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MODELING OF VIBROACOUSTIC PHENOMENA USING THE METHOD OF PARAMETERIZING THE AUDIO SIGNAL

MODELOWANIE ZJAWISK WIBROAKUSTYCZNYCH Z ZASTOSOWANIEM METODY PARAMETRYZACJI SYGNAŁU FONICZNEGO*

The article proposes an original way of modeling vibroacoustic phenomena of exploited machines/devices using the method of audio parameterization. This method extends the current approach to this type of research and consists in taking into account the psychoacoustic effects associated with the emission of vibroacoustic energy. The proposed solution is based on the determination of mel-cepstral coefficients of the examined signal and its classification, due to the impact of noise. It was presented verification of the method on the example of studies on the impact of road noise sources.

***Keywords**: vibroacoustic phenomena, signal parameterization, noise perception, modeling.*

W artykule zaproponowano oryginalny sposób modelowania zjawisk wibroakustycznych eksploatowanych maszyn/urządzeń z zastosowaniem metody parametryzacji sygnału fonicznego. Sposób ten rozszerza dotychczasowe podejście do tego rodzaju badań i polega na uwzględnianiu efektów psychoakustycznych towarzyszących emisji energii wibroakustycznej. Proponowane rozwiązanie opiera się na wyznaczeniu współczynników mel-cepstralnych badanego sygnału i jego klasyfikacji, ze względu na oddziaływanie hałasu. Przedstawiono weryfikację zastosowania metody na przykładzie badań oddziaływania źródeł hałasu drogowego.

***Słowa kluczowe**: zjawiska wibroakustyczne, parametryzacja sygnału, percepcja hałasu, modelowanie.*

1. Introduction

The source of phenomena occurring during the exploitation of machines and devices are complex vibroacoustic processes. These phenomena consist of varied impacts of vibration, noise, air and material sounds, or medium pulsation in the areas of machine parts. These impacts are emitted to the environment in the form of vibroacoustic energy. The effects of vibroacoustic phenomena are the response of machines/devices in relation to their parts and environment. Carrier of information about vibroacoustic phenomena there is a signal, which may be subject to various transformations. Analysis of the vibroacoustic signal as a way of processing data into useful information is used not only in maintenance tasks [3, 10], diagnosis [13], reliability [9], assessment of the exploitation condition of technical objects [14], but also in the tasks of the impact of machinery/devices on the environment. This impact can be analyzed in open spaces, rooms and in relation to the human body. Vibroacoustic phenomena are all vibration and acoustic waveforms, which are related in a causal way. It was found, that in the study of these phenomena, the following issues should be included [5]:

- time and spatial distribution of characteristics, describing the energy coming from the source,
- system response in the form of a vibroacoustic transition function,
- interdependence between sources.

Vibroacoustic phenomena are described with using basic physical quantities such as sound pressure, speed, acceleration, displacement, force [5]. In turn, the vibroacoustic signal can be represented by a unary function or a vector [1].

The applied methods of analyzing vibroacoustic signals use various Fourier transforms, so you can get amplitude, phase or energy spectra. Fourier transform allows obtaining valuable information by changing the signal from the time domain to the frequency domain, especially when there is a high dynamics of change of signal parameters time. Vibroacoustic research require not only measurements of vibration and noise characteristics, but also obtaining information about the studied phenomenon. The assumed research goal requires specifying the measurement methodology in relation to the technical object or its environment [5].

2. Modeling of vibroacoustic phenomena and sound perception review of current solutions

Modeling of vibroacoustic signal are subject to vibration and acoustic phenomena, generated during the exploitation of machines/devices. For analyzing vibroacoustic processes in the environment, it is used acoustic modeling of machines and devices [5]. For this purpose, methods of modeling the sound field of machines/devices are used to identify sound sources. Characteristics of sound sources can be assessed by [5]:

- a sound field generated by the source,
- a source as an emitter of vibroacoustic energy.

Methods of modeling of vibroacoustic phenomenon are widely used in tasks including identification and reduction of vibration and noise sources, analysis of vibroacoustic processes [11], the assessment of the technical condition of machines/devices and their reliability, damage analysis of machine/device elements or the assessment of the impact of vibroacoustic energy on the human body. For example, diagnostic modeling methods use a vibroacoustic signal to examine for wear or damage to technical object parts [5]. Among the known methods of modeling these phenomena, the following methods can be distinguished: pressure, intensity, reciprocity, finite elements, inversion, Fourier transforms, short-time Fourier transformation, wavelet transformation [1, 5].

(*) Tekst artykułu w polskiej wersji językowej dostępny w elektronicznym wydaniu kwartalnika na stronie ***www.ein.org.pl***

In the analysis of vibroacoustic phenomena, most often there are processed data and information directly related to the sound source, human or environment, including interactions between them. One of the goals of searching for the relationship between the characteristics of a machine/device in the environment (noise source) and the parameters of the acoustic field is the formulation of sound emission models for the purposes of assessing its impact [23]. In this regard, based on the vibroacoustic signal, appropriate physical characteristics of measured energy quantities are determined. On this basis it is defined the degree of harmful vibration or noise, negative impact of vibroacoustic phenomena on the human body, or on machine/device components or the environment. Input data for modeling vibroacoustic phenomena are acquired acoustic information from specified points of the machinery/device environment. For the purpose of determining the degree of exposure at a given point, it should be made measurements of characteristics of the noise pollution [6]. As a result of acoustic measurements, appropriate energy indicators are obtained at a given point [4, 21]. Studies on the auditory effects of human noise confirm, that noise risk assessment using only energy indicators is limited and insufficient [24, 20]. In the used approach, the subjective significance of noise impressions is ignored. It is not included interaction of sound pitch and the loudness of dynamically changing sounds. The A-, B, C- and D- filters, used in this respect, approximate the inverted shape of curves with equal loudness at different levels of sound pressure. Weighing with the A filter has become the most commonly used frequency factor, although it is not optimal for all sound pressure levels. Studies on sound perception show that the harmonics of sound with frequencies in the range 1÷5 [kHz] are more audible than the others [16]. This is important in interpreting the occurrence of elementary and complex phenomena in low and high frequency ranges [15]. In the research on the assessment of the subjective perception of human loudness for continuous sounds, it is used the method consisting in determining the total (equivalent) sound level corrected by a hearing correction filter (A). It should be noted that the frequency response of the filter (A) is an approximation of receiving audio impression of low sound levels. The analysis of the characteristics of the hearing threshold shows that hearing sensitivity is highest in the medium frequency range 1000÷4000[Hz] and it significantly decreases in the low and high frequency band. It should be noted that the characteristics of the thresholds of hearing, discomfort and pain significantly differ from each other as a function of frequency [12]. Analyzing the mechanism of acoustic impressions in relation to the characteristics of sound, it should be said, that there is no simple and clear relationship between the physical and subjective characteristics of sound. Research on subjective sound characteristics relate to the description and interpretation of auditory impressions. The results confirm that noise perception is significantly affected by among others psychoacoustic aspects of sound, signal time structure, shaping of subjective sound features in the domain of: time, frequency, time-frequency [8, 17, 18, 19]. The undertaken research on the assessment of the quality of audio signals is based on the use of two categories of methods [26]:

- subjective methods - consisting in assessing the human auditory impressions,
- objective methods - consisting in using approximate mathematical models to include perception mechanism.

Subjective methods of assessing sound quality use auditions of the assessed signal for the purpose of determining, perceived by the listener, the degree of similarity of both signals, the degree of their difference, or the level of discomfort caused by the presence of interference or distortion. These methods allow the evaluation of a direct response from the recipient, and individual subjective factors significantly affect a single assessment. Among the objective methods it can be distinguished:

- signal methods - most often the assessed signal is compared with the original signal, without distortion (reference signal). It is also used methods which do not take into account the reference signal.
- parametric methods - the quality of sound is assessed on the basis of knowledge about the applied processing technique and knowledge of its parameters, which are the input arguments of the assessment algorithm.

The review and interpretation of models and methods for analyzing vibroacoustic phenomena and sound quality assessment shows that the features contained in the audio signal can be a valuable source of information in the study of noise perception. It is important to take into account the psychoacoustic effects of the impact of vibroacoustic energy on the human body, both during its implementation of operational activities, maintenance, as well as being near sources of vibration and noise emissions.

The problem undertaken by the author concerns the inclusion, in modeling tasks of vibroacoustic phenomena, of psychoacoustic effects resulting from the exploitation of machines/devices.

3. A way of modeling of vibroacoustic phenomena with the use of the signal parameterization method

Auditory sound impressions resulting from acoustic processes are strongly dependent on frequency, due to the physical conditions of acoustic wave propagation and perception. It should be regarded as significant, non-linearity and the range of perception of sound phenomena, in relation to the amplitude of sound pressure and frequency [2]. The author proposed a new approach to modeling vibroacoustic phenomena occurring in the environment of operated machines/devices. This approach relies on extracting selected features from the acoustic signal and from the audio signal. The acoustic signal is represented by the physical features of the sound. The audio signal contains information representing the subjective features of the sound. In the proposed method, the starting point for modeling vibroacoustic phenomena are the features of the acoustic signal and the features of the audio signal. Fig. 1 presents a method of modeling vibroacoustic phenomena occurring during the exploitation of machines/devices, which takes into account the psychoacoustic effects of noise.

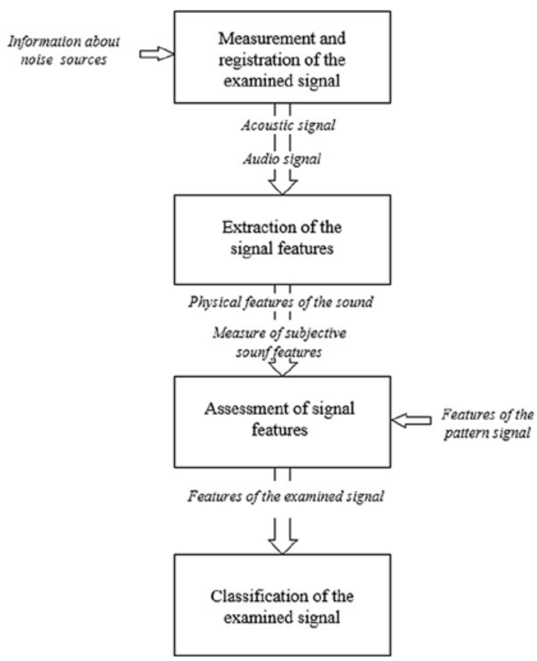


Fig. 1. The method of processing the features of the acoustic signal and the features of the audio signal

According to Fig. 1, the author proposed the following actions for the classification of the examined signal:

- measurement and recording of the examined signal: it is an information source for the acquisition and processing the features of the acoustic signal and the features of the audio signal,
- extraction of signal features: involves the use of a method/model for assessing physical and subjective features of an acoustic and audio signal,
- assessment of the features of the examined signal: the basis for the assessment of the features of the audio signal is the degree of compatibility of its features with the features of the reference signal,
- classification of the examined signal: the assignment of the audio signal to a given class of the reference signal based on the assessment of features.

The proposed method assumes, that assessment and selection of reference signals will be based on playing back recorded audio signals to recipients as part of a psychoacoustic experiment. Assignment of reference signals to the appropriate noise assessment class is realized on the basis of the results obtained from the signals presented during the experiment.

3.1. Description of the method of parameterizing the audio signal

In the considered research problem, the proposed method of parameterizing the audio signal is based on a perceptual model. This model uses the properties of the hearing mechanism of the human ear, characterized by a non-linear perception of the height of the frequency of received sound signals on a mel-scale. Method of parameterizing the audio signal, applied by the author, in modeling vibroacoustic phenomena is original and not found in the literature of the subject. The MFCC (Mel-Frequency Cepstral Coefficients) method is one of the most commonly used methods of speech signal parameterization. It allows the determination of a set of cepstral coefficients, i.e. the features of the signal from the melody spectrum [7]. Mel-cepstral coefficients are patterned on the processing of the acoustic signal in the cochlea of the human hearing organ. Their task is to reflect the natural response of the auditory system to sound stimulation. The proposed method consists in modeling the parameters extracted from the audio signal, which strongly depend on the subjective listening impressions of sound. The prerequisites for the implementation of the study of vibroacoustic phenomena by mel-cepstral coefficients was the recognition of the possibility of using the method of parameterizing the audio signal. This method allows taking into account, among others:

- randomness of signals commonly found in exploited machines/devices,
- variability of the frequency structure in the signal waveform,
- estimation of the signal spectra on a subjective perceptual scale,
- nonlinearity of the perception of the frequency of sound by human.

Determination of mel-cepstral coefficients gives the possibilities of effective classification and assessment of the examined audio signals. Lack of universality of solutions in the recognition of acoustic signal patterns for the purposes of their assessment is not due to the imperfection of methods but to the complexity of the source signals. For this reason, the transformation of the analyzed signals is used to obtain the appropriate space of sound features. For the purpose of determining the vector of MFCC coefficients, it was applied an algorithm to extract the features of the audio signal (fig. 2).

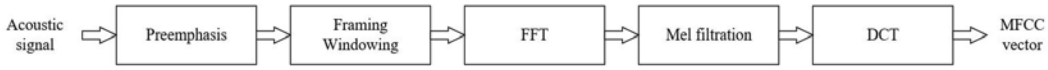


Fig. 2. Parameterization procedure of the audio signal using MFCC coefficients [25]

According to the procedure presented in Fig. 2, it was adopted staged realization of signal features processing [25]:

Stage 1: Preemphasis process consisting of forming filtration, which results in weakening of low-frequency components and amplification of high-frequency components.

Stage 2: Signal framing, i.e. dividing the signal into short fragments called frames. It is possible applying overlapping successive time frames. Then, at this stage, windowing is carried out using Hamming window:

$$Ham(N) = 0{,}54 - 0{,}46\cos(2\pi\frac{n-1}{N-1}) \tag{1}$$

where:

N - frame length,
n=1,2,...,N.

Stage 3: Execution of the Fast Fourier Transform (FFT) algorithm on the windowed signal in individual frames and determination of the module of estimate of the power spectral density of the signal.

Stage 4: Performing mel-filtration using a set of bandpass triangular filters with frequencies determined in accordance with:

$$f_{mel} = 2595\log_{10}(1+\frac{f_{\mathrm{Hz}}}{700}) \tag{2}$$

The calculations used the logarithm of energy, which reduces the sensitivity of filters to very loud and very quiet sounds and allows modeling of non-linear amplitude sensitivity of the human ear.

Stage 5: The final stage of the procedure is the use of the discrete cosine transform (DCT). The resulting vector MFCC coefficients is calculated according to the relationship:

$$MFCC_n = \sqrt{\frac{2}{N}}\sum_{i=1}^{N}\log(S_i)\cdot\cos\left[\frac{\pi n}{N}(i-0{,}5)\right] \tag{3}$$

$$S_i = \sum_{k=1}^{N}\left|X_r(k)\right|^2 H_i(k) \tag{4}$$

where:

i - filter number numer filtra,
X_r - frame spectrum,
H_i - filter set,
S_i - band energy,
n - coefficient number,
N - number of filters used.

In most recognition systems n is 1, and the coefficient $MFCC_0$ is omitted. The generated vector coefficients $\mathrm{MFCC_n}$ takes the form:

$$MFCC_n = \langle MFCC_1, MFCC_2, MFCC_3, ...MFCC_{13}\rangle \tag{5}$$

In the presented research method, it was assumed, that reference signals identified as part of the psychoacoustic experiment will be parameterized to determine mel-cepstral coefficients. According to the procedure (Fig. 2), mel-cepstral coefficients are determined for each examined audio signal, which will then be evaluated using pattern signals. Based on the determined MFCC coefficients, the classification of the examined signal consists in the assessment of the degree of compliance of its features with the pattern signals. In order to classify the examined audio signals (based on a set of standard signals), as a measure of assessment, the author proposed the smallest distance between two series of feature vectors, i.e. MFCC coefficients using the DTW method (Dynamic Time Warping). The signals emitted by vibroacoustic sources are characterized by dynamic variability in time, which causes that the features of these signals are also subject to variability. Dynamic Time Warping (DTW) is one of the methods used in speech recognition. In particular, this method was primarily used in recognizing isolated words and searching for passwords. This method is used to recognize and classify the matrices of features of the MFCC coefficients as a non-linear time transformation. It involves the transformation of the timeline, to better match two time sequences. Determining the distance using the Dynamic Time Warping (DTW) method consists in:

- calculating the so-called matrix of local distances d(m, n), which is created by calculating Euclidean distances between each vector of the examined signal and the pattern signal,
- determining the sum of local (Euclidean) distances, which is the distance accumulated along the optimal path in the local distance matrix, from the lower left corner of the matrix to its upper right corner.

Particular DTW distances between the pattern signals and examined signal can be determined according to the following formula:

$$d_{mn}(X,Y) = \sqrt{\sum_{k=1}^{K}(x_{k,m} - y_{k,n}) \cdot (x_{k,m} - y_{k,n})} \qquad (6)$$

where:

K- signal dimension,

m, n - vector sequences of X and Y features.

The accumulated distance calculated along the optimal path is the smallest of the possible accumulated distances of individual vectors of the MFCC coefficients of the examined signal and the pattern signal. It was assumed that the criterion for the classification of the examined signal in the scope of assessing noise annoyance will be the smallest DTW distance between mel-cepstral coefficients of the examined signal and a given pattern signal of the same time length.

3.2. Example of using the method and classification of the examined audio signal

The audio signal selected for examine was a sound sample called DW13a_9_56, characterized by an equivalent sound level L_{Aeq}=56[dB(A)]. The signal was recorded in a point located at the height of the road lane and it included the measurement of the immission of road noise sources of moving vehicles on a cobblestone pavement. Due to the research goal, the measurement was not made in accordance with the applicable methodology for carrying out environmental measurements [22]. The following set of devices and measuring aids were used for recordings and measurements:

- Brüel & Kjær 2238 Mediator sound level meter,
- Brüel & Kjær 4188 measuring microphone for measurements of a free field,
- ZOOM Handy Recorder H4n sound recorder,
- microphone tripod.

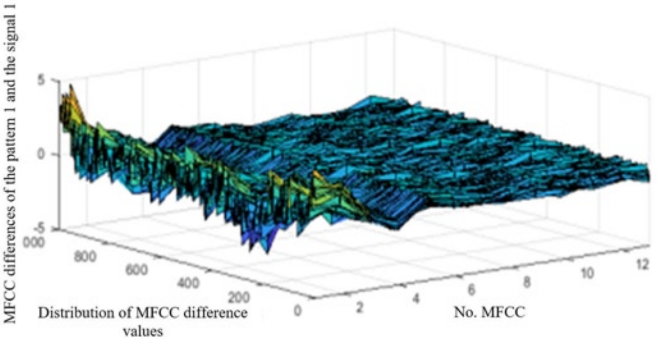
The time of recording and measurements was set at 5 [min]. During recording and measuring, the microphone was placed on a tripod in the middle of the pavement, at a height of 1.7 [m], which corresponded to the approximated height of the head of a potential passerby. It was assumed that in the subjective assessment of sound samples (in a psychoacoustic experiment), there participated people with normal hearing. Ten 10-second files were selected for each 5-minute file. In the carried out experiment, there attended 80 people aged between 22-50 years, who met the above assumption. In ongoing psychoacoustic research, to assess the annoyance of road noise from the 30 audio signals presented, a point scale in the range of 1 to 5 was proposed. Appropriate labels have been assigned to the assessed signals, i.e.:

- not at all - grade 1,
- little - grade 2,
- medium - grade 3,
- very - grade 4,
- intolerable - grade 5.

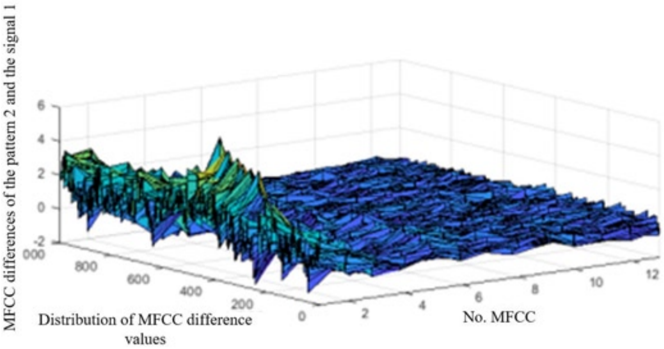
Taking into account the extent of the obtained sound level values corresponding to different measurement locations (characterized by variable buildings, variable road infrastructure system) and different road surfaces (i.e. pavement, asphalt), it was decided that the selected signal samples will be prepared in such a way that their equivalent

Table 1. List of σ and $\bar{x}$ values of MFCCn mel-cepstral coefficients for pattern signals

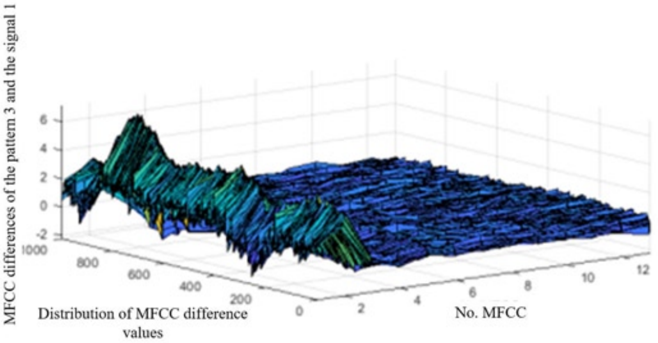
	Pattern signal 1		Pattern signal 2		Pattern signal 3		Pattern signal 4		Pattern signal 5	
Coefficient No.	σ	$\bar{x}$	σ	$\bar{x}$	σ	$\bar{x}$	σ	$\bar{x}$	σ	$\bar{x}$
MFCC1	1,25	-2,07	1,40	-2,08	1,16	-1,99	1,38	0,61	1,40	1,43
MFCC2	1,27	-18,01	1,81	-16,32	1,62	-14,21	2,01	-11,14	2,00	-10,60
MFCC3	0,35	3,26	0,31	3,20	0,35	3,11	0,43	3,27	0,43	2,79
MFCC4	0,25	-0,51	0,25	-0,65	0,24	-0,99	0,33	-0,75	0,61	0,13
MFCC5	0,23	0,38	0,23	0,30	0,20	-0,09	0,28	-0,30	0,36	0,03
MFCC6	0,20	0,06	0,22	0,03	0,20	-0,03	0,31	-0,07	0,32	0,03
MFCC7	0,18	0,28	0,18	0,29	0,18	0,16	0,24	0,24	0,29	0,26
MFCC8	0,20	0,11	0,19	0,08	0,18	-0,09	0,21	0,24	0,32	0,19
MFCC9	0,18	0,13	0,20	0,16	0,18	0,13	0,21	0,47	0,34	0,18
MFCC10	0,17	0,02	0,18	0,03	0,17	0,10	0,20	0,02	0,37	0,14
MFCC11	0,16	0,06	0,17	-0,03	0,18	0,06	0,20	0,18	0,39	0,04
MFCC12	0,17	0,02	0,18	-0,02	0,17	-0,01	0,27	-0,22	0,42	-0,02
MFCC13	0,16	0,03	0,17	-0,03	0,18	0,02	0,18	0,20	0,44	-0,01



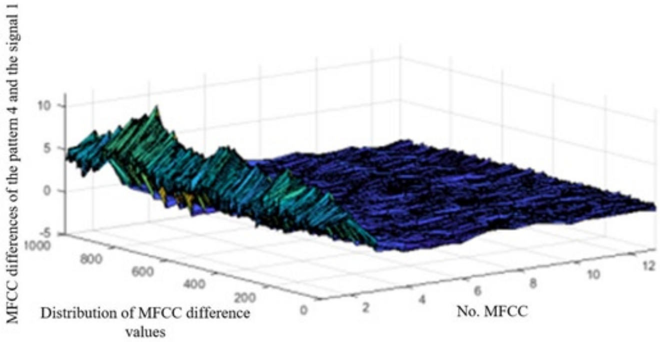
Pattern signal 1/examined signal



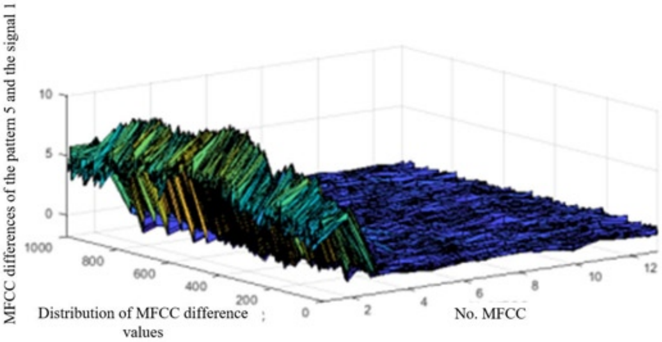
Pattern signal 2/examined signal



Pattern signal 3/examined signal



Pattern signal 4/examined signal



Pattern signal 5/examined signal

Fig. 3. Comparative analysis of MFCC coefficients of the examined signal with pattern signals

level corresponded to one of the following five values, i.e.: 56, 62, 68, 74, 80[dB(A)]. Input data for determining the MFCCn vector were audio pattern signals in the time domain, selected on the basis of a psychoacoustic experiment and representing the appropriate classes of noise annoyance. As a result of the procedure (Fig. 2), 13 mel-cepstral coefficients were calculated for each of the pattern signals and the examined signal. The calculations were carried out in the Matlab R2018a environment using the function (mfcc). Standard deviation (σ) and average ($\bar{x}$) values were determined for each of the coefficients for 1000 data (values were calculated with a step 0,01[s]), (Table 1).

The comparative analysis of the MFCC values of the examined signal with the pattern signals showed that the largest (local) relative differences between the MFCC coefficients occurred when the examined signal was combined with the pattern signal 4 (Fig. 3).

Individual DTW distances between the pattern signals and the examined signal were determined using the function (dtw) in the Mathlab R2018a environment. After substitution into (6), there have been determined DTW distances between MFCC coefficients for individual pattern signals and the examined audio signal DW13a_9_56.

Table 2 shows that the smallest accumulated DTW distance was obtained for the variant of assessment of the examined signal DW13a_9_56

Table 2. The summary cumulative DTW distances of the examined signal with the pattern signals

No.	Type of variant	Accumulated distance DTW
1.	Pattern signal 1/examined signal	197,829
2.	Pattern signal 2/examined signal	249,447
3.	Pattern signal 3/examined signal	304,864
4.	Pattern signal 4/examined signal	496,279
5.	Pattern signal 5/examined signal	537,361

together with the pattern 1, which means that the examined signal has been classified into the noise annoyance class: not at all.

4. Discussion of results and conclusions

The obtained results of the classification of the examined audio signal, based on the assessment of its features, i.e. MFCC coefficients, justify the application of the method of parameterization of the audio signal in modeling of vibroacoustic phenomena of machines / devices, taking into account the psychoacoustic effects of noise. The presented method can be directly used in the tasks of assessing sound sources, shaping employee health, or developing design and construction solutions for noise protection. The proposed method of parameterization of the audio signal expands and complements the applied energetic approach to modeling vibroacoustic phenomena of machines/devices. The possibilities of cepstral analysis based on the characteristics of human hearing (mel-scale), were an important arguments for the author to look for a new way of research in the field of modeling vibroacoustic phenomena. The presented method allows for classification of any audio signal in terms of noise assessment based on the mel-scale of hearing, for the identified source type of vibroacoustic energy. An important advantage of using the method of parameterizing the audio signal is the ability to model cepstrum, which allows you to include noise perception and interpretation of the information contained in the spectrum of the audio signal. The parameters determined in a perceptual scale reflect natural sound experiences, which is important for the assessment of psychoacoustic noise and vibration phenomena. The developed method of modeling vibroacoustic phenomena is based on the assessment of the acoustic/audio signal using the following methods:

- methods of scaling the audio signal (subjective assessment of acoustic impressions as part of a psychoacoustic experiment),
- methods of parameterizing the audio signal (assessment of signal features on the mel-scale), which is an objective description of perceived noise.

The undertaken research will be continued in the area of applicability of methods and models describing the psychoacoustic effects of noise in the time-frequency domain and other perceptual scale, e.g. the bark-scale. The method of modeling vibroacoustic phenomena proposed by the author gives new possibilities in the scope of machine/device assessment, at the stage:

- design and construction, as an additional criterion in the process of optimizing the silent operation of elements,
- exploitation, as part of diagnostic tests for the purposes of verifying the degree of wear of individual components and subassemblies, and, as a result, assessing their reliability level.

The processed information in the form of acquired features from the audio signal is an added value for the acoustic signal and can be effectively used in supporting decision making of the above-mentioned tasks.

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