APPLICATION OF MODERN METHODS OF DIGITAL SIGNAL PROCESSING IN CONTROLLING POWER ACTIVE FILTERS

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One of the main reasons for the deterioration of the quality of electricity in distribution networks is the increase in the number of nonlinear devices that create non-sinusoidal currents during operation. Such devices are switching power supplies, variable speed electric motor drives, ballasts for electroluminescent lamps, etc. [1, 2]. Non-sinusoidal currents and voltages can be represented as the sum of harmonics, the frequencies of which are multiples of the fundamental frequency of the power supply network. The result of the "clogging" of the power supply system with harmonics is an increase in losses in rotating machines, transformers, power transmission lines, accelerated aging of electrical equipment insulation, false triggering of relay protection and automation devices [3].

An active filter is a complex device consisting of a power unit (inverter) and a power unit control unit. The active filter analyzes the harmonic composition of the distorted current (voltage) at the connection point and generates a similar spectrum of harmonics in antiphase. As a result, the higher harmonics are compensated (neutralized) and the current consumed from the source retains a sinusoidal shape.

The spectral composition of the signal generated by the active filter control system must match the harmonic composition of the non-sinusoidal current generated by the nonlinear load. The only exception is the fundamental harmonic. It must be absent in the compensating current. To obtain the signal, can use a notch filter tuned to the frequency of the fundamental harmonic. However, this approach has disadvantages. First, the notch filter is a static device, and its characteristics cannot change when the frequency and amplitude of the fundamental harmonic change. Second, the nonlinear phase-frequency characteristic of the notch filter causes distortion of the compensating signal. Thus, to form the control signal of the active filter, an adaptive device is required, the characteristics of which change when the spectral composition of non-sinusoidal currents changes.

The block diagram of an adaptive noise compensator based on an FIR filter (with a finite impulse response) in the form of a digital delay line is shown in Fig. 1.

Two signals are simultaneously input to the active filter input:

– input signal x(n) – unknown in advance;

– reference signal d(n) – known in advance.

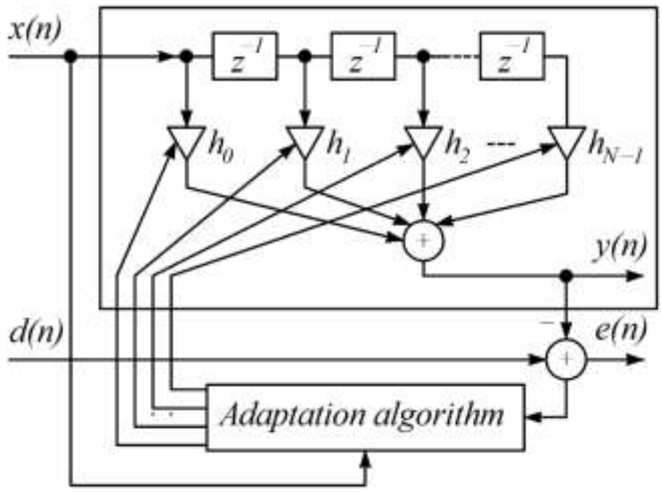

Figure 1 – Block diagram of an adaptive noise compensator

The signal x(n) is fed simultaneously to the input of the FIR filter and the adaptation algorithm. At the output of the active filter, two signals are formed: the output signal y(n) and the error signal e(n).

The goal of adaptive processing is to ensure the best approximation of the output signal y(n) to the reference signal d(n) according to a given criterion. The main difficulty in designing an active filter is associated with the development of adaptation algorithms.

The value of the adaptation step is chosen taking into account trade-off considerations: on the one hand, it affects the convergence rate of the LMS algorithm (the larger μ, the higher the LMS), and on the other hand, it affects the error signal e(n) (the larger μ, the more e(n) differs from the error signal in the filter).

The main advantage of the LMS algorithm is its simplicity (only N multiplication-addition operations are required at each step), and the disadvantage is the relatively slow convergence of the iterative procedure for calculating the parameters of the active filter.

Conclusion

Thus, the use of the proposed digital signal processing method to control the characteristics of the active filter allows the use of various adaptive methods of spectral estimation and interference compensation, as well as efficient and inexpensive hardware. The results of the research show that active filters using digital signal processing methods can change their characteristics in real time when the non-sinusoidal mode in the network changes.

References

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