METHODS OF SUPPRESSION OF CORRELATED INTERFERENCE

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Abstract. Issues of increasing information accuracy are considered important and measuring devices can be affected by various kinds of bias and various errors are caused based on this information. In order to improve the accuracy of the data and achieve a set level of quality, various filters have been developed with which we can eliminate noise and interference. Different noise elimination methods have been considered and they have been compared among themselves.
Key words: signal, filter, noise, modulation, demodulation, frequency, fourier transform.

Annotatsiya. Axborotning aniqligini oshirish masalalari muhim hisoblanadi va o'lchov asboblari har xil turdagi tashqi ta'sirlarga duch kelishi mumkin va bu ma'lumotlar orqali turli xil xatoliklar yuzaga keladi. Ma'lumotlarning aniqligini oshirish va ma'lum bir sifat darajasiga erishish uchun xalaqit va shovqinlarni bartaraf etishimiz mumkin bo'lgan turli xil filtrlar ishlab chiqilgan. Shovqinni bartaraf etishning turli usullari ko'rib chiqildi va bir-biri bilan taqqoslandi.
Kalit so'zlar: signal, filtr, shovqin, modulyatsiya, demodulyatsiya, chastota. Furye almashtirishi.

Аннотация. Вопросы повышения точности информации считаются важными, и измерительные приборы могут быть подвержены различным видам смещения и на основе этой информации возникают различные ошибки. Для повышения точности данных и достижения заданного уровня качества были разработаны различные фильтры, с помощью которых мы можем устранить шум и помехи. Были рассмотрены различные методы устранения шума, и они были сравнены между собой.
Ключевые слова: сигнал, фильтр, шум, модуляция, демодуляция, частота, преобразование Фурье.

Introduction

In order to improve information accuracy in the field of digital electronics, we need to improve the achievements of science and technology, the effective use of modern methods. We know that from year to year new technologies are coming in. IT devices receive data directly remotely or directly from the device itself, and so that when exposed to the electromagnetic field of various external influences, a distortion occurs in the data that uses digital filters to improve the data. For example, if we consider remote sensing technology. He really has a wide range of possibilities. In this too but there are external influences they reduce the accuracy of information we use filters to improve information accuracy and we achieve an increase in the quality of the information being received. Another example is digital products and the currently used methods of drying in various fruit and vegetable processing industries require the use of new processing technologies that increase the quality of the final product, shorten the processing time and improve the quality of the dehydrated material. We also use different sensors in that they are also affected by external influences causing the appearance of measurement errors i.e. humidity errors can occur when measuring temperature, we also use filters to eliminate these errors. In doing so, we will consider the methods and means of noise filtering.

Materials and Methods

Detection and examination of signals always occurs under conditions of interference caused by both external and internal reasons with respect to the information transmission channel. Industrial, atmospheric and space interference, as well as interference from radio devices of various purposes, can be attributed to external interference of natural occurrence. In addition, there may be artificial interference, specially created to suppress a clear source of information, sometimes also called staged interference. there is also a change in signals interconnected with scattered or multipath propagation of signals. When processing weak signals, internal noise of information processing devices caused by the thermal movement of electrons or other effects play a huge role The usual approach to isolating signals against the background of additive interference is to pass a mixture of signal and noise through a filter in which noise is suppressed without signal distortion. The development of such filters is a task of optimal filtration, the founder of which was Wiener. Further development of his idea took place in the works of Kalman, Busey, Kolmogorov, etc. So, for example, to detect an FM signal against a background of Gaussian interference, the optimal device is one that produces a consistent correlation conversion of signals [1].
Having high efficiency and operability, modulation and demodulation methods are widely used in communication systems. But in cases where the noise is not Gaussian, the situation becomes more complicated. Therefore, the development of devices optimal for signal processing under conditions of non-Gaussian narrowband and structural interference requires the creation of nonlinear devices of great complexity [2,3,4].
Filters used to isolate interference can have constant parameters or be adaptive [5]. The calculation of filters with constant parameters requires a priori knowledge of signal characteristics, as well as narrowband and structural interference. The parameters of adaptive filters that can be automatically rebuilt require a minimum of a priori information.
The "whitewashing" filter is the basis for optimal signal processing methods against the background of noise and narrowband interference with known interference parameters and many adaptive methods when the parameters of narrowband interference are unknown [6].
For systems operating under conditions of narrowband interference, the device performing the operation of “whitewashing” the original signal is a notch filter. Since narrowband interference is a highly correlated process against the background of a mixture of SHPS and thermal noise, it is possible to predict its values for the future with subsequent subtraction of interference from the incoming signal. Smoothing narrowband interference can significantly improve the characteristics of the communication system [7,8].
The implementation of a filter that cuts interference is possible, for example, by using delay lines with taps to implement a one-way or two-way (transversal) error prediction filter [9]. The improvement of system performance in the case of transversal filters is explained by the use of both past and future samples of the input signal for predicting interference values. When optimizing the signal processing algorithm to minimize the root-mean-square error (RMS), it is possible to use lattice (in-line) filters or solution feedback filters [10].
The idea of using an OSR filter is to eliminate a useful signal from the received signal to "bleach" a mixture of interference and noise. Since the output signal of the PU is an estimate of the transmitted information symbol, this estimate can be used to form a copy of the transmitted signal, which is subtracted from the mixture of signal, noise and interference. If the decision at the receiver output, taken according to the current estimated value of the information symbol, is correct, then as a result of subtraction, only noise and interference signals remain, to which, in turn, the "bleaching" operation is applied [11].
Incorrect evaluation of the useful signal, however, can lead to the presence of an increased value of the signal component at the input of the "bleaching" filter, which leads to a certain increase in the probability of error in the evaluation of SHP symbols. Suppression of concentrated narrowband interference can also be achieved by cutting out the part of the

spectrum corresponding to the interference from the composite spectrum of the signal, noise and interference. Such methods use real-time Fourier transform with distributed delay lines with couplers in the form of surface acoustic wave devices [12] or surface charge-coupled devices (CCD) [13]. Narrowband interference reduction by such methods involves three sequential operations: firstly, the Fourier transform of the received signal, secondly, multiplying the conversion result by the frequency response of some suitable filter to cut out part of the signal spectrum along with the interference spectrum, and thirdly, the inverse Fourier transform to restore the time form of the signal. This is achieved by performing the following three operations.
A common problem in the practical application of SAW and CCD filters is the small dynamic range of these devices, which reduces the effectiveness of the interference suppression devices using them under the action of high interference intensities [14]. A significant improvement in reception characteristics in conditions of both narrowband and broadband interference can be achieved by applying methods based on the principle of adaptive interference compensation [15]. The advantages of adaptive interference compensation methods are the ability to adapt to interference parameters, low output noise and low distortion of the desired signal. The adaptation of the processor makes it possible to compensate for both narrowband and structural interference, uncharacterized interference and, in some cases, non-stationary interference. Adaptive noise compensators are reliable systems that automatically shut down if an increase in the signal-to-noise ratio is not achieved. The noise level at the output and the degree of signal distortion in such systems are usually lower than in conventional optimal filters [16].

Results

Interference compensators use an additional channel, also called a reference channel, in which only noise acts, or where the useful signal is weak or below the detection threshold. Interference in the secondary channel passes through the filter and is subtracted from the voltage of the main channel, which is a mixture of signal and interference. If there is no signal at the reference input of the compensator and a number of other conditions are met, the interference acting on the main input can be removed without distorting the signal. As a result, the noise in the main channel is reduced or completely suppressed. In the case of periodic noise, the adaptive compensator becomes a narrow-band deflecting filter with virtually unlimited suppression coefficients and precisely follows the frequency of interference [17].

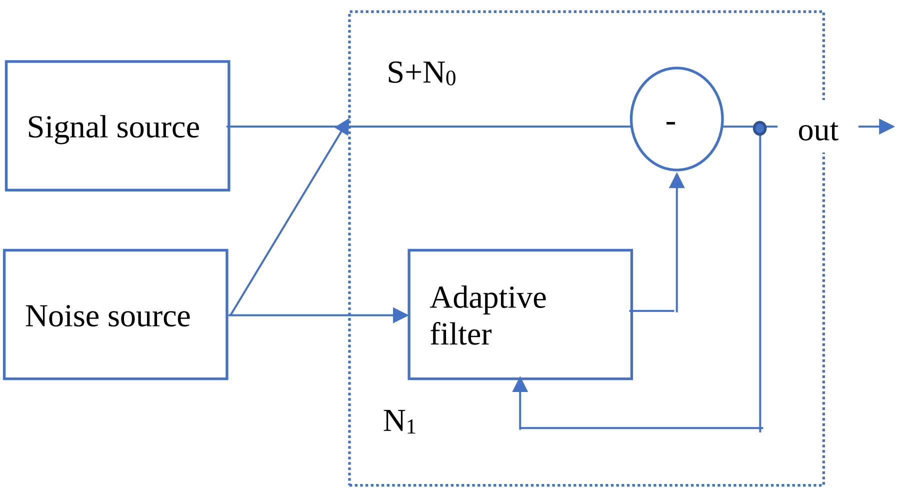

Fig.1. The principle of adaptive interference compensation.

DOI 10.24412/2181-1431-2024-3-5-12

Figure 1 is a schematic illustration of the principle of interference suppression by an adaptive interference compensator. A mixture of signal and interference s+N_0 forms the input voltage of the main channel of the compensator. The additional channel of the compensator receives interference n1, uncorrelated with the signal and serving as a reference signal for the compensator. The interference voltages n0 and n1 are correlated with each other. The interference N_1 passes through the adaptive filter and generates a noise signal y at its output.

The filter parameters are selected in such a way that the voltage y differs as little as possible from the interference voltage n_0. These two voltages are supplied to the subtractor, where the output signal of the system is generated.

The output signal of the compensator is an error signal for the implementation of adaptation. In some cases, if the compensator is inaccurately tuned to interference, the noise at the output of the compensator may increase. However, the adaptation principle allows noise attenuation with a low probability of signal distortion or an increase in the noise level at the output of the circuit. Adaptive compensators can achieve elusive, if not impossible, levels of noise reduction with direct filtering. Adaptive devices need to be pre-configured if the statistical characteristics of noise change slowly. For stationary random effects, such self-learning adaptive filters very accurately approximate Wiener filters with constant parameters [18].

The compensation system is usually constructed in such a way that its reference input contains only noise that correlates with the noise at the main input. In some cases, the reference input of the adaptive compensator gets not only noise, but also part of the useful signal. When the useful signal enters the reference channel of the compensator, distortions occur at the output of the system.

The magnitude of this distortion is determined by the level of the signal passing through the adaptive filter [19], where it is shown that if there is a signal in the reference input channel and the signal-to-noise ratio at the input of this channel is small, then the compensation degradation is not so significant that an adaptive noise compensator is not used. If the signal-to-noise ratio at the input of the reference channel is -13 dB, then the maximum signal distortion is less than 5%. The structure of the GP adaptive compensator is shown in Figure 2 [20].

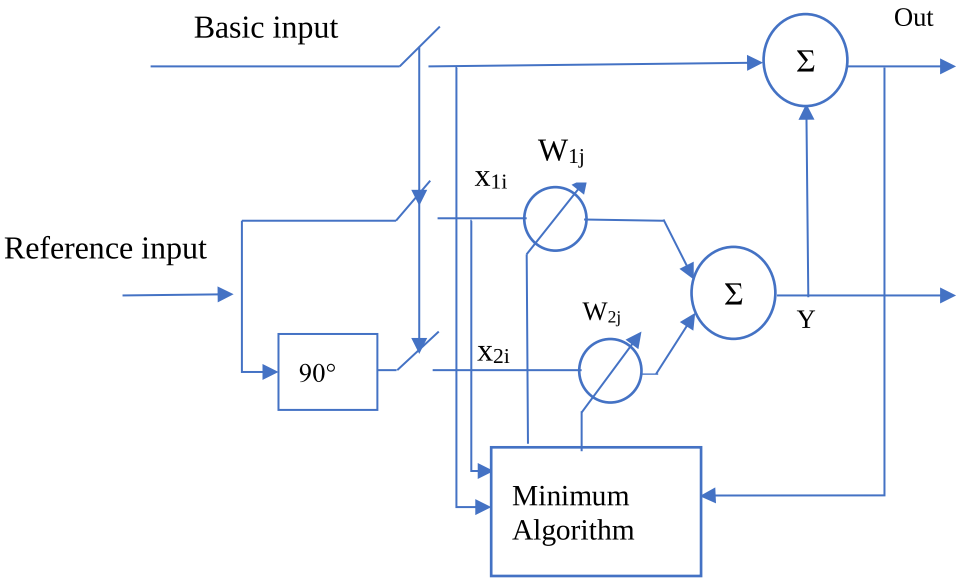

Fig. 2. Harmonic interference compensator.

It is assumed that a harmonic signal operates at the reference input cos(ω_0t+φ). The signals coming to both inputs are sampled with the frequency Ω = 2π/T rad/s. In the reference channel, the interference signal is divided into two quadrature components, which, after sampling, form samples x_{1j} and x_{2j}. All samples are synchronized and formed at time points *t=0,±T,±2T etc*. The weighting coefficients w_1 and w_2 are formed according to the algorithm of the minimum standard error at the output of the compensator.
The advantages of adaptive regenerative filters are easy adjustment of the opacity band, virtually unlimited suppression and accurate tracking of the interference frequency. In the results of the study of the effect of several runways on compensation systems [21], regeneration is established for each VP.
Let, similarly, at the input of the reformatory device there is a signal x(t) in the exterior of the additive mixture of the wanted periodic noise signal s(t) long L_cT (L_c -base noise of similar signal, T- clock frequency noise similar signal), powerful harmonic interference and white Gaussian noise ɑ(t). We assume that the possibility of error in signal reception is determined mainly by the power of the correct interference. It is required to prepare a mechanism for separating and compensating harmonic noise from the received signal. To isolate the harmonic noise from the received signal, we will use a digital two-sided transversal filter. Consonant noise suppression is performed by subtracting (compensating) the assigned harmonic noise from the received signal. Consider the characteristics of such a dampening filter in the frequency domain. Digital smoothing filter that averages the signal has an impulse response $h_{(m)}=h(mT)$ the following kind

$$h(m) = \begin{cases} \frac{1}{2\Delta+1}, m = l-\Delta, \dots, m, \dots m+\Delta \\ 0, |m-l| > \Delta \end{cases} \tag{1}$$

Here m- bar number. Δ - an integer depending on the size of the filter K. The size of the digital smoother is K = 2Δ +1.
The response of a digital smoothing filter with an impulse response (1) to the input signal is

$$y_{(l)} = \frac{1}{2\Delta+1}\sum_{m=l-\Delta}^{l+\Delta} x_{(m)} \tag{2}$$

The frequency response of a digital antialiasing filter based on the Fourier transform is determined by the following formula:

$$H(w) = \frac{1}{2\Delta+1}\sum_{m=l-\Delta}^{l+\Delta} e^{-jwr} = \frac{\sin\left(\frac{2\Delta+1}{2}\right)wr}{(2\Delta+1)sin\left(\frac{wr}{2}\right)} e^{-j\Delta wr} \tag{3}$$

The considered digital smoothing filter delays the input signal by A cycles and completely suppresses harmonic oscillations with frequencies that are multiples of 2π/M.
Denote by $a_{(m)}$, $b_{(m)}$, и $o_{(t)}$ samples of signals at the output of the digital smoothing filter, belonging to the noise-like signal, hormonal interference and noise, respectively. The estimate of the amplitude of hormonal interference in this case is equal to the signal at the output of the digital smoothing filter:

$$\mathrm{a}_{(\mathrm{m})} = b_{(m)} + o_{(t)} \tag{4}$$

It is clear that the subtraction of the harmonic interference estimates $\mathrm{a}_{(\mathrm{m})}$ from the received signal, it will allow to achieve deep cutting of harmonic interference if the condition is met $M \ll \frac{2\pi}{wr}$ which reduces the difference. However, with a reduction in size M filter, the power of white noise and noise-like signal at the output of the digital smoothing filter increases, reducing the reliability of the estimation of the amplitude of hormonal interference $\mathrm{a}_{(\mathrm{m})}$, what negatively affects the noise immunity of the converter mechanism. Find the optimal value M based on the signal strength σ^2 harmonic interference power σ_a^2, its frequencies w_a, and white noise power σ_n^2 at the input of the converter device.

To do this, we will examine the variance of the difference between the amplitude of the harmonic interference and its evaluation $a_{(m)}$, In the absence of correlation between the components of the received signal, the variance of the error in estimating the amplitude of the harmonic interference is equal to

$$a_{(er)} = \left(a_{(m)} = b_{(m)} + o_{(t)} \right)^2 = \sigma_a^2 (1 - |H(w) + \sigma_m^2 + \sigma_t^2|) \quad (5)$$

The power of a noise-like signal at the output of a digital smoothing filter is determined by the expression

$$\sigma_{a1}^2 = \frac{\sigma_a^2}{M^2} \sum_{m=l-\Delta}^{l+\Delta} (M - |k| r(k)) \quad (6)$$

Where r(k)- autocorrelation function of a noise-like signal substituting $r(0)=K_c$ and $r(k\neq0)=\frac{1}{L_c}$ we get

$$\sigma_{a1}^2 = \sigma_a^2 \frac{L_c - M + 1}{L_c M} \quad (7)$$

Similarly, for white noise we have

$$\sigma_t^2 = \frac{\sigma_t^2}{M} \quad (8)$$

In general, the frequency of harmonic interference will be much less than the ratio 2π/M (where M>3) , which allows in the expression (3) use row decomposition $\sin(w)=w-w^3$, correct when $w<\pi/2$. Given this fact, substituting formulas (5), (6) and (7) into the formula (8), and assuming $K_c >> M$, we obtain an expression for the variance of the error in estimating the amplitude of the harmonic interference

$$\sigma_{er}^2 = \sigma_a^2 \frac{M^2 w_a^2}{24} + (\sigma_b^2 + \sigma_t^2) \frac{1}{M} \quad (9)$$

The ratio of useful signal uncompensated interference is equal to

$$\frac{\sigma_a^2}{\sigma_{er}^2} = \frac{\sigma_a^2}{\sigma_a^2 \frac{M^2 w_a^2}{24} + (\sigma_a^2 + \sigma_t^2) \frac{1}{M}} \quad (10)$$

Maximum relationship $\frac{\sigma_a^2}{\sigma_{er}^2}$ located at the point

$$M' = \sqrt[3]{\frac{3}{\pi^2} L_n^2 (\gamma + \beta)} \approx 0.7 \sqrt[3]{L_n^2 (\gamma + \beta)} \quad (11)$$

Based on the conditions of the physical feasibility of the digital smoothing filter and the possibility of unambiguous determination of the signal delay time Δ at the output of the digital smoothing filter, the filter size is selected as the nearest odd integer to M'. The digital smoothing filter constructed in this way minimizes the impact of harmonic interference on the noise immunity of the converter device.

Conclusion

Consider the implementation of a linear smoothing filter to extract GP from the received signal. The effect of harmonic interference on the characteristics of devices. The search with RSF depends on both the power of the interference and its frequency. The most powerful impact is detected by GP creating SHPS on the frequency interference with a frequency close to the repetition frequency of the SHPS components. As needed After the noise period increases, its impact is reduced.

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