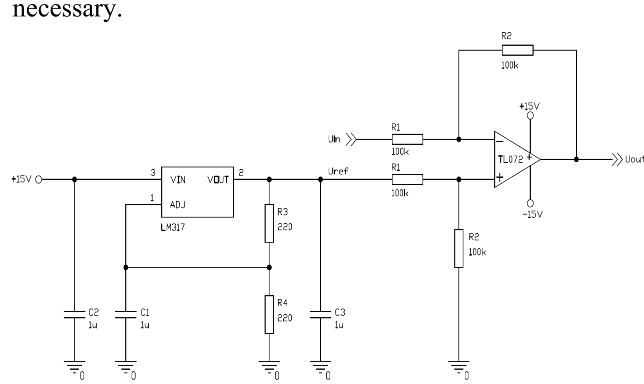

Fig. 2 Block scheme of circuit for voltage adjustment

This circuit shifts input voltage ± 1.5 V to range from 0 to 3V, without modification of measured signal. This is very important, because even small deviation in frequency of the signal can lead to inaccurate analysis. For the output voltage, next formula is valid:

$$U_{out} = \frac{R_2}{R_1}(U_{ref} - U_{in}) \qquad (2)$$

, where U_{ref} is voltage at the output of voltage stabilisator LM317, and U_{in} is voltage across sensing resistor R_M. If $R_1 = R_2$ then:

$$U_{out} = U_{ref} - U_{in} \qquad (3)$$

In such way the output voltage U_{out} is modified voltage in the range 0 V - 3 Vdc which is supplied into A/D converter.

3. FILTERING OF INPUT SINGAL

The Hall sensor with closed loop was used in proposed system. This sensor is capable to sensing AC, DC or impulse currents. Due to use of AD converter in microprocessor, the voltage level from current sensor needs to be shifted in range 0 to 3V.

Digital filters

Digital filter as representation of digital system, process digital input and after processing, at the output of the system, digital signal appears. The role of the digital filter is to affect the spectrum of the input signal in desired form. This means either select a part of the spectrum which will stay unchanged and suppress the rest. (low pass filter, high pass filter, band pass or band stop, or shaping the frequency response. [1] [2]

For given application, a digital IIR filter was chosen, due to its small time delay during processing of input signal (measured current). Another advantage of this type of filter is small memory usage. Unlike FIR filters, stability of IIR filters is not always guaranteed. After the proposal, check whether all poles of transfers function lies inside unit circle in z-domain. Another disadvantage of IIR filters is sensitivity to saturation of processor's arithmetic and sensitivity to quantization of values. Advantages of IIR filters are lower degree of transfer function at the same requirements. Consequently, memory demands for accumulation of coefficients and state variables are lower and also, time for processing of input variable is lower also. General transfer function of IIR filter is in form:

$$H(z)=\frac{Y(z)}{X(z)}=\frac{\sum_{i=0}^{M} a_i \cdot z^i}{\sum_{i=0}^{M} b_i \cdot z^i} \tag{4}$$

In final it is possible to use design tables of analog filters, because each digital filter can be described by its analog equivalent.

Main criterion for filter design is its stability, that means that all poles of transfer function must belong to internal part of unity circle line (see fig. 3).

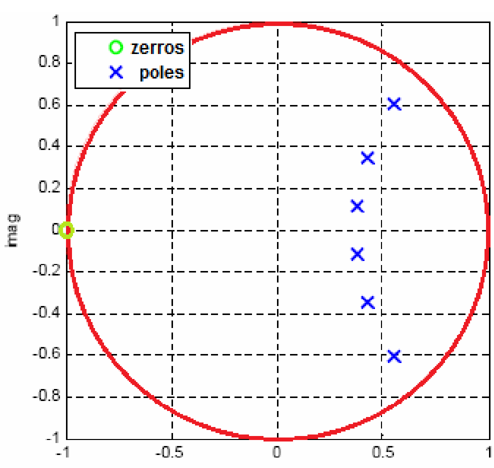

Fig.3. Distribution of poles and zeros of proposed system

For the analysis of spectrum of supply current we have decided to assign antialiasing function for proposed digital filter, what means that it will behave as low-pass filter. For the sampling frequency of f_s = 192307,69 Hz we design filter with cutoff frequency f_c = 26 kHz. Digital filter in such way won't change amplitude frequency spectrum in the range from 0 up to 26 kHz. During filter design we have utilized methodology of analog to digital transformation. First we have researched analog low-pass which was consequently through the use of bilinear transformation transformed into p and z plain. Next, with the use of tables for design of Butterworth filter we have selected analog filter whose low-pass was 6th grade polynomial function. This proposed filter is giving compromise between fastness and slope of transition zone of modulated frequency characteristic. Analog form of proposed filter is given by next equation:

$$H(p)=\frac{1}{p^6+3{,}863703\,p^5+7{,}464102\,p^4+9{,}141620\,p^3+7{,}464102\,p^2+3{,}863703\,p+1} \tag{5}$$

Using mentioned tables of Butterworth filter we get equation (5) in digital form what is given by next equation:

$$H(z)=\frac{b_0+b_1z^{-1}+b_2z^{-2}+b_3z^{-3}+b_4z^{-4}+b_5z^{-5}+b_6z^{-6}}{1+a_1z^{-1}+a_2z^{-2}+a_3z^{-3}+a_4z^{-4}+a_5z^{-5}+a_6z^{-6}} \tag{6}$$

where the coefficients of polynoms are:

b_0 = 0,001552195356958 b_1 = 0,009313172141747
b_2 = 0,023282930354368 b_3 = 0,031043907139157
b_4 = 0,023282930354368 b_5 = 0,009313172141747
b_6 = 0,001552195356958 a_1 = -2,733967214763194
a_2 = 3,600856207068566 a_3 = -2,715090353346278
a_4 = 1,219380058434311 a_5 = -0,304731364108709
a_6 = 0,0032893169560607

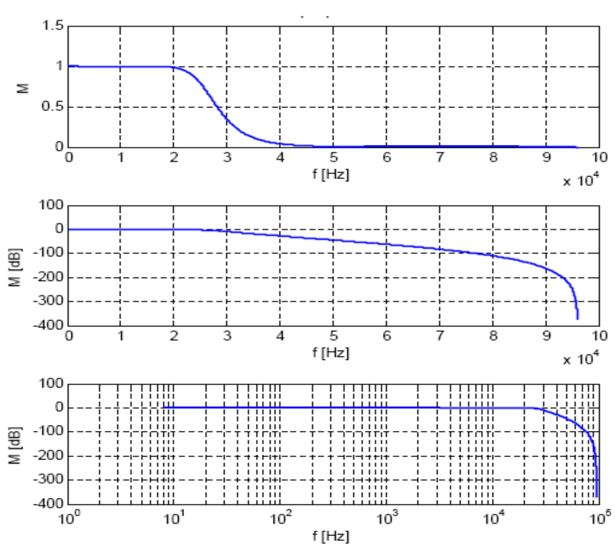

Fig.4 Frequency characteristic of proposed filter

Fig.4 is showing modulated frequency characteristic of proposed digital filter. At the cutoff frequency the attenuation is around -3dB. This confirms proper design.

4. COMPUTATION OF DISCRETE FOURIER TRANSFORM AND AMPLITUDE SPECTRUM

For computing of discrete Fourier transform the FFT algorithm with "in place computing" was used. In program, two fields each with 512 samples were used. Both fields were in 16b floating point form. One of the field was for real part and second was used for imaginary part of complex number. For realization of FFT function, standard function-void fft(float32*p_real, float32*p_imag) was used. Input to the function is array of real numbers, but computing of imaginary numbers is also possible. Generally, output from the discrete Fourier transform is sequence of complex numbers. From above mentioned function, the output is array of real parts of numbers and array of imaginary numbers. Hence, the input values are overwritten with output values. Advantage of this algorithm is low memory usage [3] [4].

For amplitude spectrum computing, function „void Spectrum_Amp(float32 *p_real, float32 *p_imag)" was used. this function reconstruct the frequency axis and compute amplitudes of each harmonic component by equation:

$$A[n] = \frac{2}{N}\sqrt{X_{real}^2[n] + X_{imag}^2[n]} \tag{7}$$

where:

n - is n-th component of Fourrier transform

X_{real} - is real part

X_{imag} - is imaginary part

Function, which resolution is determined by number of samples N, assign the value of amplitude to each frequency from frequency range, whereby identify the spectrum of analyzed signal, in our case the current from converter.

5. Implementation into microprocessor

Algorithms for filtering, computing and analysis of measured signal are realized in software form in microprocessor. For this purpose a 32bit float-point microprocessor TMS 320F28335 Delfino. This 150MHz microprocessor contains float-point C2000 core and powerful peripherals, from which a 12bit A/D converter was used for measurement of signal from Hall sensor. Software for computing and analysis was written in standard C-language. Maximum frequency of analyzed signal is up to 20kHz, so the sampling frequency of 192kHz was chosen, which means, that Shannon criteria was observed

Implementation of digital filters

Digital filter in device for real time measurement and analysis of recuperative currents from electric locomotive acts like antialiasing filter, which is basically low pass filter. For sampling frequency f_s=192307.69 Hz, a low pass filter with cut-off frequency f_c=26kHz was designed. Digital filter do not affect amplitude frequency spectrum from 0 to 26kHz, but all higher frequencies will be damped.

Filter was designed using analog to digital transformation method or design by emulation method. First, analog low pass filter was designed and consequently, this filter was transformed to discrete z-domain by bilinear transform. By use of tabs for design of Butterworth filter, an analog low-pass filter of 6-th degree was chosen. Degree of this filter gives good compromise between computing speed and rate of Bode characteristics. Transfer function of analog filter is in form:

$$H(s) = \frac{1}{s^6 + a_1 \cdot s^5 + a_2 \cdot s^4 + a_3 \cdot s^3 + a_4 \cdot s^2 + a_5 \cdot s + a_6} \tag{5}$$

This function was transformed by bilinear transform to z-domain. This two steps can also be performed in reverse. For implementation in microprocessor, transfer function in z-domain must be transferred into time domain. After transformation we get two differential equations, state and output. These two differential equations can be easily implemented into microprocessor.

Realization of window function

Because of analyzed signal consists of combination of sinusoidal waveforms and duration of transient effects is longer than window, a Hanning window was chosen. This window has good frequency resolution and limited spectral leakage. Realization of the window function consists in multiplication of the window value, with the value of the sample. This is performed by function „void Okno_Hann(float32 *p_pole)".

Computing of discrete Fourier transform

For computing of discrete Fourier transform the FFT algorithm with "in place computing" was used. In program, two fields each with 512 samples were used. Both fields were in 16b floating point form. One of the field was for real part and second was used for imaginary part of complex number. For realization of FFT function, standard function - void fft(float32*p_real, float32*p_imag) was used. Input to the function is array of real numbers, but computing of imaginary numbers is also possible. Generally, output from the discrete Fourier transform is sequence of complex numbers. From above mentioned function, the output is array of real parts of numbers and array of imaginary numbers. Hence, the input values are overwritten with output values. Advantage of this algorithm is in low memory usage and relatively short time necessary for computing of this function. For application in real time, computing time of algorithms has big impact on performance of whole device. Time necessary for computing of all algorithms was 6.8μs, whereby single precision 32b floating point operands were used for variables.

5. SYSTEM FOR REAL TIME ANALYSIS OF CURRENT'S SPECTRUM

Our proposal of system for real time analysis of current's spectrum is primarily indented for use in railway applications. But this system can be normally used in every application where exact measurement of recuperative current is necessity. As we previously mentioned, according to standard UIC 550-3, the recuperative current which is flowing through rail back into power network cannot affect signal devices and other equipment which could be potentially safety risk for railway system.

Whole analysis of recuperative current must be done in real time. It means that if analyzed current contains specific harmonic content, or if content of higher harmonics is above allowable limit, then the proposed system has to perform certain measurements. Control system and control algorithm will prevent distorted current from flowing back into the distribution point and power network.

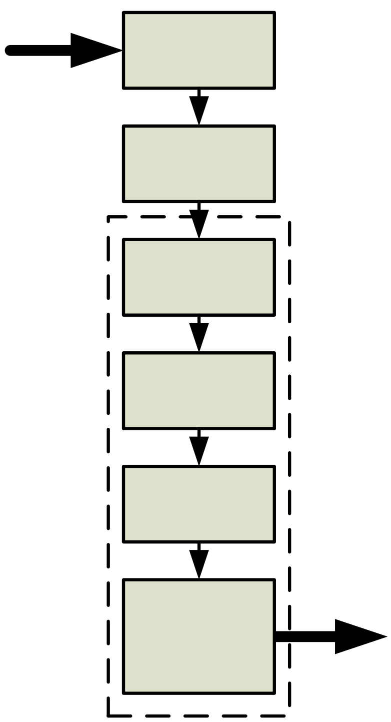

Fig. 5. Block scheme of system for analysis of recuperative currents

Fig. 5 shows block scheme of proposed system. It consists from current sensing device, and from microprocessor (in our case we have been working with TMS320F28335). Analog to digital conversion and all functions of filters and computational algorithms which are necessary for measurement and analysis of harmonic content in recuperative currents are in digital - in software form.

6. SIMULATION AND EXPERIMENTAL VERIFCATION

For verification of proposed algorithm - digital filter and its accuracy, the simulation in MATLAB environment was made. Input to the simulation was signal, which contains 20 harmonic functions. Each function has amplitude equals to 1 and frequencies were in range from 5kHz to 100kHz. Sampling frequency was 192kHz

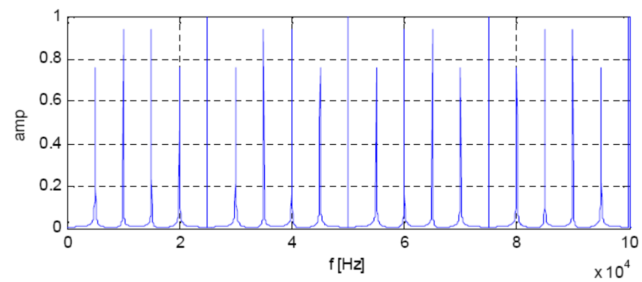

Fig.6. Amplitude spectrum of original input signal

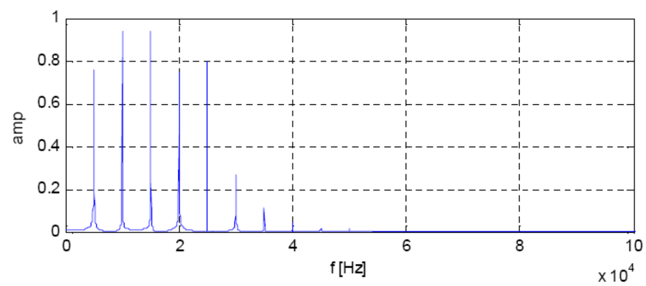

Fig.7. Amplitude spectrum of filtered signal

Fig. 7. shows amplitude spectrum of filtered signal. From figure it is clear to say, that frequencies of each harmonic component are clearly identified, but due to spectral leakage, some frequencies are still present also above range of 26kHz. But this problem has only minor impact on application of real time measuring device. Results from simulations also show good accordance with standard MATLAB function fft().

Experimental verification

For verification of whole system, measurements and tests of sensor and digital filter were made. On fig.8 the waveforms of analyzed signal are shown, where upper wave is related to current and bottom wave is measured voltage. On fig.9 the spectrum of this signal is being showed, whereby it was computed through the use of MATH function which is given by measuring oscilloscope. From this figure, the basic harmonic content (frequency around 50 Hz) of measured signal is able to be clearly identified. It serves for comparison with our proposed system, whose real time spectrum analysis is shown on fig.10.

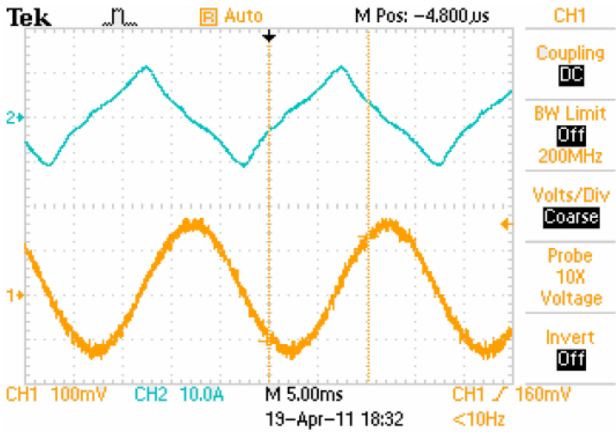

Fig.8. Signal waveforms used for verification of proposed system (up-current, down-voltage)

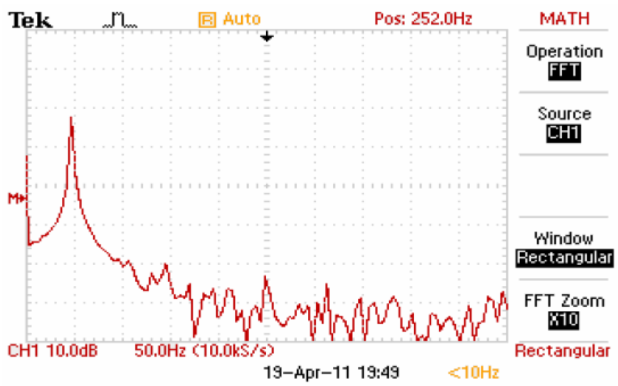

Fig.9. Amplitude spectrum measured on sensing resistor (computed by oscilloscope)

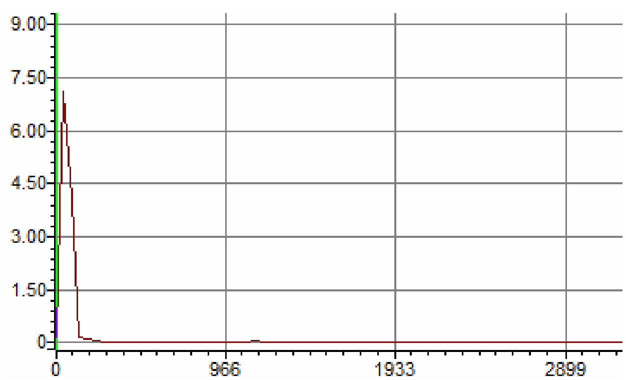

Fig.10. Amplitude spectrum computed by proposed system

From fig. 10 can be seen that proposed measuring system is acting correctly and that it analyzed measured current waveform in expected manner. There exist some problems according accuracy, what was caused by utilization of low-performance sensing resistor and microprocessor unit.

5.CONCLUSION

In this paper the system for analyzing of spectrum of supply current is proposed. All system, except the current sensor is in digital form. Proposed filter and computing algorithm shows good results for application as system for monitoring the recuperative currents from converter in electric locomotive. Performance and resolution is limited by used microprocessor, so for other application the more powerful processor should by used.

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