Two-channel acoustic noise control system

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Abstract. The results of the development and experimental verification of a two-channel acoustic noise control data processing system that provides real-time control of non-deterministic broadband acoustic noise are presented. The system is implemented using the LabView graphical programming language G. The length of the measured realization is determined by the sample size ratio to the sampling frequency, and the number of measurements of each sound signal is determined by the ratio of the analysis time to the length of the measured realization. The measured realization of the audio signal in the form of 40,000 16-bit binary numbers is processed using the fast Fourier transform. As a result of processing, a discrete audio signal spectrum is obtained in the range from 0 to 20 KHz in steps of 1 Hz. To check the quality of the system’s functioning, a pair of microphone capsules of the MDN-318, wm61, etc. types, widely used in computer audio equipment, preliminarily selected according to the amplitude-frequency characteristics. The results of system verification using a laboratory acoustic chamber are presented. The advantage of the system is the receipt of a difference spectrum of controlled signals. The conclusion is drawn about the possibility of using this system for a qualitative assessment of changes in the noise level along the propagation path, as well as additions to the results of measurements performed by standard methods.

1. Introduction
The problem of monitoring the territories of cities and settlements in recent years has become increasingly important. The habitat condition of cities and, especially, large ones is determined by a combination of climatic, technogenic and planning factors. It is known [1] that in large agglomeration cities, the state of the atmosphere surface layer can vary significantly from region to region, which is determined by the features of the area layout, the activity of traffic flows, architectural features of residential and technical buildings, the saturation of the area with industrial enterprises, the nature of the underlying surface, and etc.

Among various factors affecting human life, in such an environment, acoustic noise is not the last place, which is the third most important among environmental hazards. It is generally accepted that in recent years the noise level in the technosphere has been increasing by at least 0.5 dB per year. Thus, the task of studying and analyzing the acoustic noise of technogenic origin, the nature of its distribution becomes relevant [2].

Tools currently used for measuring sound pressure levels of acoustic signals — sound level meters — belong to different classes and have different capabilities [3]. In the classic version, the sound level meter is a measuring system in which the microphone performs the function of the sensor, there is an amplifier, a set of band-pass filters, and the result is displayed either using a dial gauge or a digital display. Evaluation is carried out either on average over the entire range, or at the geometric mean
frequencies of the octave ranges. More complex systems provide simultaneous measurements in the octave (doloctave) ranges, which makes it possible to save the measurement results, to carry out built-in and subsequent external computer processing of the received information. The sizes of the ranges and the weighted average frequencies corresponding to these ranges are determined by row R10 according to [3].

More complex systems which are measuring complexes, including powerful sound amplifying stations, a set of radio meteorological instruments, server devices, and information processing programs are known [4]. The systems are stationary and cannot be used to monitor technogenic noise characteristic of the urban environment. Namely, in an urban environment, tasks that require evaluating and comparing non-deterministic acoustic signals in areas remote from each other are relevant, which requires the presence of at least a two-channel measurement system. Such systems can provide an estimate of the noise level attenuation during propagation from the source to the operating point in the case of non-repeating impulse noise, when analyzing the propagation of acoustic signals deeper into the living area, assessing the effectiveness of noise suppression means, etc.

It should also be noted that when conducting measurements using octave or doloctave filters, some of the useful information about the nature of the broadband signal may be lost. The sound pressure level in this case is fixed at specific geometric mean frequencies, and the presence of bandpass filters “cuts off” all other signals. It was shown in [5] that with such measurements, some of the significant information about the nature of broadband signals may be lost, which will not allow a correct assessment of the noise propagation nature.

Figure 1 shows an instant slice of the spectrogram measurements taken on the streets of the city of Murom. The vertical red lines show some values of the geometric mean frequencies of the third-octave bands (2000, 2500, 3150 and 4000 Hz), in addition, the frequency values for the case of the octave bands (2000 and 4000 Hz) are additionally highlighted. The green mark on the vertical line related to the frequency of 2500 Hz shows the average value of sound pressure for a given frequency.

Figure 1. Instant spectrogram slice

At the same time, two amplitude bursts at frequencies of 2340 and 2450 Hz can be seen in the spectrogram. Their level exceeds the indicated average value by 21.2 and 16.2 dB, respectively. Obviously, in this case, the amplitude burst at a frequency of 2450 Hz is taken into account to a large
extent in determining the average value. The degree of accounting for the second burst (2340 Hz) is already determined by the characteristic of the bandpass filter used in the sound level meter. It is also obvious that in the case of measurements in the octave range mode, both of these bursts will not be taken into account when determining the average value at frequencies of 2000 and 4000 Hz.

The loss of information about pulsed random acoustic noise can be even more significant when analyzing the nature of the propagation of such noise deep into a residential area, choosing noise protection equipment when the value may not be of the average level, but pulsed non-deterministic bursts of the noise signal amplitude.

Thus, the urgent task is to create simple multichannel tools for analyzing the characteristics of various noise signals based on well-known computer programs using commonly available peripheral devices and systems that provide real-time and full-spectrum analysis.

2. System description

The program developed for the simultaneous analysis of the spectra of 2 audio signals uses the graphical object-oriented programming language GOOP LabView [6]. The choice of LabView environment is due to the convenience of writing programs and its wide capabilities for creating object-oriented multi-threaded applications for working with devices in real time [6, 7, 8]. Inputs include: analysis time Time (sec), sample size (# of samples), and sampling frequency (Hz). The length of the measured realization is determined by the ratio of the sample size to the sampling frequency and for the same values selected by default (40,000) is 1 second, the number of measurements of each sound signal is determined by the ratio of the analysis time to the length of the measured realization. The constants specify the number of channels for connecting each of the microphones (in this case there are two), the number of bits — 16 — of the digitized sound signal and the number of channels for connecting microphones to USB PC ports. Connection channel numbers are determined using the Acquire Sound utility of LabView before starting the program.

The program of a two-channel analyzer of spectra of sound signals in the GOOP language of the LabView environment is presented in figure 2.

Measurements of the signal from the output of each microphone are made by a separate object in a separate stream with the required number of repetitions. Each object is created by its own Sound Input Configure constructor, reads the current realization from 40,000 samples from the microphone output using the Sound Input Read method and is deleted from the memory after completing the measurement sequence using the Sound Input Clear destructor. The ability to read realizations sequentially is set for each constructor using the Continuous Samples option. Reading individual realizations from the outputs of both microphones takes place in parallel and also takes 1 second as for a single microphone, since the processor is freed up in the pauses between the individual measurements for a time inverse to the sampling frequency.

The measured audio signal realization in the form of 40,000 16-bit binary numbers is processed using the fast Fourier transform (FFT). The first half is extracted from the obtained complex two-sided spectrum, the imaginary part of each spectral line is discarded, and the remaining real part of each spectral line, except the zero one, is multiplied by 2. The result is a discrete spectrum of the audio signal in the range from 0 to 20 KHz with a step of 1 Hz.

Upon completion of each of the 2 streams (cycles) of measurements, the corresponding spectrum of the sound signal averaged over a given analysis time should traditionally be calculated according to the well-known principle:

\[ f_{cp}(n) = \frac{\sum_{i=1}^{n} x_i}{n}, \]

where \( x_i \) — received current spectrum value; \( n \) — number of realizations.
Figure 2. Diagram of a two-channel analysis of the spectra of sound signals
This approach is traditional and classic. However, it does not allow tracking changes in the spectrum of the signal in real time, which would make it possible to visually observe the spectrum of rapidly changing broadband signals with a variable initial spectrum.

In this system, another option was chosen to obtain the resulting spectrum [9]. The essence of the adopted approach is to calculate the current average at each iteration of the averaging cycle using a recurrence relation of the form

\[ f_{av} = \frac{(n-1)f_{av} + x_n}{n}, \]  

where the initial value \( f_{av} \) at \( n = 0 \) can be any, since it is multiplied by zero.

To obtain the desired ratio, it is possible to perform a simple transformation of relation (1), where the following measurement \( x_{n+1} \) is added:

\[ f_{av}(n) = \frac{\sum_{i=1}^{n-1} x_i + x_n}{n}. \]  

Using formula (1) again, we obtain the expression for the sum included in the numerator of the right-hand side of (3):

\[ \sum_{i=1}^{n} x_i = nf_{av}(n). \]  

which we substitute in (3). Then the resulting mean value at each step is obtained using the relation

\[ f_{av}(n) = \frac{(n-1)f_{av}(n-1) + x_n}{n}. \]  

This option makes it possible to monitor changes in the average spectrum in real time. In this case, there is no need to set the observation time in advance, since it becomes possible to stop the measurements when the required accuracy is achieved, if the essence of the controlled process of signal propagation is understood, etc.

The obtained values are displayed on the screen in the form of continuous spectra with Spectrum Out 1 or Spectrum Out 2 objects and are stored in one-dimensional arrays Spectrum Out Num 1 or Spectrum Out Num 2.

The average difference spectrum is calculated in a separate, third flow with a given number of repetitions. Information about the current spectra calculated in the first 2 flows from the realization lasting 1 sec enters this stream through the local variables Sp1 and Sp2. At the same time, through the local variables In1 and In2, discrete realizations of the initial time signals from the microphone outputs in real time are received and displayed on the screen by the objects Time Domain Sequence In 1 and Time Domain Sequence In 2. The received current spectra (in dB) are displayed on the screen by Spectrum Continue 1 and Spectrum Continue 2 objects, and are also subtracted at each discrete frequency. The resulting difference in the logarithms of the spectra is summed with the previous current difference, originally set to "0". Upon completion of the third stream, the average difference of the spectra logarithms is calculated, which is also displayed on the screen by the Spectrum Subtract object as a separate graph and stored in a one-dimensional Spectrum Subtract Num array.

The number of repetitions in the third stream is two more than in the first two streams, and the first iteration of the third stream is used only for 1-second delay, which is spent on reading the current realizations from the microphone outputs. The delay is carried out by the Wait (ms) utility, an important feature of which is that the central processor is freed for the remaining time until the end of the delay after performing the necessary calculations. The first (zero) iteration of the third stream is skipped by
number. The second iteration of the third cycle is used to compensate for the time taken to calculate the spectra in the first two streams. The condition for skipping the second iteration is the absence of a difference from “0” in the values of the local variables Sp1 and Sp2 corresponding to the current spectra in each of the first two streams. Zero values are specially assigned to these variables when the program starts.

After completing the flows, from three one-dimensional arrays formed with their help, containing two average spectra of acoustic signals, as well as the average difference of the logarithmic spectra of the same signals, a two-dimensional array of the 31st row is formed using two nested loops. This size in this case is determined by the choice of a one-third octave mode of fixing the measurement results, which can be reduced, if necessary, to 11 lines when the octave mode is selected. Each line contains four elements: the value of the selected frequency (Hz), the average values of the spectra of the 1st and 2nd acoustic signals (dB), as well as the average difference of the logarithmic spectra of the same signals (dB). The generated two-dimensional array is saved in a file in the form of a spreadsheet.

To calibrate the measurement system at a fixed frequency using a sound level meter or an external calibrator, an additional variable must be added to the system input by which the current spectra Sp1 and Sp2 must be multiplied. As a result, the data obtained will correspond to the scale of the sound level meter or calibrator.

3. Experimental system check
To test the system, the simplest microphones consisting of general-purpose microphone capsules of the types MDN-318, wm61, etc., widely used in computer audio equipment, and specially designed capsule holders were used. The acoustic signal is noise-like, such as “white noise”, with a fairly uniform distribution within the range of 20-20000 Hz, also generated using the LabView. Assessment time is up to 60 sec.

Previously, measurements were carried out using several tens of capsules in order to identify similar patterns of amplitude-frequency characteristics (AFC).

The test was carried out using a laboratory acoustic chamber designed for use in the educational process and research work of students, the description of which is given in [10]. The chamber has external dimensions of about 2 m, 1 m and 0.8 m with a vertical orientation. In order to equalize the characteristics of the chamber and reduce the echo level, the inner surface is covered with sound-absorbing material - acoustic foam using the so-called bass traps for absorbing low-frequency acoustic signals.

A broadband speaker system is integrated into the bottom of the chamber. The direction of acoustic signal emission is from the bottom up.

During the measurements, the microphones were placed in random pairs in the upper part of the acoustic chamber. Two holders were used to fix the capsules. Then, pairs having similar signal spectra were selected.

The results of the comparison of the most optimal pair of capsules in this series are presented in figure 3. The first and second spectra are the spectra of the acoustic signal recorded by the first (in this pair) and second capsules, the third one is the difference spectrum (with subtraction of the second from the first).

As follows from a comparison of the signal spectra, there is no need to talk about the full correspondence of the AFC of the microphones, since the nature of the change in the average difference line has a difference reaching up to 15 dB, which does not allow us to talk about the use of such capsules to build a precision measuring system.

At the same time, it should be borne in mind that the current methods of measuring acoustic noise provide for the analysis of signals at the weighted average frequencies of the octave or doleoctave ranges [3]. Since, as already noted, the developed system ensures the storage of data in the form of a base built on recording the sound pressure level exactly at the average frequencies with the possibility of selecting ranges, we will use the data for fixing these measurements and present in the form of graphs the difference in levels in the third-octave and octave ranges (figure 4).
An analysis of these AFC shows that the deviation amplitude of the signal difference in the one-third octave range is 12.5 dB, but in the octave range, the amplitude deviation in general is 3.5 dB. The latter result is generally consistent with the “Type 2” classification in the US ANSI (American National Institute of Standards) classification of noise measuring instruments [ANSI / ASA / IEC 61672-2-2019. International standard. Electroacoustics - Sound level meters]. For instruments of this type, the required
measurement accuracy is of the order of ± 2 dB. Such sound level meters are used for general purpose measurements: street noise, night noise, hum of cars, kitchen appliances, etc.

4. Conclusion
An analysis of the results of the studies shows that the use of the developed measuring system in combination with the simplest microphone capsules of wide application for precision measurements of a monitoring nature is not possible due to too large differences in their individual AFC.

However, their use in systems for the qualitative assessment of changes in sound pressure level, where measurements can be carried out with some error, is quite possible. For example, the use of such a system with the considered pair of capsules is quite possible in cases of evaluative measurements when analyzing changes in acoustic signals along the propagation path (at small distances), evaluating the effectiveness of noise protection means for general use, in other cases. In addition, it is possible to use this system as an addition to measurements made according to standard methods.

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