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MODELING OF ACOUSTO-ELECTRIC DEVICES AND SYSTEMS BASED ON A GENERALIZED SIGNAL CONVERTER CONNECTION

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A B S T R A C T

The use of electronic systems is widespread in various metrology systems related to the monitoring of acoustic noise in urban or industrial environments. A method for analyzing equipments intended for acoustic-location and acoustic measuring devices and systems (DS) based on structural approximation is proposed. The method involves performing a structural approximation of the devices used based on the developed generalized circuit. In this particular instance, the analysis is conducted utilizing the apparatus of continuous piecewise-linear functions and transfer functions of the signal converter. Their utilization facilitates the investigation of devices (systems) of arbitrary structure, characterized by random non-linear characteristics and a variety of inertial components such as blocks and units. The study of specific devices is based on the finite expressions of the transfer functions of a generalized signal converter connection. Several levels of disclosure of the converter connection are considered. Using the aforementioned methodology, it is feasible to synthesize acousto-electric devices (systems) at both the analytical and structural levels. As nonlinear differential equations of random order that describe the response of devices and systems can be reduced to linear equations of the first order in the general case, the inverse solution (synthesis) is not a complex task, both at the analytical and programmatic levels.

1. INTRODUCTION

Electronic systems are widely used in various metrology systems related to monitoring acoustic noise in urban or industrial environments. In relation to urbanized areas, certain prerequisites for the design and operation of such systems are formulated within the framework of the "Safe City" Program. Requirements for the architecture of the hardware and software complex (HSC) at the municipal level (The Unified Requirements for the Technical Parameters of the Hardware and Software Segments of Safe City. Hardware and software complex "Safe City", 2017) establishes that the composition of the unified information and telecommunications infrastructure of the HSC should be formed by a distributed network of

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software and hardware systems that provide monitoring and are united by a single transmission data network. HSCs basic information and communication infrastructure consists of a single transmission data network, a set of software and hardware that forms a telecommunications infrastructure designed to support information-transmission process.

There are similar decisions being made in other countries (Pérez et al, 2020; Fernández-Prieto, Bago & Gadeo-Martos, 2019).

Similar difficulties arise in instances of converting an electrical signal into an acoustic one, such as in acoustic sounding systems or analyzing the propagation length under intricate circumstances.

The standardization of requirements for measuring equipment requires the development of methodologies to the synthesis of measuring paths that incorporate emerging associated factors, as well as standardization of complex acoustic signal conversion systems (from primary conversion in microphone devices to information representation).

Analysis and theoretical studies of such devices and systems of various types are usually carried out according to the following algorithm (Bulkin, Grigoryuk, & Lashin, 2013; Grigoryuk, & Bulkin, 2017; Sidor, 2018; Dopf et al, 2005):

1. generation of equations of the device (system);
2. substitution of analytical expressions for the characteristics of the components of the device into the resulting equations;
3. device stability research;
4. solving the resulting equations for given types;
5. studying the stability of devices;
6. solving the resulting equations for the given types and parameters of the used (applied) generated and processed signals.

The equations of real blocks of devices and systems (EDS) systems are nonlinear and lack a general differential solution. Therefore, computational solutions for the equation on a computer are commonly employed.

When multiple blocks of a device are connected in series, a nonlinear function can be generated, such as the input signal from the preceding block, which is then transformed into a nonlinear function for the subsequent block, and so forth. The result is a complex non-linear function (complex differential equation).

When configuration changes of a specific device (system) or parameter variations, or functional relationships among the constituent structural blocks are made, the entire specified algorithm is required once more. Block signals can be either deterministic or random.

At the same time, finding the response of a device, even by numerical methods, is usually associated with the organization of a lengthy and highly individual calculation algorithm for each version of the device (system).

2. MATERIALS AND METHODS

2.1 Analysis of acousto-electric devices and systems based on continuous broken-line functions

The use of continuous piecewise-linear functions (CPLF) allows to significantly simplify the analysis of the hardware component of the devices and systems (Vasilyev et al, 2012, 2013, 2020, 2022; Kuzichkin et al, 2019; Kurilov, Vasilyev & Kharchuk, 2010; Kurilov et al, 2016; Vasilyev, Kuzichkin & Surzhik, 2021; Surzhik, Vasilyev & Kuzichkin, 2020, 2021; Kuzichkin, Vasilyev & Surzhik, 2020).

Because CPLF are linear functions described by corresponding straight-line equations, they are effective for the analysis of devices and systems. Consequently, the CPLF represents a piecewise-linear function, but defined over the entire range of possible changes in the argument, that is, from $-\infty$ to $+\infty$.

This attribute enables the linearization of the nonlinear characteristics of devices (systems) without domain partitioning of argument variation into separate sections and the approximated non-linear characteristics. This enables the study of devices without limiting the scope of analysis by separating argument change into distinct sections for linearization. It is feasible to transition from non-linear equations that describe the behavior of devices to linear equations that are valid in all areas of modifications in device parameters, signals, and destabilizing factors. In this case, the response of devices (systems) to a specific impact can be analytically represented as a set of transfer functions.

After approximation based on CPLF, the equations of both individual units and the devices and systems as a whole, in general form, are linear differential equations of random order. Such equations do not have a general solution. However, the use of CPLF allows approximating this equation, to reduce it to an equivalent first-order equation and to obtain its solution in the form of an analytical expression (Vasilyev et al, 2012; Kurilov et al, 2016; Kurilov, Vasilyev & Kharchuk, 2010).

This work seeks to develop a method for analyzing devices intended for acoustic-location and acoustic measuring devices and systems, based on structural approximation.

2.2 Analysis of acousto-electric devices and systems based on continuous broken-line functions

Research has demonstrated that the method of structural approximation developed by the authors is an effective method for analyzing devices and systems.
The formation and processing of electrical signals in various devices and systems can be considered as amplitude-phase conversion of the signal. This circumstance allows generating and processing signals in various blocks of devices and systems to be considered as a process of serial parallel amplitude-phase transformations and its structural approximation in the form of a circuit of serial parallel-connected amplitude-phase signal converters.

This methodology enables the elimination of the subsequent steps during the examination of specific devices: generation of equations; approximation of characteristics of nonlinear elements; linearization of equations (if necessary); finding analytical expressions for studying stability; solving equations (in particular, in the form of transfer functions for given signals and destabilizing influences); finding analytical expressions for dynamic, frequency and noise characteristics. In this case, configuration changes of a specific device, as well as changes in the characteristics and parameters of the blocks and units that make up the device, are taken into account by simply changing the values of the coefficients in the finite expressions of the converter.

In order to guarantee the unification of analysis processes, devices, and systems, the creation of a generalized SC scheme is executed by employing a minimal number of identical blocks, with the possibility of their random quantity increment. Furthermore, the integrity of existing direct and cross connections between blocks is maintained. Similarly, the generalized scheme (Kurilov, 2006) contains a random number of interdependent channels - “lines”, which form a “frame”. Each line represents a serial parallel connection of folded zero-disclosure level converters. Each of these converters can be disclosed to the level of a random number, as it includes converters that are similar to it. The number of “lines” in a frame is also not limited and is determined only by the circuit of a specific structurally approximated device (system).

By the converter of the zero level of disclosure we refer to a functional block of approximated specific device or system that is not covered by back (or cross) connections. The transmission path of an acoustic-location system in general or a receiving analyzer of noise signals or alternative blocks that are not interconnected through back action is an example.

Figures 1 and 2 show generalized circuits of an amplitude-phase signal converter (SC) of zero and first levels of disclosure.

The following designations are used in the diagrams: SC – signal converter, AET – acoustical-electrical signal transducer, LA – linear adder, OD – operating device, EAC – electroacoustic converter, PP – processing path, WD – weight distributor.

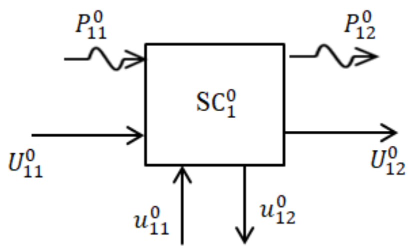

Figure 1. Generalized diagram of an amplitude-phase converter of zero-disclosure level signals

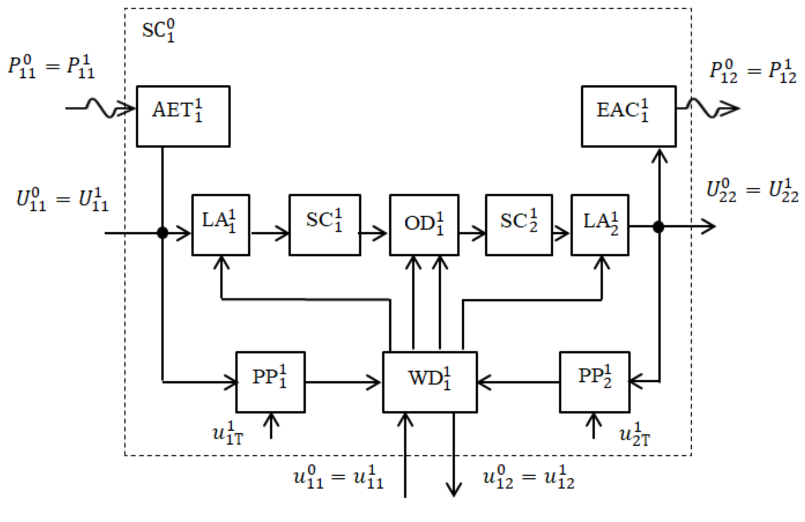

Figure 2. Generalized diagram of an amplitude-phase converter of first-disclosure level signals

The blocks are designated according to the following principle in the diagrams: $SC_{x_2}^{x_1}$ includes x_1 – disclosure level number, x_2 – block number. Given the substantial number of disclosed levels of converters, it is feasible to employ three-index designations of converters and their constituent blocks. - $SC_{x_2x_3}^{x_1}$, where x_1 – disclosure level number, x_2 – number of the block being disclosed, x_3 – block number in the disclosed diagram.

The AET and EAC blocks structurally approximate the input and output acoustic transducers. These can be either a microphone or a sonic transmitter, which can be single devices or complex systems. Depending on the type of transducer and the degree of detail of the structural approximation process, this may be either the entire transducer (such as a loudspeaker or a phased acoustic array) or a distinct component of the transducer (such as an element of a phased acoustic array).

In the OD, in a generic form, the control of the amplitude and/or phase of the electrical signal is regulated. In order to provide distinct and independent control over amplitude and phase, the block is equipped with two control inputs, which are depicted at the bottom of the OD in the diagram. Furthermore, the OD is capable of providing autonomous control over amplitude-frequency and phase-frequency characteristics, and it can also generate predetermined values for amplitude and phase signal conversions. All of the listed characteristics of the block are taken into account in the generalized transfer function of the OD.

Paths PP_1^1 and PP_2^1, respectively, implement the principles of disturbance and proportional controls. Combination control is implemented when the paths are switched on simultaneously.

Processing path (Figure 3) contains series-connected function converter (FC) and filter (F).

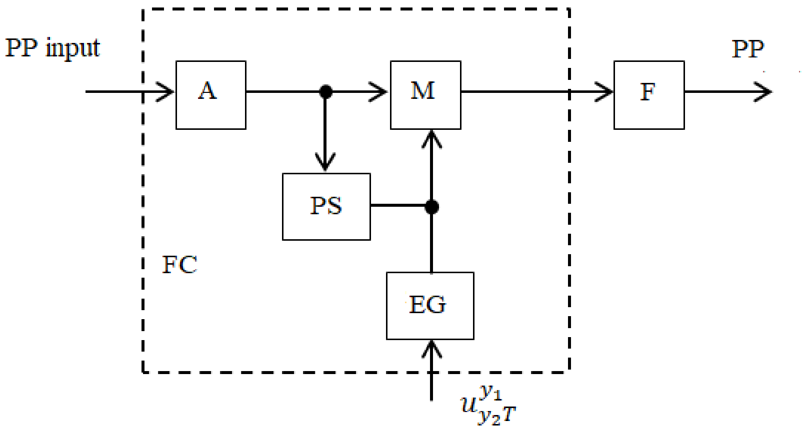

Figure 3. Processing path

The function converter includes an amplifier (A), a multiplier (M), a phase shifter (PS) and an equivalent generator (EG).

The transfer function of the amplifier in the general case is complex in nature and has the form of

$$\dot{H}_A = \dot{K}_A(U_A) + K_{A0},$$

where $\dot{K}_A(U_A)$– transfer constant of A, U_A – arriving signal of PP, K_{A0} – constant transfer ratio of A.

When assignment $\dot{K}_A(U_A) = 0$, the PP input is disabled and the EG signal is sent to the output of the path through the filter. In this case, the multiplier acts as a scale converter of the EG signal. The scale of conversation is determined by the value of the coefficient K_{A0}. If $K_{A0} < 1$, then M is the voltage divisor of the EG signal. If $K_{A0} > 1$, then M implements a linear amplification of the EG signal by K_{A0} times. When $K_{A0} = 1$, the EG signal passes to the F input without scaling conversion.

The PS transfer function has the form

$$\dot{H}_{PS} = \dot{F}_{PS}(U_A) + F_{PS0},$$

where $\dot{F}_{PS}(U_A)$ – complex component of the transfer function, in the general case depending on the parameters of the output signal of the amplifier U_A; F_{PS0} – continuous component of the transfer function PS. The component $\dot{F}_{PS}(U_A)$ in general is the signal frequency function U_A and provides, together with M, an approximation of the frequency detection process.

When $\dot{F}_{PS}(U_A) = 0$, $F_{PS0} \neq 0$ and the EG signal is turned off, at the M output we obtain the square of the input signal PP with a scaling transformation. In particular, this makes it possible to structurally approximate devices and processes of asynchronous amplitude detection.

The EG block has an primary input to which the signal $u_{y_2T}^{y_1}$ can be applied, where y_1, y_2 and T are the SC identifiers.

The following components can be structurally approximated as an EG: an unregulated active oscillator; an active oscillator with quartz and other stabilization; a voltage-controlled generator; a synchronizing generator with extrinsic stabilization; a trigger device (frequency demultiplier); a scale signal converter, as well as signal phase shifter $u_{y_2T}^{y_1}$.

The EG transfer function can be written similarly

$$\dot{H}_\Gamma = \dot{\mathrm{E}}_\Gamma\left(u_{y_2T}^{y_1}\right) + \mathrm{E}_{\Gamma 0},$$

where $\dot{\mathrm{E}}_\Gamma\left(u_{y_2T}^{y_1}\right)$ – complex component of transfer function EG, $\mathrm{E}_{\Gamma 0}$– continuous component of transfer function. As earlier $\dot{\mathrm{E}}_\Gamma\left(u_{y_2T}^{y_1}\right)$ describes the above functions of the EG, and $\mathrm{E}_{\Gamma 0}$ allows under $\dot{H}_{PS} = 0$, additionally carry out large-scale transformations of the input signal PP.

The EG, together with a multiplier, also allows to structurally approximate the modulation, demodulation and frequency conversion devices (heterodyne mixing). This PP scheme provides the ability to structurally approximate devices that provide the following signal transformations: large-scale signal transformations - amplification, voltage division; frequency conversion with a decrease or increase in signal frequency; nonlinear frequency conversion without heterodyne mixing; signal frequency division; amplitude, balance, phase and frequency modulation; amplitude non-synchronous and amplitude synchronous detection; phase and frequency signal detection.

The processing path filter is generally active and has a random order. This makes it possible to structurally approximate and mathematically describe both passive filtering devices, regardless of the type of their physical implementation, and active filters on operational amplifiers, as well as selective devices (for example, a band amplifier on transistors, etc.).

Furthermore, PP facilitates the structural approximation and description of the paths for generating control signals of all known automatic control systems, as well as the automatic adjustment of amplitude (power, amplification), phase, signal frequency, and other automatic control systems utilized in acoustic-location systems.

The presence of two paths in the signal converter circuit enables the implementation of control principles (automatic adjustment), including disturbance control, proportional control, deviation, and the principle of combination control. Additionally, direct, reverse, and cross connections can be formed within the approximating SC scheme.

The WD block distributes signals from each of its inputs to each output, with specified proportions. This enables to switch the corresponding inputs and outputs of the WD, as well as to carry out summation (subtraction) and subdivision of electrical signals with specified proportions - specified weight coefficients.

In general, the WD coefficients are complex functions. This provides the ability to structurally approximate and analyze real devices that have inertia.

When performing structural approximation and analysis of devices and systems based on SC of zero level of disclosure, the utilization of WD makes the summation and (or) branching signals from both separate converters and groups of converters, thereby forming the required structural approximating circuit.

The system of equations for the WD output signals has the form

$$\begin{cases} y_1 = n_{11}x_1 + n_{21}x_2 + \cdots + n_{n1}x_n + n_{N1}x_N, \\ y_2 = n_{12}x_1 + n_{22}x_2 + \cdots + n_{n2}x_n + n_{N2}x_N, \\ y_3 = n_{13}x_1 + n_{23}x_2 + \cdots + n_{n3}x_n + n_{N3}x_N, \\ \quad \cdots \\ y_M = n_{1M}x_1 + n_{2M}x_2 + \cdots + n_{nM}x_n + n_{NM}x_N, \end{cases}$$

where y_m – output signal m, $m \in [1;M]$, M – maximum output pin number, x_m – input signal n, $n \in [1;N]$, N – maximum input pin number, n_{nm} – transfer constant WD from the input n to the output m.

For a WD with random numbers of inputs and outputs, the expression for the signal at a random output m, in accordance with the system of equations, becomes the form

$$y_m = \sum_{n=1}^{N} n_{nm}x_n.$$

Special cases of WD are a linear adder and a linear splitter. In this case, the values of the WD transmission coefficients are respectively equal to n_{n1}=1 and n_{m2}=1.

The symbols P and U indicate the external main acoustic and electrical signals, respectively. Symbols u indicate ancillary electrical signals.

The superscript and subscript of signals are designated as $U_{y_2y_3}^{y_1}$, where $y_1$is the level number, y_2 is the number of the block to which the signal belongs, y_3 is the signal identifier. This notation system enables the compact and structural identification of signals belonging to a specific level and SC block, as well as the identification of approximating blocks and signals with a random number of them.

The signal identifier y_3 can take numeric or alphabetic notations. Numerical: 1 - block input signal; 2 - output signal. If a particular block has more than one input and/or more than one output, odd indices correspond to input signals (1;3;5,7...), and even indices correspond to output signals (2;4;6,8...). Alphabetic symbols indicate the type (character) of the corresponding block: T – signal of the processing path, etc.

Practice shows that for approximation of devices and systems it is enough that $y_1 \in [0; 9]$ and $y_2 \in [0; 9]$. If $y_1 > 9$ is required (two or three-digit numbers), then the values y_2 and y_3 are separated by a dot (for example, 12.1).

The arrows "→" show the direction and indicate the ideal path for the electrical signal. That is, the active and reactive components of the resistance of this path are equal to zero. If necessary, the parameters of real paths are taken into account in the transfer functions of the corresponding blocks, or by introducing the corresponding blocks during structural approximation. The arrows " ⁓→ " show the conditional direction of the spatial acoustic signal - to the input of the block or from the output of the block. The beginning and end of the arrow correspond to one geometric point in space. The acoustic signal from the beginning to the end of the arrow is transmitted instantly, without changing the power, phase, polarization parameter and propagation direction.

The SC diagram presented in Figure 2 allows to structurally approximate and analytically describe the processes of transformation (formation and processing) of signals in the devices and systems of the above-mentioned acoustic-location and monitoring directions. The number of approximated channels is not limited and is solely determined by the number of unified SCs used, which enables to consider and approximate, for example, an acoustic phased array, as a single system or as a set of individual elements, each of which has its own characteristics, etc.

3. RESULTS

3.1 Disclosure levels of converter circuit

Let's consider several levels of disclosure of the converter circuit. The zero level of SC disclosure is shown in Figure 1, which corresponds to a folded converter. Level 1 (Figure 2) is the basic SC. Its further disclosure allows for the random increase in the number of signal transformations in the "line". Each disclosure of the subsequent level is carried out by opening in the circuit (Figure 2) the folded converters (in this case SC_1^1 and SC_2^1) contained in the previous level.

The next level of disclosure is presented in Figure 4. To reduce the size of the circuit, only SC_1^1 is disclosed here and only the electrical signal path is shown.

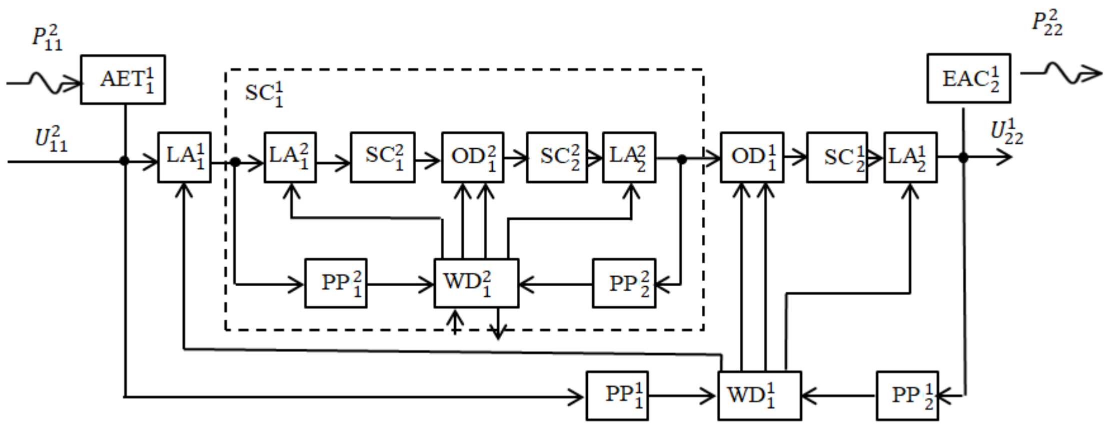

Figure 4. Generalized diagram of an amplitude-phase converter of second-disclosure level signals

Let us set the transfer functions of the blocks: SC → Π, OD → К, WD → n, PP → W. Let the superscripts and subscripts of the functions fully correspond to the upper and lower indices of their blocks. In accordance with Figures 1, 2 and 3, transfer functions of SC of various levels are obtained by multiplying the transfer functions of the constituent blocks and have the form:

level 0 $\mathrm{Q}^0 = \Pi_1^0$ – folded SC;
level 1 $\mathrm{Q}^1 = \Pi_1^1 К_1^1 \Pi_2^1$;
level 2 $\mathrm{Q}^2 = \Pi_1^2 К_1^2 \Pi_2^2 К_1^1 \Pi_3^2 К_2^2 \Pi_4^2$;
level 3
$\mathrm{Q}^3 = \Pi_1^3 К_1^3 \Pi_2^3 К_1^2 \Pi_3^3 К_2^3 \Pi_4^3 К_1^1 \Pi_5^3 К_3^3 \Pi_5^3 К_3^3 \Pi_6^3 К_2^2 \Pi_7^3 К_4^3 \Pi_8^3$.

General expression of the transfer function for a random number of disclosed levels $A \geq 1$ is

$$Q^{\mathrm{A}} = \prod_{\alpha=1}^{\mathrm{A}} \prod_{\beta=1}^{\mathrm{B}} \prod_{\gamma=1}^{\mathrm{G}} \Pi_\beta^\alpha К_\gamma^\alpha,$$

where α - current number of the layer being disclosed; A - maximum number of layers disclosed in the SC; β, γ - coefficients (transfer function numbers) respectively SC and OD; B= 2α, G= 2α-1 – maximum values β, γ,.

A generalized conversion diagram containing M “lines” is presented in Figure 5. The right upper indices of blocks, signals and transfer functions correspond to the “line” number.

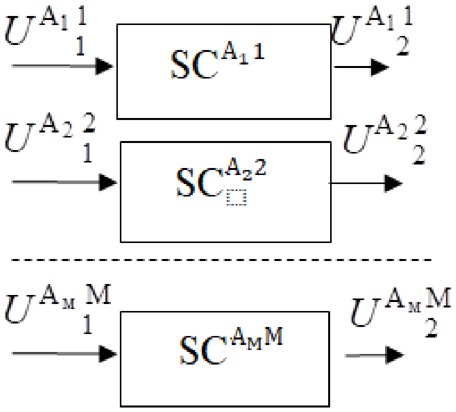

Figure 5. Generalized conversion diagram in M “lines

We denote the sets of input and output signals through the matrices $\boldsymbol{U}_{1,2}^{\mathrm{A_M M}}$, and through the matrix $Q^{\mathrm{A_M M}}$ - the set of transfer functions. Then the generalized scheme is described by the system of equations

$$\boldsymbol{U}_2^{\mathrm{A_M M}} = \boldsymbol{Q}^{\mathrm{A_M M}} \boldsymbol{U}_1^{\mathrm{A_M M}}.$$

Similar to the process of matrix generation, representing the circuit of a specific device (system) comes down to specifying the transfer characteristics in the corresponding element of the “matrix” of the generalized chart.

Let us perform a structural approximation of the electrical part, for example, of the transmission path of an atmosphere acoustic sounding system with automatic control of the output signal level (Figure 6a). The following designations are used here: DO - drive oscillator of the transmitting path, PSC - probing signal conditioner (modulating equipment), PA - prime amplifier, CPA - controlled power amplifier, POA - power output amplifier, D, F and DCA - detector, filter and direct current amplifier of the system for automatic control of the output signal level, u_{m} – shaping (modulating) signal, E_n – supply voltage.

In order to reduce the length of the figure, we will limit ourselves to approximating only the electrical path of the transmitter. In addition, since there are no general direct, reverse and cross connections in the circuit, we will exclude adder units and other unused SC blocks, as well as communication lines connecting the excluded blocks. When conducting an analysis of a circuit, this is ensured automatically (programmatically or manually), by setting the values (setting equal to zero or to one) of the corresponding coefficients in the expressions of the transfer functions of these SC blocks.

An example of a structural approximation of the electrical part of the transmission path is presented in Figure 6b.

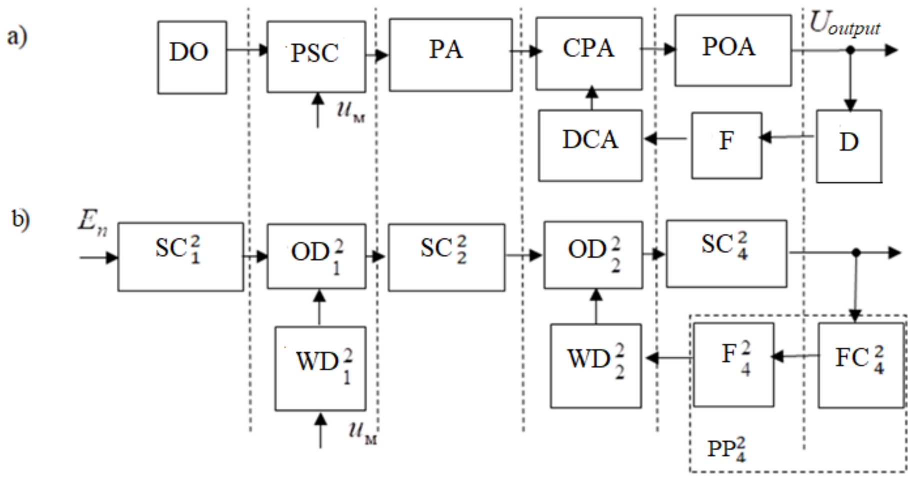

Figure 6. Structural approximation of the transmission path of an atmosphere acoustic sounding system with automatic control of the output signal level

Let us set $U_g = U_m cos\omega t$ – output signal of the generator, U_m – amplitude, ω – frequency of the DO, t – time; $\mathrm{К_M}$, $\mathrm{К_{PA}}$, $\mathrm{К_{CPA}}$, $\mathrm{К_{POA}}$, $\mathrm{D_A}$, $\mathrm{M_A}$, n_{A} – transmission coefficients, respectively, of the PSC, controlled and output power amplifiers, detector, filter and direct current amplifier of the automatic level control system; F_4^2 and FC_4^2 – filter and function converter PP_4^2. In this case, FC_4^2 carries out the function of amplitude detection of the output signal of the POA and its transfer function is D_4^2.

Then the transfer functions of the blocks of the generalized circuit is: $\Pi_1^2 = (U_m/E_n)cos\omega t$, $\mathrm{К}_1^2 = \mathrm{К_M}$, $\Pi_2^2 = \mathrm{К}_{PA}$, $\mathrm{К}_2^2 = \mathrm{К}_{CPA}$, $\Pi_4^2 = \mathrm{К}_{POA}$, $\mathrm{К}_1^2 = \mathrm{К_M}$, $\mathrm{К}_1^1 = \Pi_3^2 = n_1^2 = 1$, $W_1^1 = W_2^1 = W_1^2 = W_3^2 = 0$, $\mathrm{D}_4^2 = \mathrm{D_A}$, $\mathrm{M}_4^2 = \mathrm{M_A}$, $n_{2\mathrm{HY}}^2 = n_{\mathrm{A}}$.

Their substitution into the end transfer functions of the converter allows: to immediately obtain expressions of the transfer functions of the transmission path for the studied influences; to investigate the stability of the circuit; to obtain analytical expressions for dynamic, amplitude-frequency, modulation and other characteristics of a particular device; to conduct calculation, analysis and research of these characteristics; to optimize these characteristics based on their specific technical requirements for the path.

For the structural approximation of the transmission path, one "line" and two levels of SC disclosure were required.

3.2 Transfer functions of a nonlinear signal converter

Let us obtain the transfer functions of the nonlinear converter (Figure 2). For clarity, let's consider a SC option with the following parameters: transfer functions AET_1^1, EAC_1^1, SC_1^1 and SC_2^1 are equal to 1; transmission coefficients WD_1^1 for summarizer S_1^1 and S_2^1 are equal to 0; control OD_1^1 is carried out via one channel and $u_{11}^0 = u_{12}^0 = 0$. To reduce the length of the recording, let us denote $u_{1\text{T}}^1 = x_{\Gamma 1}$ and $u_{2\text{T}}^1 = x_{\Gamma 2}$.

Let us set the parameter of the input electrical or acoustic signal as x, and the parameter of the output electrical or acoustic signal as y. In addition, let OD_1^1 be affected by the destabilizing factor ε.

The output signals of the first (FC_1) and second (FC_2) functional converters of the PP_1^1 and PP_2^1 paths are functions of the differences in the parameters of their input and reference signals $d_1=x-x_{\Gamma 1}$, $d_2=y-x_{\Gamma 2}$. Let us denote the characteristics of functional converters PP_1^1 and PP_2^1 as $\widetilde{F_1}(d_1)=\widetilde{F_1}(x-x_{\Gamma 1})$ and $\widetilde{F_2}(d_2)=\widetilde{F_2}(y-x_{\Gamma 2})$ respectively.

Control signal equation OD_1^1, in case of deviations x and $x_{\Gamma 1,2}$

$$u = n_1 M_1(p)\tilde{F}_1\,(x\text{-}x_{\Gamma 1}) + n_2 M_2(p)\tilde{F}_2\,(y\text{-}x_{\Gamma 2}),$$

where $n_{1,2}$ – WD transmission coefficients for disturbance control and proportional control circuits, respectively; $M_{1,2}(p)$ – filter gains PP_1^1 and PP_2^1; p=d/dt – operator.

Let us denote $\widetilde{K}(u_p)$ – transfer function OD_1^1 taking into account the impact of the destabilizing factor, where $u_p=u+\varepsilon$. Then the SC equation for the output signal parameter y will take the form

$$y = x - \widetilde{K}[n_1 M_1(p)\tilde{F}_1\,(x\text{-}x_{\Gamma 1}) + n_2 M_2(p)\tilde{F}_2\,(y\text{-}x_{\Gamma 2}) + \varepsilon]. \quad (1)$$

The characteristics OD_1^1 и $FC_{1,2}$ we approximate using CPLF. Let's denote them accordingly $K(u+\varepsilon)$ and $F_{1,2}(x-x_{\Gamma 1,2})$

$$K(u+\varepsilon) = \sum_{m=0}^{M-1}[K_m(u+\varepsilon) + \text{B}_m]Q_m,$$
$$F_1(x - x_{\Gamma 1}) = \sum_{n=0}^{N-1}[K_{1n}(x - x_{\Gamma 1}) + \text{B}_{1n}]Q_{1n},$$
$$F_2(y - x_{\Gamma 2}) = \sum_{n=0}^{N-1}[K_{2n}(y - x_{\Gamma 2}) + \text{B}_{2n}]Q_{2n}, \quad (2)$$

where $K_m = \frac{\widetilde{K}(U_{\text{p}m+1}) - \widetilde{K}(U_{\text{p}m})}{\Delta_u}$, $K_{1n} = \frac{\tilde{F}_1(\text{D}_{1n+1}) - \tilde{F}_1(\text{D}_{1m})}{\Delta_{d1}}$, $K_{2n} = \frac{\tilde{F}_2(\text{D}_{2n+1}) - \tilde{F}_2(\text{D}_2 n)}{\Delta_{d2}}$ and $B_m = \widetilde{K}(U_{\text{p}m} - K_m U_{\text{p}m}$, $B_{1n} = \tilde{F}_1(D_{1n}) - K_{1n}D_{1n}$, $B_{2n} = \tilde{F}_2(D_{2n}) - K_{2n}D_{2n}$– coefficients of line segments, approximating characteristics OD_1^1 , FC_1 , FC_2 at approximation nodes m and n; M and N – maximum numbers of approximation nodes; $U_{\text{p}m}$ and D_{1n}, – values of u_p and d_1 d_2 at approximation nodes m and n respectively; $\Delta_{u,d1,d2}$ – variable approximation step u_p, d_1, d_2, $Q_m = Q_m(u+\varepsilon)$, $Q_{1n} = Q_{1n}(x - x_{\Gamma 1})$, $Q_{2n} = Q_{2n}(\text{y} - x_{\Gamma 2})$ – functions for including segments of approximating straight lines.

The inclusion function Q_m or $Q_{1,2n}$ are different from zero and equal to one only on the interval between nodes m and m+1 or between nodes n and n+1 respectively

$$Q_m(u+\varepsilon) = \frac{1}{2\Delta}\sum_{\lambda=0}^{1}\sum_{\gamma=0}^{1}(-1)^{\lambda+\gamma}\left|u + \varepsilon - U_{pm} - \gamma\Delta_u + \Delta(1-\lambda)\right|,$$
$$Q_{1n}(x - x_{\Gamma 1}) = \frac{1}{2\Delta}\sum_{\lambda=0}^{1}\sum_{\gamma=0}^{1}(-1)^{\lambda+\gamma}|x - x_{\Gamma 1} - D_{1n} - \gamma\Delta_{d1} + \Delta(1-\lambda)|, \quad (3)$$
$$Q_{2n}(y - x_{\Gamma 2}) = \frac{1}{2\Delta}\sum_{\lambda=0}^{1}\sum_{\gamma=0}^{1}(-1)^{\lambda+\gamma}|y - x_{\Gamma 1} - D_{2n} - \gamma\Delta_{d2} + \Delta(1-\lambda)|,$$

where Δ – arbitrary small quantity ($\Delta \to 0$), λ and γ – whole numbers,

$$u = n_1 M_1(p)\sum_{n=0}^{N-1}[K_{1n}(x - x_{\Gamma 1}) + B_{1n}]Q_{1n} + n_2 M_2(p)\sum_{n=0}^{N-1}[K_{2n}(y - x_{\Gamma 2}) + B_{2n}]Q_{2n}. \quad (4)$$

Then the converter equation for changes in the parameters of the input signal, parameters of the reference signals and the destabilizing factor is

$$\text{y} = x - \sum_{m=0}^{M-1}[K_m(u+\varepsilon) + B_m]Q_m. \quad (5)$$

Let us substitute (4) into (5), introduce the notation

$$N_{1mn} = \sum_{m=0}^{M-1}\sum_{n=0}^{N-1} K_m K_{1n} Q_m Q_{1n}\,, N_{2mn} = \sum_{m=0}^{M-1}\sum_{n=0}^{N-1} K_m K_{2n} Q_m Q_{2n}\,, N_m = \sum_{m=0}^{M-1} K_m Q_m\,,$$
$$R_{1mn} = \sum_{m=0}^{M-1}\sum_{n=0}^{N-1} K_m B_{1n} Q_m Q_{1n}\,, R_{2mn} = \sum_{m=0}^{M-1}\sum_{n=0}^{N-1} K_m B_{2n} Q_m Q_{2n}\,, E_m = \sum_{m=0}^{M-1} B_m Q_m.$$

and after transformation (5) we obtain

$$\begin{gathered} \mathrm{y} = \{[1 - n_1 M_1(p) N_{1mn}]x + n_1 M_1(p) N_{1mn} x_{\Gamma 1} + n_2 M_2(p) N_{2mn} x_{\Gamma 2} - N_m \varepsilon - \\ - n_1 M_1(p) R_{1mn} - n_2 M_2(p) R_{2mn} - E_m\}/[1 + n_2 M_2(p) N_{2mn}]. \end{gathered} \tag{6}$$

The model for the formation of the deviation of the converter output signal parameter, constructed according to (6), is presented in Figure 7.

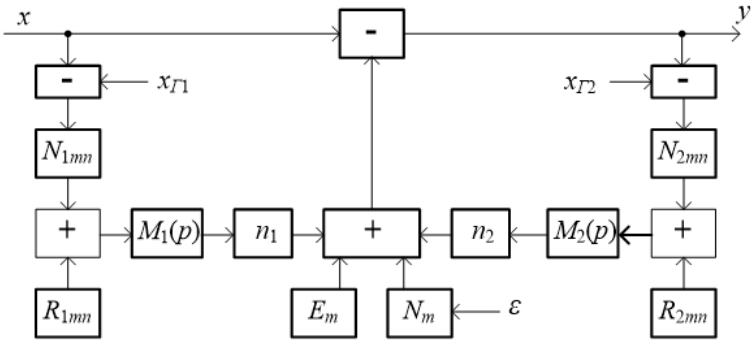

Figure 7. Model for the formation of the deviation of the converter output signal parameter

Let's transform (6)

$$\mathrm{y} = \frac{1 - n_1 M_1(p) N_{1mn}}{1 + n_2 M_2(p) N_{2mn}} x + \frac{n_1 M_1(p) N_{1mn}}{1 + n_2 M_2(p) N_{2mn}} x_{\Gamma 1} + \frac{n_2 M_2(p) N_{2mn}}{1 + n_2 M_2(p) N_{2mn}} x_{\Gamma 2} - \\ - \frac{N_m}{1 + n_2 M_2(p) N_{2mn}} \varepsilon - \frac{n_1 M_1(p) R_{1mn} + n_2 M_2(p) N_{2mn} + E_m}{1 + n_2 M_2(p) N_{2mn}}. \tag{7}$$

The transfer functions of a nonlinear signal converter with unit characteristics approximated by CPLF have the form $K_{\alpha\beta} = \frac{\beta}{\alpha}$, where α and β are the impact and response of the converter, respectively. Unlike the transfer functions of a linearized system, these functions are valid not only in the neighborhood of the point of stationary mode, but also for any changes in α and β.

As follows from (7), the expressions of the transfer functions of the converter with combination control take the form

$$K_{xy} = \frac{1 - n_1 M_1(p) N_{1mn}}{1 + n_2 M_2(p) N_{2mn}},\ K_{x_{\Gamma 1} y} = \frac{n_1 M_1(p) N_{1mn}}{1 + n_2 M_2(p) N_{2mn}}, \\ K_{x_{\Gamma 2} y} = \frac{n_2 M_2(p) N_{2mn}}{1 + n_2 M_2(p) N_{2mn}},\ K_{\varepsilon y} = \frac{N_m}{1 + n_2 M_2(p) N_{2mn}}. \tag{8}$$

Let us denote by the symbol G the deviation of the parameter y due to the influence of the constant components of the characteristics of the converter units

$$G = \frac{n_1 M_1(p) R_{1mn} + n_2 M_2(p) N_{2mn} + E_m}{1 + n_2 M_2(p) N_{2mn}}. \tag{9}$$

Let us transform (7) taking into account (8) and (9)

$$y = K_{xy} x + K_{x_{\Gamma 1} y} x_{\Gamma 1} + K_{x_{\Gamma 2} y} x_{\Gamma 2} + K_{\varepsilon y} \varepsilon - G. \tag{10}$$

Expression (10) fully describes the static and dynamic modes of a nonlinear SC with combination control. This is a piecewise-linear differential equation that holds validity for arbitrary values and types of input influence, for arbitrary deviations of signal parameters and arbitrary types of signals from reference generators and the destabilizing factor, as well as for arbitrary characteristics of the converter units and characteristics of path filters PP_1^1 and PP_2^1.

In order to derive analytical expressions for specific characteristics, it is enough to substitute the values of the acting disturbances and the transfer functions of the filters into (10).

For example, let us consider the stationary mode of the converter under the influence of a deviation in the parameter of the reference signal FC_1 ($x=x_{\Gamma 2}=\varepsilon=0$). The stationary mode corresponds to the end of all transient processes in the system ($p \to 0$).

Let us denote the values of the transfer functions of the filters in the stationary mode as $M_{1,2}(p)=M_{1,2}(0)=\gamma_{1,2}$, and here we denote the values of quasi-static deviations of the parameters by superscripts "*". Then, after appropriate substitution in (10) and transformation, the equation for the stationary mode of the signal converter will take the form

$$y^* = \frac{\sum_{m=0}^{M-1}\left\{K_m \sum_{n=0}^{N-1}[(n_1\gamma_1K_{1n}x_{\Gamma 1}^* - \mathrm{B}_{1n})Q_{1n} - n_2\gamma_2\mathrm{B}_{2n}Q_{2n}] - B_m\right\}Q_m}{1+n_2\gamma_2\sum_{m=0}^{M-1}\sum_{n=0}^{N-1}K_mK_{2n}Q_mQ_{2n}}. \quad (11)$$

4. DISCUSSION

The numerical evaluation of diagrams of the static characteristics of the converter with specific approximated characteristics of the units can be carried out using (10) or directly substituting the numerical values of the approximation coefficients and the actuating quantities $x_{\Gamma 1}^*$ into (11). For options for build-up a converter with proportional control or combination control, due to back action, when calculating Q_{2n}, a delay of one calculation step should be introduced.

The investigation of specific devices and systems is conducted by utilizing the transfer functions of a generalized circuit. The utilization of CPLF enables the investigation of devices (systems) of free structure with optional non-linear characteristics and random types of inertial component of blocks and links. The study of specific devices and systems is carried out based on finite expressions of the transfer functions of a generalized signal converter connection.

Using this method, the computation processes when studying devices and systems are simple and do not require specialized computing devices and programs.

5. CONCLUSION

A new method of analysis and study of devices and systems intended for acoustic measurements is proposed, and the developed method is justified. The method involves carrying out a structural approximation of the DS based on the developed generalized circuit.

The suggested method enables the creation of a specialized software environment (software shell), which can significantly simplify and automate the processes of complex analysis and research of arbitrary acousto-electric devices.

When performing research, changing parameter and characteristics are entered programmatically, and the desired output data is calculated. The studied graphs, tables and diagrams of the DS with specific parameters are constructed.

Simultaneously, renowned software environments such as MicroCap, OrCad, and others can be utilized as support programs, for instance, to calculate the nonlinear characteristics of specific blocks and units of the DS or to calculate the characteristics of the influence of specific destabilizing factors on a specific unit.

On the basis of the developed method, it is possible to synthesize acousto-electric devices at the analytical and structural levels. As nonlinear differential equations of random order that describe the response of devices and systems can be reduced to linear equations of the first order in the general case, solving the inverse problem (synthesis) is not a challenging task, both at the analytical and structural levels.

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List of abbreviations and designations

A – amplifier
AET – acoustical-electrical signal transducer
CPA - controlled power amplifier
D - detector
DCA - direct current amplifier
DO - drive oscillator
DS – devices and systems
EAC – electroacoustic converter
EDS – equations of devices and systems
EG –equivalent generator
F – filter
FC – function converter
HSC – hardware and software complex
LA – linear adder
M – multiplier
OD – operating device
PA - prime amplifier
POA - power output amplifier
PP – processing path
PS – phase shifter
PSC - probing signal conditioner
SC – signal converter
WD – weight distributor

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