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RESEARCH OF SOUND PROCESSING METHODS IN DIRECTIONAL MICROPHONE ARRAYS

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ABSTRACT:

The article is devoted to the study of sound processing methods in directional microphone arrays, one of the main stages of research work on solving the problem of creating directional microphone systems for remote reception of sound information. Based on the results of the study, it is planned to use modeling and an algorithm for processing sound in oriented microphone grids, as well as in the development of a layout of an oriented microphone grid.

KEYWORDS: directional microphone, directional microphone array, spatial correlation scale, the coherent field, the isotropic field, the diffusion field, delay and addition algorithm.

INTRODUCTION:

In today's era of information globalization, the role of information technology in our daily life is growing. In particular, the processing of audio signals has become a daily necessity. The field of voice processing underlies many areas, such as remote speech detection (for example, in devices with voice control), remote identification of the owner of the voice, remote audio retrieval.

Various acoustic devices are used in the sound processing. One of the main elements of acoustic devices is a simple omnidirectional microphone. The development of acoustic technologies has led to the emergence of special devices designed to receive sound data from distant sources - Directional microphones. Directional microphones can be used to remotely distinguish a useful sound signal coming in a specific direction from interference and noise in space.

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MATERIALS

The noise suppression efficiency of microphone grilles depends on the acoustic state (acoustic area of the noise).

There are three types of microphone grilles: coherent, isotropic and diffuse acoustic fields. These fields are characterized by the degree of spatial correlation of the noise. The measure of spatial correlation is a coherence function and it satisfies the condition $0 \leq | G_{ij} (f) | 2 \leq 1$. The coherence function is defined by the following formula:

$$G_{ij}(f) = S_{ij}(f) / (S_{ii}(f) \times S_{jj}(f))^2 \qquad (1)$$

where i- and j- are points in space, $S_{ij}(f)$ is the spectral density of the signals, and f is the frequency of the signals [5].

Shown below is how microphone arrays work in different acoustic fields.

The coherent field is created by a source located at a point in space in free space or in a room with a small level of reverberation. In this field, the sound signal coming from the source, without echo, reaches the microphones in the desired direction. In this case $|G_{ij}(f)|\ 2 \approx 1$ accepts an oblique microphone. Under such conditions, the signals coming from the noise sources that differ from the selected direction in the microphone grills are in different phases, so they attenuate each other when added. The efficiency of suppression of coherent interference of microphone grids in a coherent field is determined by the form of a directional diagram [5].

The isotropic (noncoherent) field is generated by scattered noncoherent noise sources and is manifested in the form of independent signals for different microphones. In this case $|G_{ij}(f)|\ 2 \approx 1$ accepts an oblique microphone. The isotropic field is also called spatial white noise. This acoustic field is rare in real conditions [5].

Because the noise in different microphones in the noncoherent isotropic field is not coherent, microphone grills lose their orientation relative to the noise. In this case, due to the addition of signals in unrelated microphones, the noise power at the output of the MICROPHONE GRILLS is reduced, and the selected signal strength is maintained at its original state. MICROPHONE GRILLS attenuate noise and increase the signal / noise ratio to a 10log10 (M) obliquely oriented microphone. Where M is the number of microphones in the microphone grilles [5].

Table 1 below shows the degree of noise suppression depending on the number of microphones.

Table 1. Noise suppression depending on the number of microphones.

M	10	20	40	80	160	320
NR, dB	10	13	16	19	22	25

For example, MICROPHONE BARS consisting of 320 elements increase the signal / noise ratio up to 25 dB compared to a single microphone and allow to receive audio data from a distance of 150-200 m. To double the range of sound in a non-coherent area, you need to increase the number of microphones by 4 times. A 4-fold increase in the number of microphones results in a 2-fold increase in the linear size of MICROPHONE GRILLS [5].

A diffusion field is an area formed by coherent sources that are prone to reverberation. In this case, the distance between the microphone grids and the source will be much larger than the reverberation radius. Under diffusion field conditions, the noise reaches the MICROPHONE GRILLS from all directions while partially maintaining the coherence property. The coherence function in this field is calculated as follows [5]:

$$G_{ij}(f)=\sin(2\pi f d_{ij} / c) \quad (2)$$

The diffusion field is a noncoherent field in the high frequency range, i.e. when the wavelength is smaller than the distance between the microphones. In the low frequency range, the signals in all microphones are coherent when the wavelength is close to the linear dimension of the MICROPHONE GRILLS. Under these conditions, noises in different directions may also be present in the useful signal structure due to the fact that noises in different directions are received through the side leaves and the main leaf at the output of the microphone grilles. To narrow the directional diagram by 2 times, you need to increase the line size of the microphone grills by 2 times. In the high frequency range, only the signals of neighboring microphones are coherent, in all other cases the condition of noncoherent diffusion field is fulfilled. In the high frequency range, the efficiency of the device increases as the linear size of the microphones and the number of microphones increase [5].

In the process of isolating the sound source signal in the selected direction through the microphone grilles, it is necessary to consider how far away it is. There are two concepts in this regard [6]:

– Distant Source;
– A Close Source.

An audio signal is viewed as a plane wave at a great distance, and as a spherical wave at a short distance.

The calculation of the delays of oriented microphones between the elements of the microphone grill system was carried out when the sound wave propagated in the right direction (Figure 1).

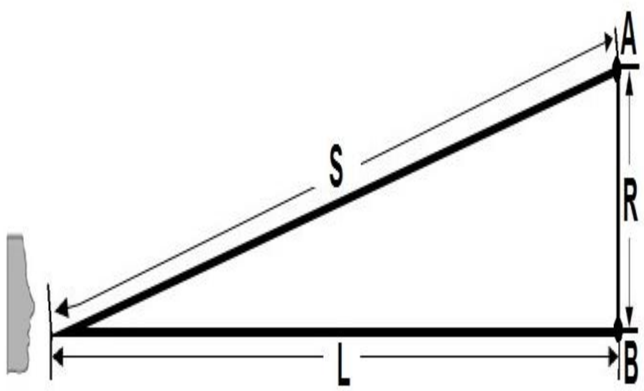

Figure 1. Scheme for determining the delay time in the microphone system.

If the audio signal travels in the correct direction along paths S and L, the difference between the arrival times at points A and B is expressed by the following formula:

$$\tau=(\sqrt{(L^2+R^2)}-L) / c \quad (3)$$

where L is the distance between the microphone grille system and the sound source standing on it, R is the distance between the center of the microphone grille system and its outermost element, and c is the speed of sound in the environment.

Figure 2 (3) shows a graph of the time dependence of the delay time when the distance between the center of the microphone grille and its outermost element is R = 0.5 m, based on formula (3).

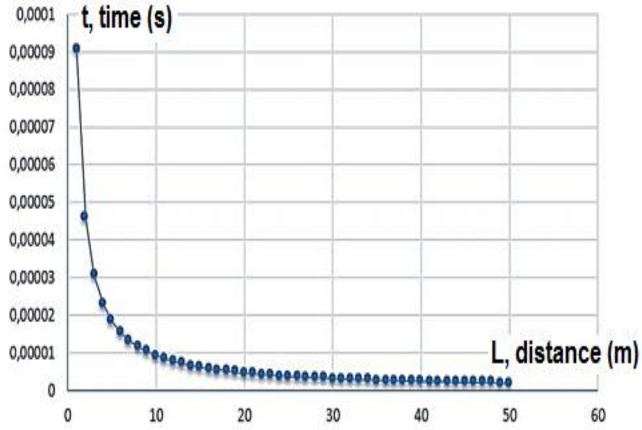

Figure 2. The delay time between microphones depends on the distance.

As can be seen from Figure 2, the delay time at a distance of 10 m is 10 μs. As the distance increases, this distorted microphone gets even smaller. Therefore, this oblique microphone can be neglected over long distances.

A remotely oriented microphone, which determines whether a sound source is far or near, must meet the following condition [6]:

$$L>(2\times D^2) / \lambda \quad (4)$$

where D is the linear size of the microphone grid, l is the sound wavelength.

For example, for linear microphone arrays with a linear size of 35 cm at a frequency of 5 kHz, this distance is L> 3.6 m. As the linear size of the microphone arrays increases, this distance increases rapidly.

METHODS:

There are many algorithms used in microphone grid systems to process audio data. In general, in microphone grid systems, audio data processing algorithms can be divided into two groups, adaptive and non-adaptive. If the acoustic state changes, you have to use an adaptive algorithm. Each of the algorithms used in microphone grille systems will be aimed at solving some kind of problem in accordance with the purpose. Examples of such issues are:

- Processing in real time (on-line) or post (off-line);
- For processing signals from sources over short or long distances;
- Signal processing in stationary or dynamic acoustic mode.

The desired algorithm is selected depending on what problem it is designed to solve.

The most widely used algorithms in the processing of audio data in microphones are:

- Delay and addition algorithm;
- Filtering and addition algorithms;
- Signal processing algorithms under the grid;
- High-level orientation algorithms;
- Algorithms for generating spatial zero;
- Algorithm for minimum noise variance (mvdr);
- Algorithm for the suppression of side flaps (GSC);
- De-Reverberation algorithms;
- Wiener's post-filtering algorithms;
- Algorithms for separation of acoustic sources.

In addition to the algorithms listed above, classical single-channel algorithms (equalizers, dynamic filters, correlated noise filters, etc.) are also used in audio signal processing. Acoustic field visualization, sound source tracking (state tracking) and other additional algorithms are also effective algorithms in the processing of sound data in microphone arrays.

The following are the characteristics of the most commonly used algorithms in audio signal processing in MICROPHONE GRILLS (Table 2).

Table 2. Orientation diagram properties of generation algorithms

Algorithms	Fixed / adaptive	Configuration of microphone grilles
Delay and addition	Fixed	Broadside
Delay and add under the grid	Fixed	Broadside
High level of focus	Fixed	Endfire
GSC	adaptive	Broadside
Viner post-filtering	adaptive	Broadside ёки Endfire

It is recommended to use fixed algorithms in environments with a constant acoustic state. This is due to the fact that such algorithms do not give good results in an environment where the acoustic state changes. In such cases, a solution to a particular problem can be found using special adaptive algorithms.

Table 3 below shows the advantages and disadvantages of the most commonly used audio signal processing algorithms in microphone arrays.

Table 3. Disadvantages and advantages of algorithms widely used in microphone arrays

Algorithms	Advantage	Disadvantage
Delay and add	Easy to apply in practice	Lack of flexibility in an acoustic environment
Filter and add	Designed for broadband signals	Lack of flexibility in an acoustic environment
Delay and add under the grid	Designed for broadband signals; Reduces the number of microphones in microphones	Lack of flexibility in an acoustic environment
High level of focus	Optimizes the transmission function of microphone grilles	Diffusion performance
GSC	Adapts to noise mode;	Possible distortion of useful signal content
Viner post-filtering	Minimizes noise signal strength at output	Possible distortion of useful signal content

Each of the algorithms listed in Table 3 has advantages and disadvantages. However, the implementation of the put-and-add algorithm is simple and highly efficient. Practical application of the rest of the algorithms is relatively difficult and expensive. Therefore, for the processing of sound data in the microphone system, the algorithm “delay and addition” was chosen (Russian -

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"«задержка суммирования»", English - "delay-sum").

The essence of the "delay and add" algorithm is that, since each element of the microphone grid system is located at different points in space, the sound signal from one source reaches these points at different intervals. Naturally, the instantaneously phase-shifted microphones of signals received at different times will also differ.

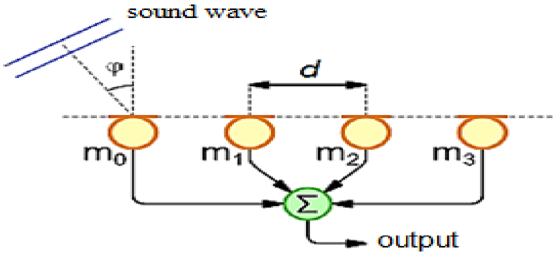

Figure 3. Image of a 4-element microphone array.

If there is a single sound source and the signal coming from it is added simultaneously without any change in the signals received by the different time-controlled microphones using the microphones, the phase difference will distort the signal structure and the resulting signal may not provide any information (Figure 4a). If the parameters of the signal from the source and the propagation velocity, as well as the distance between the microphones, are accurately oriented by the microphones, it will be possible to calculate the signal arrival time for each delay element. If the required slope-oriented microphones for each element are selected correctly, the signal received through them can be reduced to a microphone oriented to the same phase slope by delaying to the corresponding slope-oriented microphones. Adding these signals through an adder results in a signal whose microphone amplitude increases relative to the original signal amplitude depending on the number of microphones (Figure 4b).

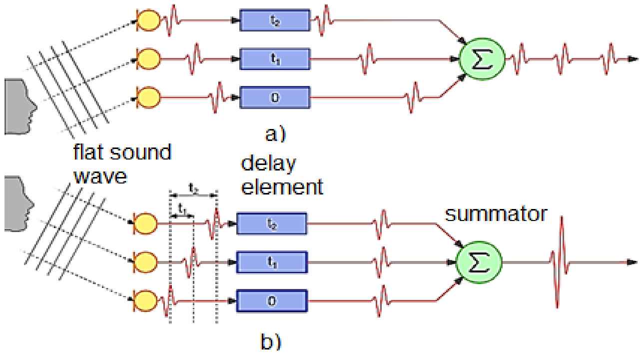

Figure 4. Adding signals: a) in different phases, b) in the same phase.

For any linear microphone array, the microphone delay time is determined by the following formula [5]:

$$\tau_n=(n-1)d\times\sin\varphi / c \qquad (5)$$

where d is the distance between microphones, an oblique microphone n lies in the range 1≤n≤N for microphone grids with N microphones, φ is the direction of the sound signal and the angle between the microphone system, c is the speed of the sound signal in the air (c = 340 m / s).

Below is a diagram of the "postpone and add" algorithm (Figure 5).

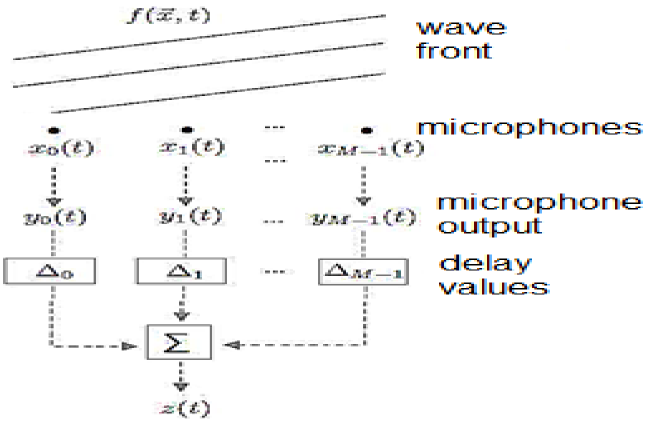

Figure 5. Delay and addition algorithm scheme.

RESULTS:

The resulting signal processed by the microphone array in the frequency range is determined by the following formula [2]:

$$y'(f) = \sum_{n=1}^{N} w_n(f)x_n(f) \qquad (6)$$

where $w_n(f)$ is the amplitude weight, calculated as follows:

$$w_n(f) = \frac{1}{N} e^{\frac{-2\pi f}{c}(n-1)\sin\varphi} \quad (7)$$

To simplify the appearance of the formula (6), the following view can be made:

$$y'(f) = w(f)^T x(f) \quad (8)$$

where w(f) and x(f) are calculated as follows:

$$w(f) = [w_1(f) \ \cdots \ w_n(f) \ \cdots \ w_N(f)]^T \quad (9)$$

$$x(f) = [x_1(f) \ \cdots \ x_n(f) \ \cdots \ x_N(f)]^T \quad (10)$$

If the angle between the normal axis of the microphone system and the direction of the sound signal is 0°, then the form of formula (7) will be as follows:

$$w_n(f) = \frac{1}{N} \quad (11)$$

The resulting signal in the time field can be expressed by the following formula:

$$y(t) = \frac{1}{N}\sum_{n=1}^{N} x_n(t-\tau_n) \quad (12)$$

DISCUSSION:

The calculated y(t) signal ideally represents the wanted signal. In a real case, the resulting signal may contain additional noise generators. A good result can be achieved by clearing the content of the resulting signal from such noise generators using filters of a certain type.

CONCLUSION:

This article discussed the advantages and disadvantages of methods and algorithms for processing audio signals in MICROPHONE Arrays, and analyzed the "delay and add" algorithm among them. The main advantage of this algorithm is its ease of implementation. This is because delay and add operations do not require high-level technical tools and large amounts of money. The main disadvantage of the algorithm is the lack of flexibility in the acoustic environment. An effective result can be achieved by processing the received signal in order to overcome these disadvantages. In addition, a number of algorithms for processing sound in microphone array systems are based on a delay and add algorithm. Based on this, the goal was to use a delay and add algorithm when processing audio data in a microphone system.

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