PHILOLOGICAL SCIENCES

PHILOSOPHY OF SOUND: WAVELET TRANSFORM FOR PROCESSING OF AUDIO SIGNAL

Nuradin G.,
candidate of philosophical Sciences, Associate Professor,
Taraz regional university named after M.H. Dulaty,
Kazakhstan, Taraz
Adilbek B.
Master student,
Taraz regional university named after M.H. Dulaty,
Kazakhstan, Taraz

ABSTRACT

Research objects: Sound philosophy. Processing audio data based on the use of the Wavelet Transform and their philosophy

Aim of the search: The main issues under consideration relate to the nature of sounds. Research of the principles of signal noise purification using wavelet transforms, evaluation of the quality of the restored signal and determination of the best order of Dobeshi wavelets, the threshold method, furthermore determine the number of decomposition levels.

Approaches to research: The research used philosophy of sound and methods of mathematical analysis, Fourier's fast transformation algorithm, continuous and discrete wave transformations. The significant part of the research were the deeper search of sound philosophy and presented by computer experiments on processing specific signals, aimed at obtaining the all-important statistical data and determining the characteristics of final algorithms.

Relevance of work: The successful implementation of the prospects for the development of info telecommunication Technologies is largely based on the achievements of digital signal processing, designed to solve the problems of generating, receiving, transmitting and processing information in real time. The implementation of complex algorithms for processing digital signals requires the use of effective basic algorithms (spectral analysis, filtering, compression and signal synthesis) that use the relevant technical process sparingly. Among other tasks of processing digital signals during transmission over radio communication channels, processing methods, as well as compression and recovery of transmitted signals with minimal distortion, are particularly relevant.

Keywords: audio signals, sound philosophy, speech recognition system, Fast Fourier transform, wavelet transform, spectrogram, segmentation.

INTRODUCTION

Advanced digital information transmission systems contribute various functional content services to a large number of collective and individual users. At the same time, the fundamental problem of creating digital communication systems is the processing of transmitted information, reducing redundancy and restoring it.The development of the advanced methods and devices for processing and compressing audio information is a prerequisite for more efficient use of communication channels that make certain the preservation of existing frequency plans, the output of part of the frequency spectrum for the transmission of additional types.

Customer service through mobile and satellite communication systems, multi-program Television, high-definition television, multi-program audio broadcasting, organisation of interactive communication systems, video conferencing, etc.

Due to the active development of digital information processing systems, the development of signal compression algorithms in communication systems based on modern computing methods has recently become relevant. One of them is wavelet signal transforms.

Due to the above, one of the most all-powerful is wavelet transform and is flexible tools for the study and digital processing of signals: in addition to their filtering and compression capabilities, analysis based on wavelet functions allows you to resolve problems. Identification, modelling, approximation of stationary and non-stationary processes, study of enquiry about the presence of spaces in derivatives, search for data connection points, search for signs of fragmentation of information. Such opportunities which has future, which provide a further encouraging wavelet transform, are based on the nature of its multi-scale nature.

Philosophy of sound

When a person opens the door to the world, he hears the very first sound, and even at the very last moment. And from the moment of birth to the moment of entry into the Black Earth, a whole life passes between them. All this is based on the cacophony of noise, tones, rumbling, music, sounds in general.

But today it is found that sounds and noises are completely different concepts, since they have different origins. Of course, both noise and sound are consequences of mechanical vibrations of a wave nature. It's just that noise is a combination of many sounds, and sound is one wave, that is, vibrations perceived by the senses of animals and humans.

As it has been known since ancient times, there is no absolute peace in space. The importance of auditory information was known even before life appeared on earth. The absence of a direct dependence of sound on the openness or closeness of space, light and shadow, etc. indicates its solidity and reliability as an information carrier. Sound consists of direct waves, through which it carries sound information.

The main issues which are on the table concern the nature of sounds. Sounds enter the content of auditory perception. But what are they? Are they events? What is the relation between sounds and wavelets? However, it turns out that a fruitful way to organise these issues deals with the wavelet transform of sounds.

The word is silver, and silence is gold. However, the background of the word also depends on what prism we look at. There is a place on our planet that can drive anyone crazy with its silence. This is an echo-free camera in the Orfield Laboratory in Minneapolis, USA [Orfield Laboratory in Minneapolis, USA, 2015].

Orfield's echo-free camera is the quietest place in the world, meaning 99.9% echo-free. In an echo-free chamber, a room whose walls do not echo the sound. In addition, the walls of such a chamber are designed in such a way as not to miss external sounds. Fully insulated. It is not wrong to call this the place where modern torture studies are conducted, and to call it hell on earth. "If the neighbours that prevent you from sleeping at night hit your anger, believe me, the world without sound is much scarier." As mentioned above, 99.9% without echoes, in a room free of sound waves, if a person can breathe for 45 minutes, the remaining 0.1% is terrible. This room takes first place in the Guinness Book of records, but being in it for too long can cause hallucinations in a person. If we summarise from this, a person cannot live without noise and echoes. Automatically leads to a rhetorical question. Hence the question of how the deaf live. For example: the famous composer Beethoven is known to be deaf, but he can create great works. How? He listened through his teeth! The composer put the end of the stick on the piano, and clenched the other end into his teeth - this way the sound reaches the inner ear, which is absolutely healthy than the outer ear of the composer (https://www.classicfm.com/composers/beethoven/guides/deaf-hearing-loss-composing/).

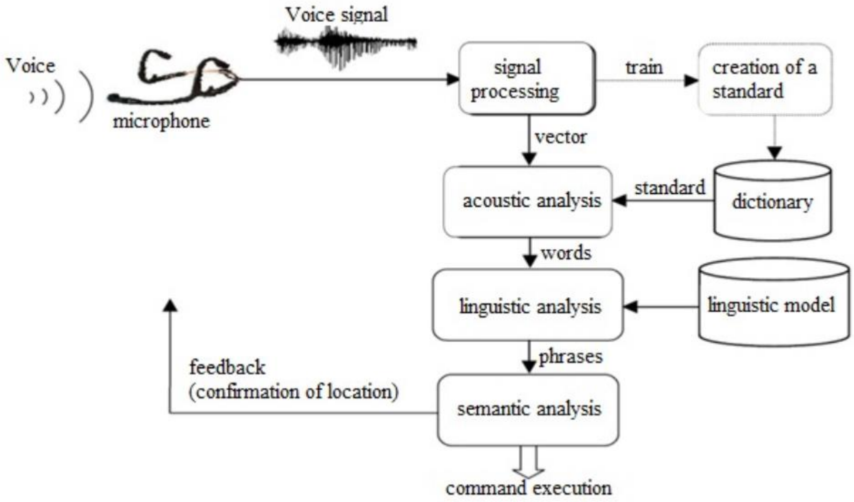
Sound is a wave, as it travels through the air. The first thing comes to our mind when we hear about a wave is a sea. Every sound that comes out of our speech is like the waves of the Sea moving in endless waves. Waves are formed when the wind hits the surface of the water. In the open ocean, waves rise to a number of heights. As the wave approaches the shore, its lower part hits the bottom, and due to the friction force, the movement of the wave slows down, and the movement of the ridge continues. The wave narrows, rises high and falls to the shore. Every sound that comes out of us, like a wave, at every moment takes on high and low tones. If we study this process, then when a wave approaches the shore, its lower part hits the bottom, after a high tone, just like a sound, the tone of the sound necessarily decreases, and then an Echo is heard. Waves are best ways to research sound. Of course, by hearing the sound through our ears, we can tell if the sound is loud or just coming out. In these representations we can study the sound through the wavelet transformations, see the structure of the sound more clearly, and see the graph more clearly. Sound is a complex phenomenon. When considering it consists of different levels. When studying signals, you can use Wavelet Transform, Fourier transform to create a wave graph, control by sound, filter, and clean the sound from noise.

Formation of sound signals

All methods of solving the difficulty of recognising sound recording signals can be divided into two types: discriminant and structural. The essence of the discriminant approach is to form a space of signs of speech images, in which complementary speech images form a common population – taxa or clusters. This approach has a number of disadvantages. fundamental of all, due to the limited power of the training sample, this leads to the use of estimates instead of true statistical characteristics of the probability density function for each cluster, which show the way to a violation of the optimal conditions of classifiers based on statistically conclusive criteria, and in consequence recognition errors. Secondly, this method cannot be applied straightaway to speech signals in unified sound recognition difficulty due to the altitude onus variability of natural speech and, as a result, the inability to build a clear learning model with all imaginable precedents.

Secondly, this method cannot be applied straightaway to speech signals in unified sound recognition difficulty due to the altitude onus variability of natural speech and, as a result, the inability to build a clear learning model with all imaginable precedents.

Voice control systems belong entirely to bionic methods of object control. Indeed, the technology of processing audio signals in voice control systems is a set of methods and tools for implementing the language model –the typical sequence of actions (stages of sound signal processing) in the voice control system that implements the levels of the language model is shown in picture 1.



In voice control systems, biopotentials generated by the brain are transformed into acoustical vibrations in human speech, which are transmitted to the control object, where they are converted into electrical signals, then processed by filtering and amplification, analysed and enforced in the form of command, control signals.

Signal analysis

The initial frontiers of wave-technological processes are assumed in order to consider signals, as well as wave-technological processes of noise suppression, which include the following processing frontiers and consideration of information in order to effectively suppress signals in the presence of specific measurement circumstances:

1) preparatory processing as well as the study of information;

2) research, systematisation, clustering, segmentation and selection of websites for the purpose of processing information;

3) mental research for the purpose of adaptive selection of processing methods and characteristics;

4) noise absorption;

5) the study is also the interpretation of the processing results.

Preparatory processing is also the study of information. This information often includes a huge number of single as well as numerous damages, for this reason, preparatory processing of information is needed in order to disclose them, eliminate and fill in the deficiencies that have arisen, and correct the damage found. Single and numerous interruptions appear as a result of severe measurement errors. These errors are more correctly detected as a whole and are also excluded from subsequent consideration at the very, very source of information processing.

The presence of the subsequent elimination of noise (the presence of the creation of limits) of the terrain depicted in the base of approximations is not foreseen in any way, so they do not include noise in any way. The presence of the application of the interpolation scheme of filling should take into account acquired correlations. Research, systematisation, clustering and selection of websites for the purpose of processing information. At this stage, the following actions are performed:

1) selection of structures in the base of consideration of errors, which consists in the selection of errors in the base of the reconstruction of waves using the popular distribution force and the systematisation of measurement zones in the base of subsequent measurements. transient significance of the usual difference in error, asymmetry or the boundary of the noise distribution; research into heteroscedasticity is also uniformity, independence is also correlation;

2) selection of graphs by the characteristics of derivatives supported on the wave transformation;

3) compartmentalisation of sites according to sharp changes in data determined by the analysis of wave coefficients.

Some of Kazakh letters like ң“, ғ, ү, ұ, қ, і, ө” are difficult to process and problems with the pronunciation of speech when working out the sound. It is difficult to remove noise and processing does not always preserve sound correctly. Because some frequencies are like noise and rub the sound quality.

Signal processing methods

Mental study for the purpose of adaptive selection of methods and processing characteristics. The stage of mental consideration is of great importance, which makes it possible to choose: a more optimal method of noise reduction, the number of degrees, the length of the filter, the best basis, as well as the choice of different characteristics for the purpose of specific methods (for example, the optimal length of the block in order to shorten the turbulent coefficients for the presence of processing structures).

Conscious research contains the main subgroups of algorithms. The main subgroup of algorithms includes methods of adaptive selection of a signal-dependent basis in the base of criteria characterising the price of signal approximation in the established basis. After this, minimisation or maximisation is performed (due to the chosen value, polyadelphic. letter, irregularities or concavity of the nature of the price function), ensuring the selection of the optimal basis.

The selection of the optimal basis is performed according to the following aspects: disorder in the base of Schutz coefficients and Gini, Emlin aspect, FL strength, Atkinson aspect. The 2nd category consists of independence (Box-Pierce, Lee-McLeod, Lung-Box, Lagrange multipliers) and normality (d\'Agostinho-Pearson, Yarku-Bera, Gary, Shapiro-Wilks, Shapiro-Frank, Lilly-force). The best basis corresponds to more independent parts or circumstances related to the usual ones.

Removing noise

In considering options when working with significant information, the form of a difficult dynamic move is, as a rule, unknown. In addition to the fact that this phenomenon takes place, a special stage of variability should be noted, in that case, increased attention to the increased risk of hum. As a result of flawless implementation, for the purpose of noise reduction, a wavelet technique was made based on the application of basis functions, having a form with a small-sized stand, as well as a sense of the boundary selection scheme.

The research stage is found in itself a large number of operations, the restoration of the venous bed: four-track, the determination of the marginal result, the selection of the case and the detection of the filter, it requires the reconstruction of the disturbance, various observation restrictions, and the calculation of wave coefficients.

Before rebuilding the driving signal, it is recommended to use a flow meter to achieve results in order to improve the noise reduction results at the turn. The change in skewed results in the original results with a broken outcome at the source of the signal due to a significant MNDVP, as well as at the end of the signal due to the opposite MNDVP. Extreme results are offset by the combination of left and right effects with significant standards that arise from filter detection as well as the number of degrees.

Increasing the probability of obtaining an extrapolation line with a total or other-or parametric (for example, polynomial) modification, regarded at the boundary with a suitable texture as well as an estimation system, with the implementation of an arbitrary classical period (invariant or classical periodic) simulation. After expiration of the exception, additional pointers should be removed.

The process of reducing the wave coefficients consists in preserving estimates of the wave coefficients, expressing the signal, assisting the operator, removing the noise, but retaining part of the signal. There are 2 applications of noise elimination: export and Bayesian. Up to the last period, the coverage covers methods of straight-line demonisation, but non-linear territories are present and difficult.

In the 1990s, Donoho and Johnston used a very common boundary method that was able to select the best non-linear estimator in the base of the right basis. In this regard, various effective algorithms for determining nonlinear thresholds began to appear. Linear wave estimators belong to the class of projection estimators, where projection operators use wave cores. These operators are linear in relation to the data.

Classical methods for determining the nonlinear limit are equivalent to estimating the signal on average through a core adapted to smoothness. In general, the limit value consists of changes in the wave coefficients according to the specified limit type and the reduction rule. Setting the wave coefficients to zero is equal to the local average of the noisy data, which only happens if the received true signal is smooth.

Direct continuous wavelet conversion and wavelet signal analysis

«The term "wavelet"means"small (short) wave" in English. Wavelets are a general name for the category of exact functions of a specific type, they are local according to the period and frequency, but without exception, all functions come out with one main (generating) function shift line, also stretching according to the period axis. The wavelet analyses the period functions, in the process of which the rearrangements are analysed from the point of view of the doubts localised in the period and frequency. As a rule, wavelet transforms are divided into discontinuous (DWT) and continuous (CWT) transformations[В.В.Витязев вейвлет-анализ 2001].

$f(x)$ function for all $R(-\infty,\infty)$defined on the real axis and $L_2(R)$let's assume that it belongs to the space, i.e.,$\int_{-\infty}^{\infty}|f(x)|^2dx<\infty$.

$f(x)\in L_2(R)$ the continuous wavelet transformation of a function is a function of two variables:

$$C(a,b)=\langle f(x)\psi(a,b,x)\rangle = \int_{-\infty}^{\infty} f(x)\psi(a,b,x)dx, \quad (2.1.2)$$
$$a,b\in R, a\neq 0,$$

In here$\psi(a,b,x)\equiv\psi_{ab}(x)$wavelet waves are generated by $\psi(x)\in L_2(R)$ waveform copies are scaled and displaced, and their totality $L_2(R)$ creates a new foundation of the space.$\psi(x)$ the wavelet function has a zero value of the integral, i.e.,$\int_{-\infty}^{\infty}\psi(x)dx=0$and $\psi(\omega)$ – Wavelet transform.

Generating functions can be different functions with a compact carrier-time as well as a place, limited by the period axis, and in this or another facet, have a spectral shape localised in the frequency axis. Similarly to the Fourier transform, $L_2(R)$ the basis of the place must be built with one manufacturing function, the measure of which must be the same 1. In order to overlap the place with the local purpose of the wavelets of the whole period axis, the shift procedure (shift according to the time axis) is applied:

$\psi(b,x)\equiv\psi(x-b)$, where the value of b for NVP is also a continuous value. To cover the entire frequency range of space $L_2(R)$, a time-scale operation of a wavelet is used, in which an independent variable is continuously variable:$\psi(a,x)=|a|^{-1/2}\psi\left(\frac{x}{a}\right)$. If the wavelet time image expands (the value of parameter" a "increases), then its" average frequency " decreases, and the frequency image (frequency localisation) shifts to lower frequencies. Thus, by shifting along an independent variable (x − b), the wavelet moves along the digital axis of an arbitrary signal, and the scale variable "a" (at a fixed point of the time Axis (x − b)) has the opportunity to "see" the frequency spectrum of the signal at a certain interval around that point.Using these operations, the wavelet basis of a functional space is formed by scale transformations and generative wavelet displacement $\psi(x)\in L_2(R)$:

$$\psi(a, x) = |a|^{-\frac{1}{2}}\psi\left(\frac{(x-b)}{a}\right); a, b \in R; a \neq 0. (2.1.3)$$

it is easy to check whether the wavelet norm $\psi(a, b, x)$ is equal to the norm ψ(x), which provides the normalisation coefficient $|a|^{-1/2}$.

When the generating wavelet ψ (x) is normalised to 1, the entire wavelet group is also normalised. If the orthogonal requirement of additional functions is met, then the functions $\psi(a, b, x)$ will be the orthonormal basis of the space $L_2(R)$. In picture-8 shows the expressions that define the most commonly used wavelets.

Wavelet name	Wavelet definition
wavelet-HAAR	$\psi(x)=\begin{cases} 1, & 0 \le x \le 1/2; \\ -1, & 1/2 \le x \le 1; \\ 0, & x<0,\ x \ge 1 \end{cases}$
wavelet-FHAT ("French hat")	$\psi(x)=\begin{cases} 1, & \lvert x\rvert \le 1/3; \\ -1/2, & 1/3 \le \lvert x\rvert \le 1; \\ 0, & \lvert x\rvert > 1 \end{cases}$
wavelet-MHAT ("Mexican hat")	$\psi(x)=\left(1-x^2\right)\exp\left(-\frac{x^2}{2}\right)$

Picture-2. Most commonly used wavelets

Let's go back to the continuous transformation of wavelet (2.1.2). The function $C(a, b)$ carries information about the frequency-time structure of the function f(x).

In fact, by setting small values of the scale but, we acquire information about the specifics of the signal f(x), which is insignificant in terms of size, but large values but make it possible to establish a "global" information about the signal (this is divided according to the whole signal). This model can be characterised by an uncertainty condition with the aim of the wavelet functions, but directly: an excellent solution according to frequency (small Fourier transform width of the wavelet function) leads to a poor resolution according to the period (huge width of the wavelet function) also opposite.

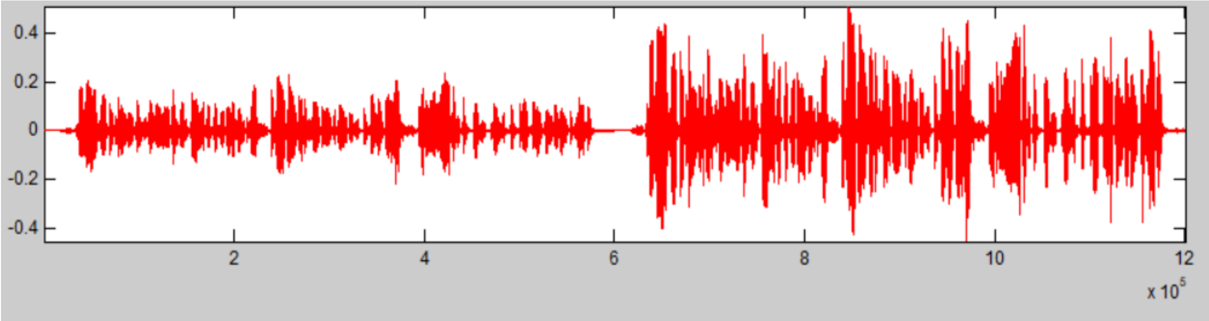
Concentrate interest, then that the smallest volume of the window of the figure of the signal does not have to be higher than the highest frequency of the harmonic in order to reflect the highest frequency of the signal. In case the sign has spectral elements corresponding to the current value of a, then the antiderivative of the wavelet derivative with a signal in the interval containing the given spectral part, provides a relatively large significance. Otherwise, I can observe in picture 8 that the product is less than or exactly also zero. With an increase in the scale of the signal (width of the window), the change selects all the lowest frequencies without exception.

Graphs:

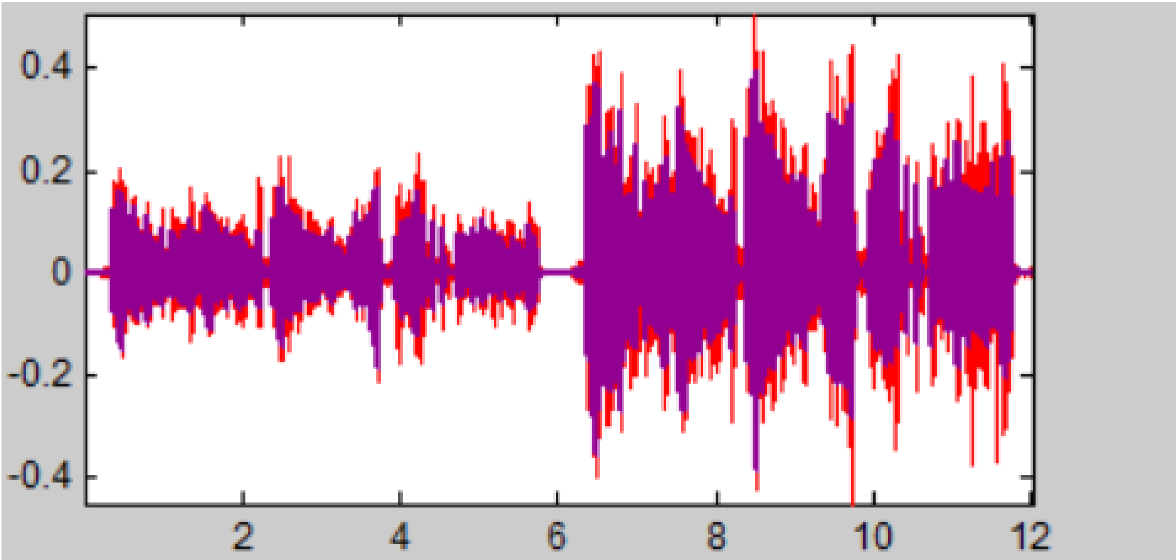
The microprogram mechanism for the implementation of discontinuous wave transformations involves a library of functions for the purpose of accurate prediction of research (MATLAB) and computer algebra concepts (MATHCAD, Mathematica, LabView with National Instruments). These devices are usually targeted at certain DWT implementations designed to perform highly specialised practical issues, such as effective reduction, signal attenuation, obstacle avoidance, steganography, etc. The main distinguishing features of such add-ons include: There are no basic restrictions.

The volume of the information review window is also the computational difficulty of the DWT method, a huge set of DWT functions, the probability of synthesising the base function around a certain problem, the lack of basic restrictions according to transient costs, high accuracy. In the MATLAB concept of accurate prediction, in an instructive manner, in order to extract the initial wave data.

The graphs show that the study of sound and connection with the waves they are correct. There are shown the difference between audio, and Echo, and prove that they are multi-level.



Picture-3. Translating the audio file into a signal



Picture-4. Cleaning audio from excess noise

CONCLUSION

We went deeper into the philosophy of sound and fully discovered the connection with the current wavelet transformation in modern times . We have fully identified the differences between sound, signal and Echo, and have clearly demonstrated them using a popular, convenient programming language that is now multi-level. The unique properties of the wavelet Transform and fast wavelet Transform algorithms have made them a powerful and effective tool for analysing and synthesising signals and images of various types. The number of publications is steadily growing and cannot be counted due to the large number of practical applications. The range of questions on the wavelet Transforms app is so wide that its description Requires a multi-volume publication. Thanks to the use of wavelet transformation, good results have been achieved in many areas of science, technology, medicine and economics. First of all, wave conversion is used in the analysis of non-stationary signals, where it is more effective than the traditional Fourier transform and is used in: radio engineering and radio communications; physics; laser technology; seismic and hydroacoustics; medicine and biology; hydrodynamics, meteorology, aviation, etc.

Promising results for the use of wavelet Transforms in digital communications, in particular transmultiplexers, systems with broadband signals, are known. The use of wavelet packages for anonymous communication and is promising in more accessible systems.

Wavelet Transform is widely used to delete and compress signals, images, and Multimedia Information. Wavelets are used to detect a signal against a noise background and recognise it. For example, the US Navy used the wavelet transform tool to detect and recognise submarines. Many researchers abroad and in Kazakhstan use wavelet transformation to identify and recognise local features of an electrocardiosignal.

Since the wavelet transformation is easy to generalise to sets of any size, it can be used to analyse and recognise multidimensional patterns. Wavelet Transform is increasingly used in time series Research and forecasting. In fact, a time series is any function (or signal) represented at separate points in time. In addition to the considered discrete signals, the time sequence can be a sequence of readings of ambient temperature or pressure, the value of shares or the dollar exchange rate at certain points in time, internet traffic, and so on.

In the literature, the paper considers the prospects for using wavelet transformations to analyse time series. Successful attempts to use Wavelet Transforms to predict events such as weather forecasts, earthquakes, tsunamis and other natural disasters, various engine failures (both occurred in the United States and here) are well known.

References

1. Voskoboynikov YE, Belyavtsev VG Nonlinear algorithms for filteringvector signals / Autometry. - 1999. - No 5. - 97–106 p.

2. Voskoboynikov Yu.E., Mukhina IN Adjusting the image recovery algorithm by specifying local noise / signal ratios // Autometry. - 1999. - No 4. - C. 71–83.

3. https://plato.stanford.edu/entries/sounds/

4. Dremin IL Wavelets and their application / IL Dremin, OA Ivanov, VA Nechitaylo // Success of physical sciences. - 2001. - Vol.171, No5. - P.465-501.

5. https://www.bostonreview.net/articles/dmitri-tymoczko-sound-philosophy/