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V.A. ZELINSKY, Leading Engineer, International Research and Training Center
for Information Technologies and Systems of the NAS and MES of Ukraine,
ave. Acad. Glushkov, 40, Kyiv, 03187, Ukraine, ORCID: https://orcid.org/0000-0001-6464-1702,
zelinsky_v@ukr.net
S.V. LYSOCHENKO, Ph.D of Phys.-Mat. Sciences, Senior Researcher, Educational and
Scientific Institute of High Technologies of Taras Shevchenko National University of Kyiv,
ave. Acad. Glushkov, 4G, Kyiv 03127, Ukraine, ORCID: https://orcid.org/0000-0002-5283-4336,
lysoch@gmail.com
V.V. ILCHENKO, Dr. Doctor of Phys.-Mat. Sciences, Prof., Educational and
Scientific Institute of High Technologies of Taras Shevchenko National University of Kyiv,
ave. Acad. Glushkov, 4G, Kyiv 03127, Ukraine, ORCID: https://orcid.org/0000-0001-5844-2248,
volodymyr.ilchenko@knu.ua
Yu.L. SHEREDEKO, researcher, International Research and Training Center
for Information Technologies and Systems of the NAS and MES of Ukraine,
ave. Acad. Glushkov, 40, Kyiv, 03187, Ukraine,
ORCID: https://orcid.org/0000-0003-4496-2407
semant@i.ua

ALGORITHMS FOR SEPARATING THE SUM OF VIBRATIONS AND IDENTIFYNG THEIR SOURCES

The situation is considered when signals from vibrating objects propagate in an elastic medium and are registered by several sensors remote from each other. The task is to interpret the registered signals in the form of the sum (superposition) of several components. It is assumed known (for physical reasons) belonging the structure of components to one of the types, namely – to harmonic oscillations with certain laws of change in frequency and amplitude. The algorithms of interpretation of the total signal are investigated, their speed and accuracy are estimated.

Keywords: *seismic waves, wave package, superposition principle, division of the signal into components, identification of signal sources.*

Introduction

There are many practical situations where an investigated object vibrates and the waves from these vibrations propagate in an elastic medium and can be measured by sensors placed in certain coordinates at a distance from each other. The task is to distinguish waves from different sources (each object can emit several wave packets of different frequencies), to determine the position of these sources and their properties. In this paper, the general structure of the system for solving these problems is considered. The main focus is on the problems of separating the total signal received on sensors from different sources into components of a certain structure.

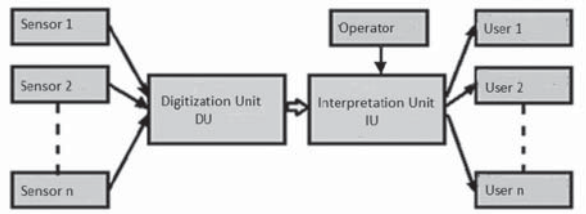

Figure. Scheme of registration and interpenetration of seismic waves

The General Structure of the Vibration Interpretation System

The general formulation described in the introduction also includes the practical task of building a security system that registers seismic waves propagating over the soil surface from moving objects and identifies them. Further, using this example, the problem of separating signals and identifying vibration sources will be considered. When considering, we will take into account the peculiarities of both the sources themselves and the propagation of seismic waves over the soil surface, but the general structure of the algorithms can be used for other similar practical situations. The scheme of registration and interpretation of seismic waves, shown in Figure, is also quite common. Further, its components are considered.

Sensors. Seismic wave sensors are placed in fixed coordinates on the soil surface. Distances between sensors are chosen for the following reasons: the shorter the distance between the sensors, the more similar the signal they registered will be (which is desirable for its separation and identification of sources), but lower accuracy in determi-ning the delays of the same signal front on different sensors, which reduces the accuracy of determining coordinates. In particular, the work [1] examines the placement of 4 sensors in the vertices of a square on the soil surface with a side length of 20 meters. The following requirements for the sensor system are important:

– the coefficient of transmission of sensors for the frequencies of the input signal, within the range allocated for measurement, must be the same. That is, the ratio of the amplitudes of the selected frequency band (other frequencies are filtered) that come from the source should not be distorted by the sensors since they contain information about the source;

– to get a signal from the sensor without distortion, the amplitude characteristic of the sensors should be linear. The very environment transmit significant mechanical vibrations with non-linear distortions that are usually not taken into account when processing. Accordingly, the upper limit of the input signal of the sensors, where it is desirable to maintain the linearity of its amplitude characteristic is limited to this indicator for the environment.

Digitization Unit. Sensors usually registere a weak signal in analog form. For processing on a computer, signals from sensors must be converted into a digital form, which is carried out by a digitization unit (DU), the main element of which is an analog-to-digital converter (ADC).

Before digitization, signals from sensors must be filtered from needless frequencies whose limits is set from the following considerations:

- higher frequencies fade faster when propagated in the medium, which means they may not reach the sensors from sources outside a required distance – too high frequencies cannot be used to detect such sources, so they should not be processed;

- higher frequencies require higher ADC sampling frequency, and therefore higher processing speed. With insufficient handler power it is worth refusing to use too high frequencies;

- identifying a source based on low frequencies requires more time (at least the algorithms described below suggest just such a dependence). Having set the minimum time to identify the source (for example, 1 second) and the number of oscillations in the signal that are needed to detect the presence of a source (for example, 10 periods), we determine the lower limit of the desired frequencies (we get 10 Hz).

High frequency filtering usually occurs immediately before the ADC to get rid of additional high frequencies, which appear as noise in the process of transporting signals from sensors to DU.

Filtration of lower frequencies, in particular when using piezo sensors, can be provided directly

at the output of the sensor: the greater the load resistance, the lower the frequencies which is left [2].

Digitization should take place in parallel for all connected sensors. That is, it is necessary to obtain n numbers that characterize the measurement of the signal of each of the n sensors in the same discrete of time.

Interpretation Unit. Measurement data must be accumulated over a period of time; the greater the number of measurements compared in the process of their interpretation, the better the result can be achieved. On the other hand, the time of accumulation of measurements is one of the components of the delay in identifying the source of vibrations and therefore it is necessary to determine its compromise value.

A similar delay for the accumulation of a measurement packet is appeared when transferring measurements from the DU to the interpretation unit (IU), if they are located at a long distance. Transmitting a stream of measurements to an IU located at a considerable distance from the DU will allow the IU to work with multiple DU's and cover a large area of observation.

It may be acceptable to refuse to transfer measurements over a long distance and combine IU and DU in one microcontroller. Let's try to justify such a decision:

- the capabilities and power of modern non-expensive microcontrollers allow you to perform processing which IU needs quickly enough; they are small in size, can be powered by a battery for a long time without recharging, can transmit the results of processing over a wired (Ethernet) or over wireless (WiFi) network. That is, the microcontroller can be placed next to the sensors and work autonomously, providing information about the sources of vibrations for mobile users at a distance of the wireless network;

- it is possible to cover large areas by observation not only by connecting several DU to one IU, but also by combining IU into a network. It is enough for the user to connect to one IU in order to obtain information about source, which observed by at least one of the IU, distributed over the territory. To do this IUs exchanges measurement processing results and deliver information to the user in the area of his interest;

- implementation of DU and IU in one microcontroller completely eliminates the need to accumulate a measurement packet for its remote transmission, and hence the delay associated with the accumulation of such a packet. Measurements after ADC can be perceived without delay by the microcontroller in interrupting from the timer asynchronously to the processing. The time spent on interruptions can be neglected because their frequency is equal to the sample rate of the analog signal is much (thousands of times) less than the frequency of the microcontroller processor. Read measurement values from ADC fill the buffer for processing at the rate of measurements – without delays.

- data transmission over distances occurs only after their processing – IU transmits only the results of measurement processing to users and, possibly, to other IU when they are combined into a network. The digitized flow of measurements, the dimensions of which are much larger than the results of its processing, is processed at the place of its occurrence – after the ADC. Thus, network traffic is significantly reduced compared to the option of remote DU and IU.

Given the preferences specified of association DU and IU, in the future we will orient ourselves for such a decision.

Operator. The operator (group of experts) sets the parameters for the operation of DU and IU, in particular, the sample rate of measurements, the number of levels of discretization of signal, the coordinates of sensors, the speed of wave propagation, data for signal recognition, etc., and starts-stops the session of its operation. Evaluations and calculations of parameters are carried out based on the results of interpretation of measurements accumulated in previous sessions.

Users. Users of the system we will call executive devices that respond to certain situations in the observation zone, and programs that reflect the results of the interpretation of measurements in a form convenient for human perception. In particular, such programs can be hosted on several remote computers (including mobile ones) and receive information from the IU via WiFi and Ethernet. Users can receive information about the vibrating sources that are in the observation zone of all UIs

connected to the network. The program for archiving the results of IU's work in order to analyze them by the operator, to adjust the parameters of the next session we also consider as a user.

Algorithm for Separating the Components of the Total Signal

Streaming Processing of Measurements. Measurement processing should take place in real time – with minimal delay and at the pace of their receipt. To do this, IU implements two separate processes: reading and processing, which have a common memory area – n cyclic buffers (CB) , one for measurements from each sensor.

Reading:

- starts by interruption from the timer, reads measurements from ADC (from all sensors almost simultaneously);

- distributes measurements to the CB of the corresponding sensors and adds them on free space to those previously recorded. When the end of the CB is reached, the data is written from its beginning to the place that is released after they are processed.

Processing:

- launched by the operator at the beginning of the IU session, asynchronously interrupted to record the next measurement (of all sensors) by the reading process and completed by the operator with the corresponding command;

- in parallel processes measurements of all CB and creates a list of sources of vibrations with their coordinates and belonging to classes;

- frees CB from processed data to fill them with new measurement during the reading process;

- transmits information about the detected sources to users.

Processing must have time to release CB before it is filled with new measurements. The dimensions of the CB are the same and must be sufficient to accommodate data that is interpreted simultaneously, as well as have sufficient reserve given the uneven processing.

Portrait of the Source of Vibrations. It was found above that the input data for the separation algorithm is a signal formed as a result of the imposition of components from several sources registered by each of the n sensors. As such components, we take sine waves that can vary in frequency, amplitude and relative position over time. The source generates several of the following components that overlap each other, creating a portrait that can be expressed as a model:

$$P = \sum_{k=1}^{K} A_k(t) \cdot \sin(\omega_k(t) \cdot t\text{-}\varphi_k), \quad t^s{}_k < t < t^e{}_k, \quad (1)$$

where t is the moment of time (discrete of measurement); $A_k(t)$, $\omega_k(t)$ – amplitude and frequency of the k-th component, respectively, (are not permanent, depend on time); φ_k – phase of the k-th component; $t^s{}_k$, $t^e{}_k$ – moments of the starting and ending of the component; K – number of components in source portrait. In the future, such components of the portrait will be denoted by an abbreviation PC.

Assumptions about the structure of the portrait (a list of interdependent parameters) are based on the implementation of Hooke's law for most environments in which vibrations are created and propagated – proportionality of the force arising in the material when it is compressed or stretched to the value of such deformations. In particular, the operation of the technical system (TS) is accompanied by collisions (shocks, friction) of structural elements, which means deviations from the equilibrium position of the corresponding material, followed by their oscillations with its own frequency and attenuation. The frequencies of collisions during the operation of the TS, the natural frequencies of its structural elements, the degree of attenuation of oscillations and the ratio of their amplitudes can be signs of the TS class since they depend on their design. Usually, TS change their operating mode, which means that the amplitudes and oscillation frequencies change. The external elastic medium (soil, air, etc.) perceives the vibrations of sources through tangential surfaces – vibrations are propagated and registered by sensors. A person or an animal, when walking strikes directly on the surface of the soil – the oscillations of soil are occurs with its own frequency.

In general, different frequencies propagate with different attenuation and speeds, which distorts the signals received on the sensors. But in any case the structure of portrait P remains unchanged. Usually PC parameters change smoothly and limited – sud-

den changes do not occur often (they are taken into account as separate PC). By separation the portrait on PCs, the changes of which occur smoothly, we will ensure the proximity of the frequencies on which each of PC decomposes. Since the close frequencies have close speeds and attenuation, then the distortion of the PC will be minimal and hence the minimal dependence of the parameters of the PC from the distance to the source. Significant differences can be between the parameters of different PC, frequencies of which differ significantly. Thus, if the frequencies of PCs, which is belong to same source, differ significantly, then the ratio of amplitudes or shifts in time will be different for these PCs and depends of the distance to this source. This fact can be taken into account when classifying or used to clarify the distance to the source.

We will interpret PC in the form of sinusoidal oscillations, the frequency and amplitude of which changes smoothly according to the piece-linear law. Besides, PC can increase and fall from / to zero amplitude values, as well as occur and break off. Let's clarify the structure of functions $A_k(t)$ and $k(t)$ from (1) as follows expressions:

$$A_k(t) = \{(a_{ki} + b_{ki} \cdot t;), t^s_{ki} < t < t^e_{ki}\}, \quad 0 <= i < I_k; \tag{2'}$$

$$\omega_k(t) = \{(c_{kj} + d_{kj} \cdot t), t^s_{kj} < t < t^e_{kj})\}, \quad 0 <= j < J_k, \tag{2''}$$

where $a_{ki/j}$, $b_{ki/j}$, $c_{ki/j}$, $d_{ki/j}$ – parameters of linear dependence of amplitude and frequency of oscillation from time; index *i/j* indicates a parameter belonging to a certain linear area; *I/J* – number of linear areas; $t^s_{ki/j}$, $t^e_{ki/j}$ – the moment of start and end (break) of a linear area. The end of the previous plot must be equal to the beginning of the next, i.e. $t^e_{ki} = t^s_{k(i-1)} (0 < i < I)$; $t^e_{kj} = t^s_{k(j-1)} (0 < j < J)$. The total length of the period of interpretation of the frequency and amplitude for the *k*-th component of the CP must be the same, i.e.

Superposition Requirement. Let's put an additional requirement on the signals perceived by the sensors – they should not be distorted by the transmission medium depending on what other signals they are superimposed (superposition principle). Usually for elastic environments, the principle of superposition is performed in the case of waves with a small amplitude. For significant deviations (compression-stretching) of the medium from the equilibrium position, nonlinear distortions arise – it is almost impossible to restore several superimposed signals in such conditions. In the case of adherence to the principle of superposition, the model of signal, reaching the IU can be expressed as the sum of the portraits of individual sources: $\sum P_g =$

$$= \sum_{g=1..G} \Big(\sum_{k=1..K} A_{gk}(t) \cdot \sin(\omega_{gk}(t) \cdot t - \varphi_{gk}) \Big), \tag{3}$$

where $A_{gk}(t)$, $\omega_{gk}(t)$ – amplitude and frequency the *k*-th component of the *g*-th source, respectively; φ_{gk} – phase of the *k*-th component of *g*-th - sources; *K* – the number of components in the portrait of the source; *G* – number of sources that are registered.

Practically, if the total signal from the source exceeds the level within which the principle of superposition is performed, then such areas must be excluded from registration by sensors; the IU should receive the appropriate attribute.

Algorithm A1. Based on streaming processing of measurements and the known structure of the components (3), the separation of the components of the total signal can be performed by the following sequence of steps:

0. Prepare measurements in CB: according to the operator's command, at the beginning of the observation session, start digitizing the measurements registered by the sensors. In an interrupt from the timer, measurements from the ADC are entered into CB in the order they arrive at a given sample rate. Initial measurements whose values do not exceed the noise level are discarded – the recording begins with the first dimension, the value of which exceeds the specified level *aMin*. In all *n* CB, *aMin* must be exceeded within a time interval not exceeding the maximum relative delay of the front of the same signal on different sensors (*zMax*). If this did not happen for the first recorded measurements, then they are discarded. Upon reaching the number of registered measurements of a given value (*kMin*), their processing starts in parallel in all *n* CB.

1. Refer to the beginning of the filled but not processed part of the CB where the essential values of the measurements remain – their level exceeds *aMin*.

2. Define options for one of the PC, which is present in the total signal according to the expression (3). Begin with generating several alternatives for each of the *n* CB. Among them, the one that has the highest appreciation of plausibility is chosen. The criteria are: sufficiency of measurements; simplicity of the model; presence in all CB; belonging to a previously defined source; belonging to a known class of sources; minimization of residue. The time shift of the selected PC in different CBs is used to determine the coordinates of its source in the manner described in [1]. PCs for which the same coordinates are determined will be attributed to the same source. The class of source, to which the defined PC is assigned, is determined or clarified. Found source parameters are passed on to users.

3. Clear the buffer from the presence of the PC defined at the previous stage. The parameters of the defined PC must be sufficient to reproduce all the measurements that it introduces into the total signal.

4. Estimate buffer residue: move the beginning of the untreated part of CB to a discrete of time earlier of which the residue is less than *aMin*. Add to CB the measurements, accumulated in timer interruption. Define a new value for the number of measurements in CB. If there are not enough of them, wait for replenishment, until the value is reached *kMin*.

5. Return to step 1 with updated content of CB.

The A1 algorithm can be executed for an arbitrary time until the operator stops its operation. CB is filled with new measurements and is simultaneously released from the PC, which was determined in step 2.

Alternative Algorithms for Distinguishing Sources. Next, we consider possible alternatives to the algorithm A1 proposed earlier.

Algorithm A2. The traditional approach to the analysis of a complex signals of any form – its representation by the Fourier series – a set of sinusoids with different (but constant) phase, amplitude and frequency. There are well-developed algorithms that allow you to get the frequency and phase spectrum of the signal. The interpretation algorithm could be as follows:

0. Fill with measurements a set of buffers for their simultaneous processing; for each sensor provide identical buffers (similar to step 0 of algorithm A1). Switch processing to measurement-filled buffers. Another similar set of buffers is switched to filling with new measurements that will be registered during processing in an interrupt from the timer.

1. Obtain the frequency and phase spectrum of the signal for each buffer, using the discrete Fourier transform. For different sensors, frequency spectra are expected to be similar as the same frequencies arrive at different sensors with small delays in time.

2. Choose several frequencies (convenient with the largest amplitudes). Selected frequencies, their amplitudes and phases can be taken as attributes, by the ratio of which to identify the presence in the total signal of a source of a certain class.

3. By delays of phases of the same frequency on different sensors, the coordinates of the source that generates this frequency are determined. On the other hand, the frequencies for which the same (with an acceptable error) coordinates were determined belongs to the same source.

4. Switch buffers when they are ready: those that were filled with measurements will become processed and vice versa – those that were processed will accumulate new measurements for the next processing.

5. Back to step 1.

Algorithm A2 has the following drawbacks: *a*) Fourier transform is quite time-consuming; *b*) the real components of the total signal received on the sensors are sinusoidal, but with changes in amplitudes and frequencies – their interpretation in the form of sinusoids with fixed amplitudes and frequencies is a roughening of the real signal model; *c*) it is problematic to determine the delays of the same signal front on different sensors – phase shift of the same frequency on different sensors can only be determined with accuracy to the period of this frequency; d) for the Fourier transform, a two measurement buffers is needed – after processing one of them, a new buffer is used. Measurements that enter the new buffer before it is filled are not taken into account for processing.

Algorithm A3. Wavelet conversion [3] differs from the Fourier series decomposition by using a wider class of basic functions (not just a sine wave) into which a signal from the sensors can be decomposed. Each of these functions has several param-

eters, including placement on the time axis. Wavelet obtained on different sensors can be compared, determine their identity and shift of time, on the basis of which to determine the coordinates of the source. As basic functions can be used waves of various shapes with zero integral value, localized in time, with the possibility of shift and scale (squeezing and stretching) along the time and amplitude. Having chosen PC from the ratio (1) as the basic functions, we have the opportunity to use the wavelet transformation apparatus. The use of the A3 algorithm similar to A2 but it allows obtaining better results due to a more adequate component model.

In the next paragraph, we will focus on detailing the A1 algorithm, which improves the approaches of the A2 and A3 algorithms by: *a*) use of PC structures close to real ones – those that are consistent with substantive (physical) considerations; *b*) use of substantive (not only formal) criteria in determining the composition of the total signal; *c*) controlled roughening of both the settlement process and the PC model for speedup the processing; *d*) using the registered measurements without waiting for the next buffer to be filled in for processing.

Selection of a Component from the Total Signal. The most uncertain is the 2nd step of the algorithm A1; the other steps are quite simple – we will not detail them.

We will consider the total signal as consisting of only one component of the known structure according to expressions (2) – we consider the other components to be interference. We will look for the parameter values of this component and remove it from the total signal in step 3 of the A1 algorithm; next we continue processing the residue in a similar way.

In the works [4, 5] it is proved that to reproduce a signal that is the result of the imposition of a sine waves, its values at certain points, which are easily to found in the total signal, are sufficient. These are the points at which the first derivative takes a zero value – local extremes on the total signal graph. To determine them, it is enough to compare the values of three consecutive measurements: if the one in the middle is more than both of its neighbors, then this is the local maximum; if the one in the middle is less than the neighbors, then this is the minimum. Operations to determine extremes and enter them into a separate buffer (similar to the CB for measurements) are performed when each next measurement is introduced in an interrupt from the timer. Thus, the CB for extremes will contains the serial numbers of selected measurements.

The extremes of the desired PC are contained among their general list in the total signal, but which ones? The general approach is to find the PC, the presence of which in the total signal is most probable. To identify this, we will generate variants of PC, evaluate them and choose the one that best meets the probability criteria given in the description of step 2 of the A1 algorithm. Note that the formal criteria proposed by the A2 and A3 algorithms are aimed only at maximizing the reduction of the signal residue. We strive to use additional meaningful information that will allow not only to more accurately identify the source, but also to simplify the calculations.

The construction of PC variants is divided into two stages: I – selection of sequences of extremes (chains) and approximation of their position by function (2''); II – for the obtained chains, determining the amplitude of each extreme and approximating them with a function (2').

Stage I:

1. From the general list of extremes present in the CB, we select their sequences so that the distance between neighboring ones in time changes quite smoothly. To do this, we will set several initial (with lower time values) pairs of extremes, from the ends of which we will generate the following (the end of the previous pair is the beginning of the next) so that the distance between the beginning and the end of the pair varies within acceptable limits. The more initial pairs are given and the less stringent conditions for the selection of subsequent pairs will be applied, the more chain options we will get. Chains with fewer, than a given minimum number of extremes are discarded because the parameters of the corresponding PC cannot be determined with sufficient reliability.

2. We approximate each chain obtained at the previous step with a piece-linear function (2'') and define its parameters. For approximation, we will use the least squares method (LSM), given by the total permissible deviation of the values of the distances between the extremes from the approximating function.

3. Let's discard chains with too many kinks of the approximating line (*J* in the expression 2''). Thus, we take into account that the frequency of a real PC usually either increases or decreases over a long time. But at the same time we leave a certain minimum number of chains for further consideration.

4. For each of the remaining chains, we fix the extremes, the amplitude value of which is maximum (we consider it the most reliable). Other extremes will be placed from the fixed at a distance (along the time axis) determined as a result of approximation in step 2 and rounded to the nearest discrete. Thus, we get the options for the positions of the extremes of the search PC. Note that neighboring extremes are opposite: one of them is necessarily the maximum, the other – the minimum.

Stage II:

1. For each chain obtained at the previous stage, we find the difference in measurement values for neighboring extremes. For an extremes with the serial number *v*, two differences will be determined: with the previous r_{v-1} and the subsequent r_{v+1} measurement value. As the value of the extremes v we take half the modulus of their average value $R_v = |r_{v-1}+r_{v+1}| / 4$. For the initial and last extremes, we will use only one neighboring: $R_0 = |r1| / 2$; RI-1= = |rI-2| / 2. The module is used to turn the lower half-waves of the sinusoid to the top because they are dependent on the upper neighbors and require compatible approximation.

2. We approximate the envelope of each measurement sequence obtained in the previous step by the function (2') similar to step 2 of the previous stage. Thus, for each chain, the calculated values of the amplitudes will be determined – these are models on the basis of which it is possible to build sinusoids of the corresponding PC.

3. Let's discard the PC variants with too many kinks of the approximating line (*I* in the expression 2'). Thus, we take into account that the amplitude of a real PC usually either increases or decreases over a long time. But at the same time, we leave a certain minimum number of PC for further consideration.

Thus, we obtain variants of PC models for all n CB, each of which is given by parameters from expressions 2' and 2''. Among them you need to choose one and remove it from the total signal from all CB. Evaluation of the probability of the presence of PC in the total signal will be carried out according to the criteria given in step 2 of the algorithm A1.

1. The criterion of a sufficient number of measurements is applied in the step 1 of the algorithm of the stage I. This criterion takes into account situations: *a*) the beginning of an observation session, when the chain, the beginning of which has not been recorded, ends with an insufficient number of measurements; *b*) short PC in a real signal; *c*) short sequels to the already removed PC (their joint with previously removed unjustifiably complicates the algorithm). In all these cases, short chains of extremes are discarded without building a CP. The remaining measurements in the CB are erased in step 4 of the A1 algorithm as new measurements appear or when the left signal moves away from other measurements for a period exceeding *zMax*.

2. The criterion of simplicity was applied at the 3rd step of the algorithms of the I-th and II stages in the construction of PC variants – their most complex models were rejected.

3. The criterion of presence of all sensors in the CB is provided by pairwise comparison of PC variants in different CBs by the following sequence of steps:

– One of the n CB we choose as a base. PCs that are defined for it will be called basic.

– For each basic PC, its parameters are compared with the PC parameters contained in other CB. Parameters for comparison are selected in a specific sequence. For each parameter, the limits of permissible deviations are set: the PC, value of the parameter of which is outside the permissible, is discarded – its other parameters are not checked. If for the basic PC there were no similar ones in all *n* CB at the same time, then it is rejected. If all basic PC are rejected, then the CB of measurements and corresponding extremes is supplemented with new measurements and the construction of the PC begins again.

– As a result, samples will be formed from *n* similar PC – one from each CB, representing the same PC, but registered with different sensors. For each sample, there is a general score as the sum of weighted deviations of the parameters of the PC in

its composition. Samples with minimal ratings are rejected. At this stage, it is possible to limit ourselves to one sample with the maximum estimate, but if there is a reserve of time for calculations and there are options with close maximum estimates, then it is possible to continue to clarify the estimates.

– For each of the left samples, we average the value of the parameters of its components – we get the parameters of the corresponding PC. PC in the composition of the sample may differ in position on the time axis and amplitude, which depends on the coordinates of the source and are not subject to averaging. It is by these parameters of the PC sample, that the coordinates of the source are determined, according to the algorithm described in the work [1].

4. Criterion of belonging to a previously determined source (if any). Attribute each of the PC defined in the previous step to the source, with the nearest coordinates. Raise the estimate of PC, the determined coordinates of which are at a real distance from the coordinates of previously determined sources.

5. Criterion of belonging to a known class of sources. Each class is characterized by several areas in the feature space. The values of the features can be determined for each source based on the parameters of the CP, which were attributed to its composition. A source can be attributed to several classes with a certain degree of belonging. Let's take the following: if the «certainty of a source's belonging to a class» increases with the inclusion of additional CP, then the estimate of this CP increases.

6. Criterion of minimizing the residue after removing the PC from the total signal. The calculation of the residue can be performed after averaging the values of the parameters of the PC (after criterion 3). But since these calculations are the most time-consuming, it is advisable to perform them at the end, when there are a minimum of PC options left.

Conclusions

The proposed algorithm for separating the sum of vibrations A1 is based on classical approaches: Fourier and wavelet transformation; the ideas of the superfast Fourier transform (SFFT) [4, 5] are used to the greatest extent. In particular, the idea of selection from the total signal only its individual values – namely extremes, is borrowed from the latter. These are significantly reduces the input data for analysis and accelerates their processing. A sequential selection of sinusoids from the total signal, proposed by SFFT, has been extended to a selection of more complex functions, namely the sinusoid in which the frequency and amplitude change slowly. Such functions adequately represent the components of the original signal, which is registered by vibration sensors. To reliably determine each new sinusoid extracted from the total signal, we evaluate several of their variants based on meaningful criteria.

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В.А. Зелінський, провідний інженер, Міжнародний науково-навчальний центр
інформаційних технологій і систем НАН та МОН України,
03187, м. Київ, просп. Акад. Глушкова,40, Україна, ORCID: https://orcid.org/0000-0001-6464-1702,
zelinsky_v@ukr.net
С.В. Лисоченко, канд. фіз.-мат. наук, ст. наук. співробітник, Навчально-науковий інститут
високих технологій Київського національного університету імені Тараса Шевченка,
03127, м. Київ, просп. Акад. Глушкова, 4Г, Україна, ORCID: https://orcid.org/0000-0002-5283-4336,
lysoch@gmail.com
В.В. Ільченко, доктор фіз-мат. наук, професор, Навчально-науковий інститут
високих технологій Київського національного університету імені Тараса Шевченка,
03127, м. Київ, просп. Акад. Глушкова, 4Г, Україна, ORCID: https://orcid.org/0000-0001-5844-2248,
volodymyr.ilchenko@knu.ua
Ю.Л. Шередеко, науковий. співробітник, Міжнародний науково-навчальний центр
інформаційних технологій і систем НАН та МОН України,
03187, м. Київ, просп. Акад. Глушкова,40, Україна, ORCID: https://orcid.org/0000-0003-4496-2407
semant@i.ua

АЛГОРИТМИ РОЗДІЛЕННЯ СУМИ ВІБРАЦІЙ ТА ІДЕНТИФІКАЦІЇ ЇХНІХ ДЖЕРЕЛ

Вступ. Існує багато практичних ситуацій, коли досліджуваний об'єкт вібрує, а хвилі від цих вібрацій поширюються у пружному середовищі і можуть бути зареєстровані датчиками, розміщеними у певних координатах на відстані один від одного. Ставиться задача розрізнити хвилі від різних джерел (кожен об'єкт може випромінювати декілька хвильових пакетів різної частоти), визначити положення цих джерел та їх властивості.

В роботі розглядається загальна структура системи для вирішення вказаних задач. Основна увага зосереджена на проблемах розділення сумарного сигналу, отриманого на кожному датчику від різних джерел, на складові, та розпізнаванню цих джерел – визначенні класів, до яких вони належать.

Метою статті є аналіз існуючих підходів до розділення сигналу на складові певної структури. Пропонується оригінальний підхід і відповідний алгоритм, який його реалізує. Перевагою запропонованого алгоритму є швидкість роботи та здатність інтерпретувати сигнал суперпозицією компонент, які реально генеруються вібруючими джерелами.

Методи. Робота базується на ідеях розкладу сигналу на складові, зокрема Фур'є та вейвлет-перетворення. Використано методи над швидкого перетворення Фур'є, методи кусочно-лінійної апроксимації, традиційні підходи розпізнавання образів та комп'ютерне моделювання.

Результати. Обгрунтовано і реалізовано алгоритм інтерпретації сигналу у вигляді суми коливань близьких до гармонійних. В алгоритмі використана ідея над швидкого перетворення Фур'є з модифікаціями, які дозволяють визначати для складових положення у часі, зміни частоти та амплітуди. Виявлені складові розподіляються за джерелами; їхні параметри використовуються як ознаки для розпізнавання – визначення класу кожного джерела. Алгоритм є ефективним для визначення координат та швидкості рухомих об'єктів у реальному часі.

Висновки. Запропонований алгоритм розділення суми вібрацій А1 базується на класичних підходах: Фур'є та вейвлет-перетворення; в найбільшому ступені використані ідеї над швидкого перетворення Фур'є (НШПФ). Зокрема з останнього запозичена ідея вибірки із сумарного сигналу лише окремих його значень – а саме екстремумів, що суттєво скорочує вхідні дані для аналізу і пришвидшує їх обробку. Послідовна вибірка із сумарного сигналу синусоїд, запропонована НШПФ була розширена до вибірки більш складних функцій, а саме синусоїд, в яких частота та амплітуда повільно змінюються. Такі функції адекватно представляють складові оригінального сигналу, який реєструється датчиками вібрацій. Для надійного визначення кожної нової синусоїди, що виймається із сумарного сигналу, оцінюємо декілька їх варіантів на основі змістовних критеріїв.

Ключові слова: *сейсмічні хвилі, хвильовий пакет, принцип суперпозиції, розділення сигналу на складові, ідентифікація джерел сигналів.*