XVII IMEKO World Congress
Metrology in the 3rd Millennium
June 22–27, 2003, Dubrovnik, Croatia

ACTIVE DISTORTION REDUCTION OF POWER SOURCES

Balázs Vargha, László Sujbert

Department of Measurement and Information Systems
Budapest University of Technology and Economics, Budapest, Hungary

Abstract – Quality of power sources is mainly increased by using better components and tricky electronic circuits. The paper introduces a control loop, which is similar to that used for the active cancellation of acoustic noise signals. In the feedback loop, the distortions of the signal are estimated by a combination of an adaptive Fourier analyzer and a resonator-based observer. Only the distortions should be compensated, therefore the actuator signal has low power. The paper introduces the active distortion reduction system and presents practical results, as well.

Keywords: adaptive Fourier analyzer, distortion reduction, resonators

1. INTRODUCTION

The objective of the research is to reduce the harmonic distortions of power sources. In practice, a generator is a non-linear system, therefore the output of such a system may contain distortions. Manufacturers try to reduce the harmonic content of the output signal by the use of more precise components, which behave more linearly.

Two different approaches can be used. The first off-line method assumes the use of an arbitrary waveform generator and a digitizer. The algorithm tries to produce a signal with a given spectrum on the load. The digitizer measures the signal on the load, and by knowing the error of the desired and the realized spectra on the load, the digital input of the waveform generator is modified in order to decrease the error. By making several attempts, the algorithm converges to produce the required spectra [1]. This general solution can be also used to generate a sine wave with small distortions. In this case, the desired spectrum is the theoretical spectrum of a sine wave. The method is not adaptive, it is valid only for the setup used during the error minimization algorithm. A new setup calls for a new compensation.

The second approach, which will be discussed in this paper in detail, is an active method. A resonator-based observer determines the harmonic content of the signal, and the distortions are subtracted from the output signal. It can be assumed that the harmonic content of the output signal is a relatively small part of the total power of the signal. In this case, even if the signal power is quite large (e.g. the mains), the power of the correction signal is quite small. For example assuming a signal with 1 kW power and having 10% distortion, the harmonic reduction can be achieved with a 10 W correction signal.

The presented method is mainly based on previous results achieved in periodic noise control. The realized system was able to cancel acoustic noise [4].

2. RESONATOR-BASED OBSERVER

If a valid model of the system is available, it is sometimes possible to use the so-called prediction-correction scheme. The known model of the system will be included in the observer. The design of the measurement procedure is the determination of the right correction strategy. The input of the correction strategy is the error caused by the different state of the two systems. The correction strategy is used to reduce the effect of noise, in the ideal case (when there is no noise), it can be left out (Fig. 1.).

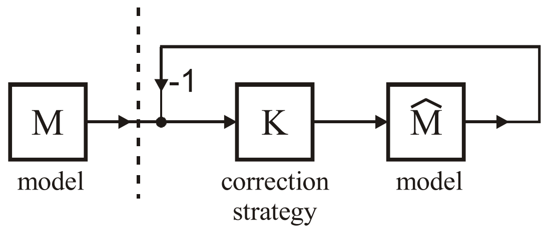

Fig. 1. Observer approach

The signal model is based on the well-known Fourier series, which is an efficient way to represent periodic signals. Resonators can generate the basis functions of the Fourier series. The initial state of the resonators determines the amplitude and the phase of the periodic signal, and the parameter of the feedback sets the frequency of the signal. Summing the output of the resonators, the result is a possible model of the signal [2].

The model can be compacted in one expression

$$y_n = \mathbf{c}_n^T \mathbf{x}_n, \quad \mathbf{c}_n = \left[c_{n,k}\right] = e^{j2\pi f_1 kn} \tag{1}$$

where $\mathbf{x}_n$ is the state vector, $\mathbf{c}_n$ contains the basis vectors of the Fourier expansion with f_1 fundamental frequency.

The harmonic cancellation follows the prediction-correction scheme, where the signal model is repeated in the observer. Ignoring the details of the observer design in case of the resonator based signal model, the description of the observer is

$$\hat{\mathbf{x}}_{n+1} = \hat{\mathbf{x}}_n + \mathbf{g}_n \left(y_n - \mathbf{c}_n^T \hat{\mathbf{x}}_n \right) \tag{2}$$

where $\hat{\mathbf{x}}_n$ is the estimated state vector, and $\mathbf{g}_n$ is called the 'correction strategy'. The observer can be seen in Fig. 2. The correction equals to

$$\mathbf{g}_n = \frac{1}{N} \bar{\mathbf{c}}_n \tag{3}$$

if $f_1 = f_s / N$, where f_s is the sampling frequency. One channel of the observer is a resonator. The observer in this case performs a recursive Fourier transformation of N points.

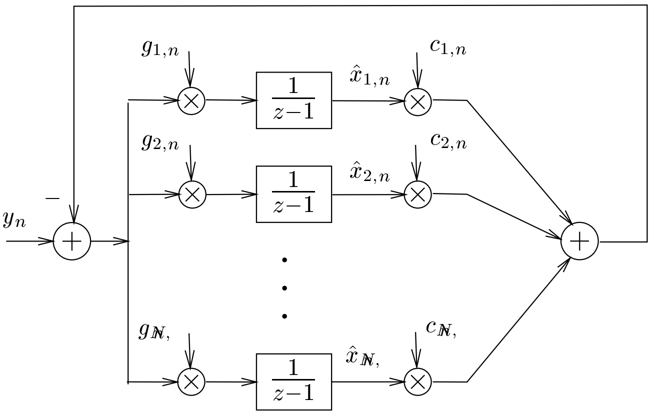

Fig. 2. Observer for periodic signals

In practical applications (as for a power source) the fundamental frequency changes and therefore the resonators can not placed uniformly. However, if a reference signal is available, it can be used to determine the exact resonator frequencies. An adaptive Fourier analyzer is used for this purpose [3].

3. DISTORTION CANCELLATION

In the resonator-based observer in steady state the input of the resonators (i.e. the feedback error) equals zero. This means that the feedback signal (the sum of the resonator outputs) cancels the input signal. If acoustic noise should be canceled, the output of the resonators should be connected to a loudspeaker and fed back using a microphone. (A multiplication by –1 is necessary in the controller.) The frequency is estimated by an independent AFA and it passes the actual resonator positions to the controller. Reference signal can be any periodic signal with the same fundamental frequency as the primary noise. The arrangement can be seen in Fig. 3.a.

Fig. 3.b shows the block diagram of the control loop, where $R(z)$ and $A(z)$ denote the resonator based controller and the acoustic transfer function between the loudspeaker and the microphone, respectively. Due to the presence of $A(z)$, the stability of the system is not obvious. The controller design will be accomplished by the appropriate choice of the vector $\mathbf{g}_n$. It is proven [3] that they should be:

$$g_{k,n} = \frac{\alpha}{N} \bar{c}_{k,n} w_k; \quad w_k = \frac{1}{A(c_{n+1}/c_n)} \tag{4}$$

where α is a convergence parameter. The vector $\mathbf{w}$ is the reciprocal of the transfer function $A(z)$ evaluated at the resonator frequencies. The actual vector $\mathbf{w}$ depends on the fundamental frequency of the primary noise. $A(z)$ is in general not analytically known and (4) cannot be calculated on-line, therefore the transfer function should be measured at a finite number of points and the inverses should be calculated off-line.

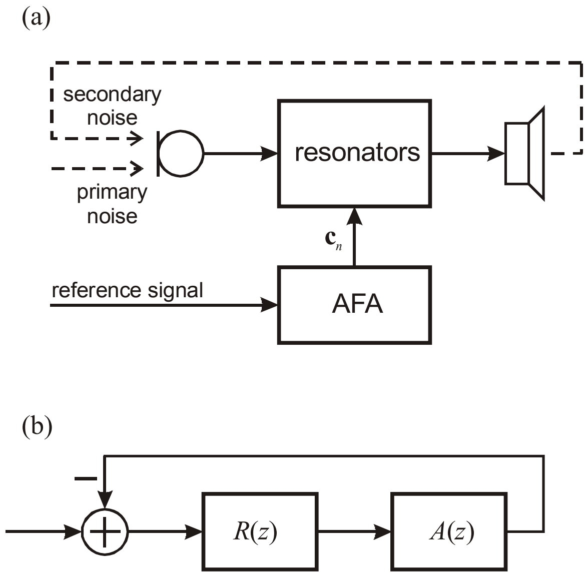

Fig. 3. Periodic noise control (a)
and block diagram of the control loop (b)

If distortions of a signal (voltage or current) should be cancelled, the output of the resonators should be connected to a sensing circuit and fed back using an actuator. However, in this case a slightly modified structure should be used which does not suppress the fundamental component of the generated periodic signal. This structure can be seen in Fig. 4. In the figure x denotes the reference signal, while e and y stand for the error and the actuator signals, respectively. In this case the error signal (from the sensing circuit) is lead not directly to the input of the resonators, but a common resonator-based observer (with resonator positions provided by the AFA) analyzes it, and the signal components excite independently the resonators. There are resonators only for higher harmonics. It can be also proven that using the same $\mathbf{w}$ vector defined in (4) can ensure the stability of the structure. The elements of $\mathbf{w}$ can be implemented as weights at the inputs of the resonators. The vector $\mathbf{w}$ can be calculated from the measured transfer function. This measurement can be done by the utilization of the resonator-based observer, as well [5].

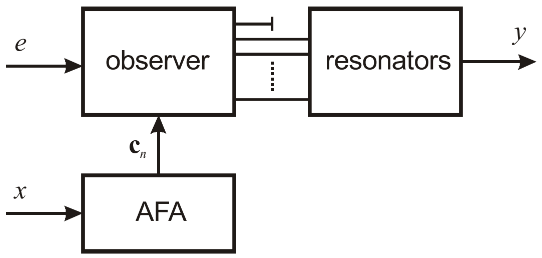

Fig. 4. Adaptation of the resonators

In practice the high-power signal is measured by the use of a precise transformer (Fig. 5.). All the signal-processing algorithms are realized in the DSP. The correction signal is amplified with an audio power amplifier, and fed back. The A/D and D/A conversion circuits and the DSP determine the limit of the correction.

The correction system is non-linear, as well. It contains a transformer to cancel the harmonics. The transformer is a non-linear device, therefore it can also cause unwanted spectral content. Another problem with the transformer is that it injects also power to the output of the control system. As a result, the error signal will be different. The power amplifier steering the transformer is also a non-linear component. However, the controller suppresses the distortion of the signal apart from its origin, so the result is a pure sine wave.

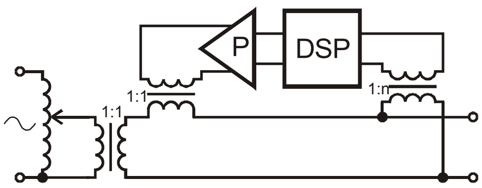

Fig. 5. Test set-up

5. MEASUREMENT SET-UP

The control loop was also realized in order to test the system under real circumstances with real signals. The input signal was generated from the line power. The electronic parts of the measurement setup correspond to Fig. 5.

The control algorithm was implemented on a commercially available evaluation board, containing an ADSP-21065L floating-point DSP and an audio CODEC (AD1819). Actually, three sets of resonators had to be implemented. One was required to realize the adaptive Fourier analyzer, which estimates and follows the fundamental frequency of the input signal, even if it is changing. In case of a power source, it cannot be supposed that the fundamental frequency is stable. However, the harmonic cancellation should work for a slightly changing fundamental frequency. For the reasons described previously, the AFA is only used to update the resonator rotating vectors, which determine the basis vectors of the resonator. The input signal is processed as it is described in the previous section.

The implementation of one resonator requires a vector-vector multiplication to calculate the reconstructed signal, and an update of the state variables. It is possible to save some processing time by utilizing the fact that the state variables contain conjugate complex pairs, and the update of the DC state requires less computation. Therefore only the half of the AC state variables should be updated and used.

6. MEASUREMENT RESULTS

First, the transfer function of the plant was measured. This is a very important phase of the test. The same setup has to be used as will be used for the active harmonic cancellation. Using the DSP and the CODEC the transfer function measurements were performed, by the procedure referred in the previous section [5]. The measured transfer function is shown on Fig. 6.

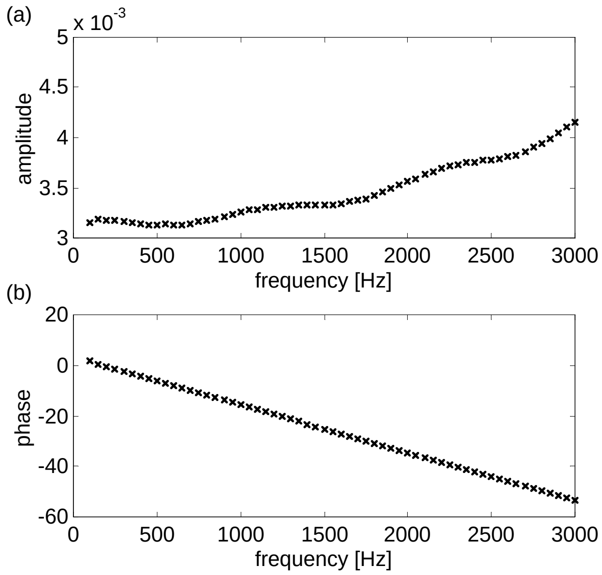

Fig. 6 The amplitude (a) and (b) phase of the measured transfer function

The transfer function is measured at 50 Hz and its harmonics. 60 AC resonator channels were planned to use (this means a total number of 121 resonator channels), therefore the transfer function was measured from 50 Hz to 3 kHz. However, the fundamental frequency of the real signal can slightly differ from the theoretical 50 Hz. The harmonic cancellation is adaptive, but the measured transfer function is not changed. In our case the transfer function (in contrast to an acoustic one) is quite smooth, and the frequency change is assumed to be small, therefore the same transfer function can be used.

The RMS value of the output voltage was set to 100 V. The measurement results of the original distorted and the compensated signal are shown on Fig. 7. The upper figure (Fig. 7.a) shows the original signal, which is divided from the line power by the use of precise transformers. The distortion of the signal was also measured, which was about 4.9%. At the peaks of the sine waves the distortions can be clearly seen. These measurement data were downloaded from a digital oscilloscope, where the sampling frequency was 50 kHz. The bottom figure (Fig. 7.b) shows the compensated signal. One can see that the ‘visible’ distortions disappeared, the measured distortion of the signal was 0.3%

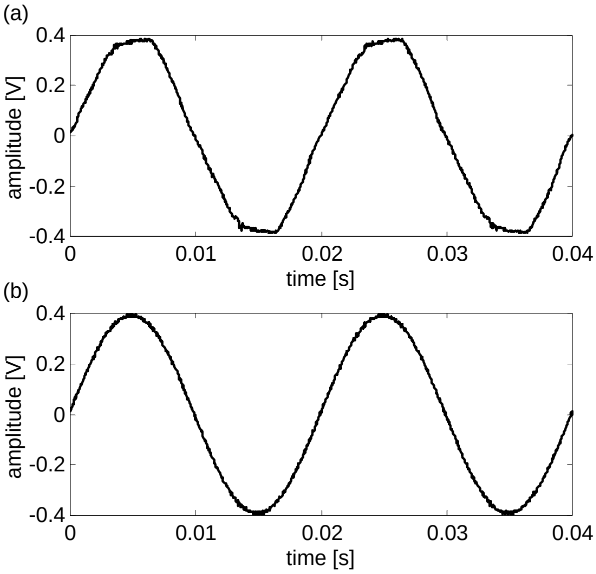

Fig. 7. DSP input signal with distortion (a), and after the cancellation (b)

The next figure (Fig. 8.) shows the Fourier transform of the distorted and undistorted signals. 60 AC resonators were used, thus harmonics in the range of 100 Hz and 3 kHz were compensated.

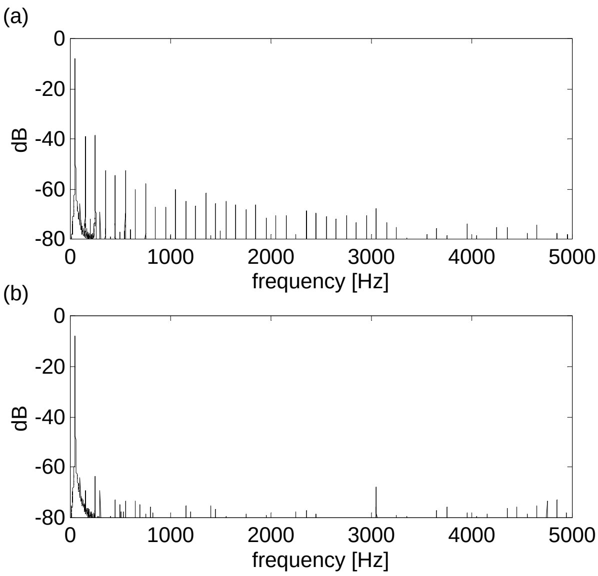

Fig. 8 Spectra of the distorted signal (a), and after the cancellation (b)

Looking at the spectra, the effect of the compensation is visible. While in the original signal the harmonic frequency components have a value between –40 and –70 dB, in the compensated case just a few components remained with a value less than –70 dB. Utilization of more resonators can result in further improvements in the harmonic cancellation.

7. CONCLUSIONS

The goal of the research was to show that the on-line suppression of the harmonic content of a high-voltage signal is possible by a compensation signal having relatively small power. The quality of signal generators can be efficiently increased using periodic noise control techniques. They have the advantage of adapting the algorithm to the signal itself. The model of the input signal is based on the Fourier representation of the signals, and the observer is designed according to the assumed signal model. The resonator-based observer estimates the Fourier coefficients of the signal. The resonators are set by an adaptive Fourier analyzer, which can follow the changes in the fundamental frequency. The method is even useful to compensate the distortions of a signal having slightly varying frequency, due to the adaptive control.

The low power of the correction signal extends the range of possible application areas. Measurement results clearly confirmed the theoretical derivations: the adaptive control can follow the changing frequency of the input signal, and the actuator signal had low power. As a result, the distortion of a signal having 100 V effective value was decreased by an order of magnitude. The method is easily applicable for current sources, as well. Possible application areas of the method are those, where a high-voltage signal is needed with increased spectral purity, e.g. power meter calibrations.

REFERENCES

[1] F. Louge, J. Schoukens, Y. Rolain, "Generation of computer controlled excitations and its application to the detection an measurement of harmonic distortions", *Proceedings of the Instrumentation and Measurement Technology Conference*, May 10-12, 1994, Hamamatsu, Japan, pp. 1385-1388.

[2] G. Péceli, "A common structure for recursive discrete transforms", *IEEE Transactions on Circuits and Systems*, Vol. CAS-33, pp. 1035-36, Oct. 1986.

[3] F. Nagy, "Measurement of signal parameters using nonlinear observers", *IEEE Transactions on Instrumentation and Measurement*, Vol. IM-41, pp. 152-155, Feb. 1992.

[4] L. Sujbert, G. Péceli, "Noise cancelation using resonator based controller", *Proceedings of the International EAA Symposium on Active Control of Sound and Vibration*, Aug. 21-23, 1997, Budapest, Hungary, pp. 905-916.

[5] L. Sujbert, G. Péceli, Gy. Simon, "Resonator based non-parametric identification of linear systems", *Proceedings of the IEEE Instrumentation and Measurement Technology Conference*, May. 21-23, 2001, Budapest, Hungary, pp. 166-169.

Authors: Balázs VARGHA, Dept. of Measurement and Information Systems, Budapest University of Technology and Economics, H-1521 Budapest, Hungary, phone: +36-1-4633587, fax: +36-1-463-4112, email: vargha@mit.bme.hu
László SUJBERT, (see B. VARGHA), phone: +36-1-4634114, email: sujbert@mit.bme.hu