The Method of Correlation Investigation of Acoustic Signals with Priority Placement of Microphones

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Abstract: Reasonable benefits of using multi-channel structure of digital correlation with high priority placement of one of the microphones for calculation autocorrelation. Peculiarities and effectiveness for application of CAD for designing of correlational back-end processors on the crystals of programmable logical integrated circuits have been determined.

Keywords: design, correlation, acoustics.

I. INTRODUCTION

Development of theoretical foundations of information technology and software - hardware correlation signal processing is actual scientific - applied problem to be solved in many industries. Identification of sources of acoustic signals (SAS) relatively to spatial placement of microphones - receivers of acoustic signals (RAS) is also included.

An example of the use of acoustic signals is sound intelligence, which is a part and a type of artillery reconnaissance. Methods and devices used in sound intelligence allow to determine the coordinates of artillery and mortar batteries of the enemy by the sound of their shots and adjust their artillery fire defining the actual shells and mines hit by the sound of explosions.

The advantages of this type of intelligence are the independence of visibility conditions due to which exploration and maintenance of fire by sound are possible at night, in fog, in the smoke, weak dependence on terrain and local items, allowing exploration in of forest and cross country, and in the mountain as well, the ability to conduct hidden reconnaissance continuously for a long time.

The given problem is a primary-industry-technical task of the special technique [1], [2], [3].

Analysis of the known research results. An example of a successful but far from optimal solution of such problem is working out localization accumulated information systems by C. Birchfield and D. Hilmar. [4], [5], [6], [7]. The determination of spatial parameters (\(\Theta\)) azimuth and distance to the SAS (\(\Phi\)) the use of a certain number of (\(q\)) correlators for a given number of those (\(m\)) microphones is taken as the base of the system.

The example of the structure of Acoustic Localization by Accumulated Correlation (ALAC) system is shown in [4] (Fig. 1).

It should be noted that the number of required correlators for a given number of chaotic space placed microphones RAS is determined by the condition of symmetry correlation matrix (1)

\[
MK = \begin{bmatrix}
1 & 2 & 3 & 4 \\
1 & R_{12} & R_{13} & R_{14} \\
2 & R_{21} & 1 & R_{23} & R_{24} \\
3 & R_{31} & R_{32} & 1 & R_{34} \\
4 & R_{41} & R_{42} & R_{43} & 1
\end{bmatrix}
\]

where \(R_{ii} = 1\), are placed along the diagonal;
\(R_{ij} = R_{ji}\) – are identical, according to the symmetry of the matrix (1).

So, according to the information technology of the correlation processing of the SAS signals, proposed in [4], the required amount of (\(q\)) correlators for a given number of the (\(m\)) microphones is determined by the expression:

\[
q = \frac{(m^2 - m)}{2}. \tag{2}
\]

The graph of dependency (2) with different numbers of microphones is shown in Fig.2.

For implementation of each correlator in ALAC system the integrated assessment multiplicative correlation function is used by the expression:

\[
L(q) = G\left(\int_{\tau_{i,j}}^{\tau_{(q)}} \frac{x_i(t) \times x_j(t - \tau_{i,j}) + \tau_{j,q})dt}{2}\right) + \alpha V_x
\]

where: \(q\) – the identifier of the SAS; \(G\) – the integrated cross-correlation function; \(x_i(t)\) and \(x_j(t - \tau_{i,j})\) – current and delayed on the time interval \(\pm \tau_{i,j}\) acoustic signals (AS)
accordingly; $\alpha V_e$ – damping energy coefficient of the cross-correlation function on the interval $\tau_{i,q}$.

The analysis of the analytic expression (3) allows to determine that the implementation of $R_{xx}$ correlators in ALAC system is based on a time delay, $x_i(t)$ multiplying and $x_j(t-\tau)$ integrating analog signals, essentially limiting its functionality, simplifying and increasing the speed and accuracy also prevents its implementation based on digital microelectronics crystals and programmable integrated-circuit logic (FPGA).

The purpose of the work is to develop and explore systemic and structural characteristics of digital special processor computing centered multiplicative correlation function.

II. MAIN PART

Formulation of the problem. In order to investigate the principles of improving and optimizing system features of digital correlators as the basic components of the discovering system, analysis of system features of special processor and computing digital multiplicative correlation estimates by the expression is performed [8]:

$$R_{xx}(j) = \frac{1}{n} \sum_{k=0}^{n} x \times x_{i-j}; \quad j \in \mathbb{Z}, m,$$

where: $R_{xx}(j)$ – is centered autocorrelation function; $x_i$ and $x_{i-j}$ – are centered digital value and analog signals $x_i(t)$ and $x_j(t-\tau); n$ – is the volume of the sample data set $\{x_i\}; m$ – is the number of points of correlation function $R_{xx}(j); j$ – is discrete digital delay unit point $x_{i-j}$ in time.

The example of the interaction in time of the digital signals $x_i$ and $x_{i-j}$, where $C$ – constant threshold and corresponding asymptotic of the correlation function $R_{xx}(j)$ is shown Fig. 3.

Fig. 2. The dependence of the required number of correlators on the number of microphones in the system

Fig. 3. Example of temporal interaction sampled in time and amplitude quantized digital signals and corresponding digital asymptotic estimates multiplicative correlation function in conventional units (c.u.)

Fig. 3 shows that at the time of coincidence the current $\tau$ digital streams until the moment $x_{i}$ of the reception of acoustic signal $x_i(t)$ by a remote microphone SAS level of correlation between the previously obtained a SAS close to
the microphone signal $x_j(t - \tau)$ to digital value $j - s$ correlator outputs do not exceed the threshold level constant $C$ and at the time of singing falling at the time fixed main lobe function $R_{xx}(j)$ and accordingly $j$ a numeric value that corresponds to the duration $\Delta t$ used for direction finding SAS. Obviously, depending on the structure of the stream of digital samples $x_i$ that reflect the analog signal generated $x(t)$ by a remote SAS in the vicinity of the main lobe $R_{xx}(j)$ functions will emerge sufficiently large lateral lobes, which require separate examination for specific researched objects that have SAS.

It shown in this paper [9] that in general for calculation the correlation function $R_{xx}(j)$ according to expression (4) based on a processing arrays of digital data $x_i$ which representing the converted ADC input analog signals $x(t)$ you should do the following:

1) define the digital evaluation expectation:

$$M_x = \frac{1}{n} \sum_{i=1}^{n} x_i;$$

2) calculate the array centered values:

$$\tilde{x}_i = x_i - M_x; \quad i \in 1, n;$$

3) calculate the variance:

$$D_x = \frac{1}{n} \sum_{i=1}^{n} (x_i - M_x)^2 = \frac{1}{n} \sum_{i=1}^{n} \tilde{x}_i^2;$$

4) calculate the correlation function $R_{xx}(j)$ centered on the expression (4);

5) calculate the normalized correlation function:

$$\rho_{xx}(j) = \frac{R_{xx}(j)}{D_x}; \quad i \in 1, m;$$

6) perform the comparison of the digital value $\rho_{xx}(j)$ which changes in boundaries $-1 \leq \rho_{xx}(j) \leq +1$, with boundary constant $0 < C \leq 1$:

$$\rho_{xx}(j) > C_0; \quad \rho_{xx}(\neq j) < C_0;$$

7) register a digital value $j$ which corresponds to the time delay $\Delta t$ of the acoustic signal $x(t)$ between two microphones placed at different distance from SAS.

This correlation algorithm of digital processing acoustic signals on a base of multiplicative function (4) is greatly simplified if before the ADC pre-differentiation analog $x(t)$ signal is performed and it is passed through a device of automatic gain control, is shown in Fig. 4.

Such pre-processing of the analog signals $x_i(t)$ and $x_j(t - \tau)$, which are formed on the outputs of the microphones allows to remove from algorithm of calculating operations (5–8) and immediately calculate correlation digital integrated assessment (4) and compare it values at all points $j$ of the constant threshold $R_{xx}(j) > C$.

As it will be shown later, the numerical value of this constant is selected due to the given parameters of digital correlator, which implements the calculation of the multiplicative function $R_{xx}(j)$. Such parameters are:

1) the number of quantization levels of analog signals $A = 2^k; \quad -\frac{A}{2} \leq x_i \leq +\frac{A}{2}$;

2) binary output bit ADC $k$

$$k = 4,8,10,12;$$

3) the sample size of the database $n = 2^r; \quad r = 4,5,6,7,8,...;$

4) the number of points correlation function $m = 2^l; \quad l = 8,10,12,...$

III. RESULTS OF THE INVESTIGATION

The basic structure of the investigated correlation processing of acoustic signals special processor designed for their processing is shown in Fig. 5.

The following notions are used in Fig.5: $M_i; \quad M_j$ – RAS microphones; $CS$ – conditioners signals; $ADC$ – analog-to-digital converter; $RG_{0}...RG_{m}$ – multi bit shift register source $\pm x_i; \quad X$ – digital binary multiplier; $\pm \sum$ – reversing saving up adder; $\geq$ – module comparing numerical values of the correlation function $R_{xx}(j)$ threshold constant $C$; encoder – encoder codes Haar – Rademacher.
The work of the digital correlator is as follows. Acoustic signals which are taken by microphone $M_j$ and with a certain time $M_j$ lag are converted to electrical signals which passing through conditioners signals (CS) are filtered, normalized on power and enter the inputs of the first (ADC1) and second (ADC2) alternating analog - digital converters parallel type.

Output $k$ – bit binaries ADC1 come to $k$ – bit input of the multibit shift register (MSR) where $RG_j$ are stored in memory registers, the outputs are fed to the first $j - x$ digital multipliers inputs ($X$). Binary $k$ – bit codes that are generated at the output ADC2 with a certain time delay corresponding to the time delay of acoustic signals $M_j$, which are simultaneously fed to all second input $(j - x) k$ – bit binary multipliers ($X$), the outputs of which received $2k$ – bit binaries enter the relevant inputs $(2k + \log_2 n)$ – bit reversible accumulating adder $(\pm \sum \ldots)$, where codes are received digital values point correlation function $R_{n}(j)$.

Simplification of operations of division by sample size data set ($n$) in formula (4) is achieved throughout its multiplicity of 2 degrees, allowing to get the average value of digital codes point correlation function by discarding four junior level in the original binary code accumulating $\pm \sum \ldots$.

Obtained codes $R_{n}(j)$ with the bit $2k$, or with less precision $2k-r$, where $r = 2, 4, 6, \ldots$ are compared to the corresponding $(j - x)$ module compare $(\geq)$, the output of which is formed $m –$ bit position code Haar type $(00 \ldots 10 \ldots 0)$, the position "1" of which corresponds to the numerical values of the time delay of acoustic signals received $\Delta t$ by spatially located microphones $M_i$ and $M_j$.

For example, when the number of points of the correlation function $m = 4096$ binary code $\Delta t$ has 12 bits, that corresponds to $\Delta t$ 0.0025% measurement accuracy and uncertainty, and at $m = 1024$, corresponds to 10-bit binary code and the error does not exceed 0.001.

VI. CONCLUSION

The proposed method of optimizing the structure of multi-channel digital correlator with priority spatial placement of a microphone and application module correlation function to process acoustic signals, can significantly simplify the algorithm of calculations, reduce the hardware complexity correlator which enhance its performance, justifying feasibility and effectiveness of these solutions in the established system of monitoring sources of acoustic signals and implementation of special processors in microelectronic implementation on crystals FPGA Altera Cyclone 3 with the help of CAD Quartus 2.

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