INVESTIGATION OF A DETECTOR OF DIGITAL AMPLITUDE-MODULATED SIGNALS

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The paper considers operation of an amplitude detector of digital signals, where the detector contains a nonlinear element (a full-wave digital gate) connected in series with a digital filter. Dependencies of the detection coefficient of the harmonic distortion factor on the digital filter parameters are established. The results are compared with the case when a half-wave digital gate is used.

Development of the microelectronic digital and analog circuitry and the advent of new components brought about some changes in into the radio reception path, especially in the places traditionally built with the use of analog decisions. In fact, all the architecture of the reception path underwent modification [1]. If an output signal of the high-frequency path of the receiver is digital, then digital detection of the digital radio signal is desirable. An amplitude detector using a half-wave digital gate has been investigated in [2].

At the same time, of interest is investigation of a detector whose main part is a full-wave gate with its input-output characteristic of the type $x = f(x_1)$, where $f(x_1) = |x_1|$. Note that this type of gate can be easily realized in widely used "without-sign" processing of analog-digital converter readings.

Consider the detection of a signal with fixed amplitude. Assume that the detector input is fed by a harmonic signal $x_1(n) = \sin \omega n$. At the output of the nonlinear element we obtain a periodic signal x(n), which, at an even *T*, meets the difference equation

$$x(n+T/2) = x(n)$$
 (1)

with initial conditions x(0), x(1), ..., x(n), ..., x(T/2 - 1), where $T = 2\pi/\omega$. Treatment of equation (1) with the aid of *z*-transformation [3] gives

$$x(n) = 2\sin\omega \sum_{m=0}^{T/2-1} A_m^{-1} B_m^{-1} z_m^n$$
(2)

(2)

where $z_m = \exp(j2\omega m)$, $A_m = z_m^2 - 2z\cos\omega + 1$, $B_m = \prod_{\substack{i=0\\i\neq m}}^{T/2-1} (z_m - z_i)$.

In order to isolate the legitimate signal (in our case — the dc component of the spectrum of the function x(n)), in series with the nonlinear element we connect a low-pass digital filter. When such a filter is realized with the aid of a first-order recursive circuit, the processes in the filter, with no regard for the effects of overflow and quantization, can be described by a difference equation $y(n+1) = x(n) + b_1 y(n)$, where y(n) is the circuit response and b_1 is the circuit parameter. To resolve this equation, let us use the method of *z*-transformations, bearing in mind that the transfer function of the network has the form $H(z) = (z - b_1)^{-1}$.

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Radioelectronics and Communications Systems Vol. 48, No. 1, 2005

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6 April 2004