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Automatic initial segmentation of speech signal based on symmetric matrix of distances

D.Peleshko, Y.Pelekh, M.Rashkevych, Y.Ivanov, I.Verbenko

Professor, Dmytro Peleshko, Department of Publishing Information Technologies, Institute of Computer Science and Information Technologies, Lviv Polytechnic National University, Lviv, Ukraine

Ph.D. student, Yuriy Pelekh, Department of Publishing Information Technologies, Institute of Computer Science and Information Technologies, Lviv Polytechnic National University, Lviv, Ukraine

PhD, Senior Lecturer, Mariya Rashkevych, Department of Publishing Information Technologies, Institute of Computer Science and Information Technologies, Lviv Polytechnic National University, Lviv, Ukraine

Ph.D. student, Yuriy Ivanov, Department of Publishing Information Technologies, Institute of Computer Science and Information Technologies, Lviv Polytechnic National University, Lviv, Ukraine

Ph.D. student, Iryna Verbenko, Department of Publishing Information Technologies, Institute of Computer Science and Information Technologies, Lviv Polytechnic National University, Lviv, Ukraine

ABSTRACT

The most common issue of a speech signals analisys and artificial intelligence systems development is determining of temporal and frequency charactristics. That's because any undetermined signal is defined as a nonlinear object. But it is always possible to select time interval from the signals with a given discretization period. Such an interval is called quasistationary interval.

The quasistationary interval in combination with the speech signal quality characteristics can be used to build a parametric model of the speech signal. As a result, such a model will be very helpful for solving different issues of artificial intelligence development processes.

In this paper the method of an initial segmentation of the speech signal via quasistationary intervals by threshold deviation of algebraic specifications of elementary areas is presented. The latter method of automatic initial segmentation is based on pseudorotation of symmetric matrix of distances and evaluation criteria. The problem and process of selection the quasistationary intervals are described in the section 1.

Indexing terms/Keywords

Speech signal; Initial segmentation; Symmetric matrix of distances; Pseudo inverse matrix; Moore-Penrose algorithm; Singular decomposition; Euclidean; Chebyshev and cosine metrics; Measure of similarity.

Academic Discipline And Sub-Disciplines

Digital signal processing.

SUBJECT CLASSIFICATION

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TYPE (METHOD/APPROACH)

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INTRODUCTION AND ANALYSIS OF THE LITERATURE

Nowadays, the speech signals processing takes important place at the present development stage of methods and tools for building artificial intelligence systems. Real-time support is a common requirement that is proffered to methods of analysis and synthesis of speech information. As a result, methods of the direction of linear prediction [5, 7] during the simulation of speech signals become significantly interesting and widely used. The main idea of these methods is to represent the signal into a form of autoregression.

There are many advantages of methods of this direction. Forecasting properties are one of the main advantages among other that should be highlighted. It is based on the capabilities to predict the next value with certain accuracy by some linear combination of the normalized amplitude values of the speech signal. Independency of processing methods from the signal processing availability in general is considerable in the described simulating procedure of the speech signal. This is exactly what makes it possible to use linear prediction methods in real-time systems procedures.

Quite simple conversion to speech creating model and relatively low computational complexity of determining prediction parameters are other advantages because the latter is based on linear algebraic procedures.

Spectral signal estimation on the quasistationary interval, the speech synthesis construction based on recursive digital filters and recognition systems based on the spectrum simulation [4] are common use of methods of linear prediction. Such a wide range of use has led to plenty linear prediction variations, in particular covariance, autocorrelation, maximum plausibility, scalar product and etc.

However, the main drawback of linear prediction methods that is often highlighted in the literature, for instance [12], is inadequate accuracy or difficulty in determining the threshold coefficients of signal simulating process on areas with low energy.

THE PURPOSE OF THE RESEARCH

Determining of temporal and frequency characteristics is one of the most common problems that arise during analysis of the speech signals and development of modern artificial intelligence systems because any undetermined signal is a nonlinear object. However, a time interval can always be selected in such a signal with a given discretization period where the values of these characteristics vary within an insignificant deviation. Then the characteristics of the signal on this interval are considered to be constants. The latter interval is called as quasistationary interval.

From time to time such periods of the speech signal are called as signal's frame [5] and the process of their selection is an initial segmentation.

If the speech signal quality characteristics are defined, then quite accurate parametric model of the speech signal can be built on the quasistationary interval. This parametric model can be successfully used for solving a wide range of problems that appear during development of artificial intelligence.

Therefore, the main objective of this work is to develop a method of automatic initial segmentation of the speech signal by quasistationary intervals that is based on pseudo rotation of symmetric matrix of distances and evaluation criteria for automatic determination of threshold values.

1. PROBLEM OF THE QUASISTATIONARY AREAS SELECTION

Let the time period is $\tau = [0; T]$, $T \in R^{1,+}$. Within the τ interval the system of open sets is

$$\Gamma = \{T_i\}_{i=1,2,\ldots},\ T_i = [t \mid t_{i-1} \le t < t_i,\ t_{i-1}, t_i \in \tau], \tag{1}$$

with diameter $|T_i| = \sup\limits_{t_a, t_b \in T_i} d(t_a, t_b)$, where d is the space metric R^1, the topology of Γ is defined. As the interval τ is a closed limited set then a disjunctive finite covering $\chi = \{T_i \mid i = 1 \ldots n\}$ with trivial intersection is always exist within the topology Γ. As a result, the following takes place

$$\forall i, j \in [1; n]:\ T_i, T_j \in \chi \wedge i \ne j \to T_i \cap T_j = \varnothing; \tag{2}$$

$$\tau = \bigcup_{i=1}^{n} T_i\,. \tag{3}$$

Let assume that all items of χ coverage have the same diameter that is equal to l: $\forall i \in [1, n] : |T_i| = l$. This determines χ as l -coverage of τ interval.

Let define $x(t)$ speech signal on the τ interval as a continuous surjective reflection

$$x: \tau \to R^1. \tag{4}$$

Since the $x(t)$ is a continuous reflection then not necessarily disjunctive χ coverage of $x(t)$ function domain is obtained.

$$\eta = \{X_i \mid i = 1 \ldots n\}, \tag{5}$$

where $X_i = \{x(t) | t \in T_i\} = \{x(t) | t \in T_i\}$ is an element of η coverage.

Quasistationary intervals Y_i of $x(t)$ speech signal are the union of disjunctive elements of η coverage based on specified membership criteria

$$Y_i = \bigcup_{j=a_i}^{b_i} X_j \,. \tag{6}$$

Where $X_j \in \eta$ is a sequential element of η coverage that meet the membership criteria; a_i, b_i are indexes within $\{X_i\}$ sequence that define the subset of elements which create Y_i in union. The difference $m_i = b_i - a_i$ between the indexes defines the number of X_i basic intervals that determines the power of Y_i: $|Y_i| = m_i l$ quasistationary interval.

The $\{X_i\}$ set creates a new η'disjunctive coverage of $x(t)$ signal space by (5) and (6). As a result, the following equality takes place

$$x(t) = \bigcup_{i=1}^{m} Y_i \,, \tag{7}$$

where $m \le n$ is a number of Y_i quasistationary intervals.

The dimension of $x(t)$ signal is the number of points within T discretization period. T defines from the (6) and (7) as

$$\dim\,(x(t)) = nl = l \sum_{i=1}^{m} m_i \,. \tag{8}$$

The problem of selection quasistationary intervals of $x(t)$ signal can be considered as a problem of constructing an conversion operator of f: $\chi \to \chi'$ or F: $\eta \to \eta'$ topologies using (5), (7) and (8).

2. CONSTRUCTION OF AGGREGATE MATRIX OF DIVERGENCES

In contrast with [18] the speech signal is not normalized nor shifted to positive area during the creation of the operator. Amplitude values of X_i element of the speech signal which correspond to the T_i basic interval are used to construct ∇_i: $T_i \to X_i$ matrix operator without any additional transformations.

Operator of ∇_i: $T_i \to X_i$ transformation of l-dimensional vectors that are defined on the T_i interval corresponds to X_i element of η coverage and constructs as aggregate symmetric matrix of distances [6]:

$$\forall i \in [1;n]: \nabla_i = \begin{pmatrix} \delta_{i,(1,1)} & \dots & \delta_{i,(1,l)} \\ \dots & \dots & \dots \\ \delta_{i,(l,1)} & \dots & \delta_{i,(l,l)} \end{pmatrix}, \ \delta_{i,(z,k)} = \left| x_{i,k} - x_{i,z} \right|, z, k = 1..l \,, \tag{9}$$

where i - index of X_i, $x_{i,j} = x(t_j)$, $t_j \in T_i$ basic interval. Matrix dimension (10) is equal to: dim $\nabla_i = l \times l$. The following matrix (10) can be obtained from matrix (9) in case of Euclidean distance.

$$\forall i \in [1;n]: \nabla_i = \begin{pmatrix} 0 & \Delta \dfrac{dx_{1,1}}{dt} & \dots & (l-1)\Delta \dfrac{dx_{1,1}}{dt} \\ \Delta \dfrac{dx_{1,1}}{dt} & 0 & \dots & (l-2)\Delta \dfrac{dx_{1,2}}{dt} \\ \dots & \dots & \dots & \dots \\ (l-1)\Delta \dfrac{dx_{1,1}}{dt} & (l-2)\Delta \dfrac{dx_{1,2}}{dt} & \dots & 0 \end{pmatrix}. \tag{10}$$

The following (11) equation is considered for solving the problem of vectors selection that perform as properties of speech signal basic interval for each X_i element of η coverage.

$$\nabla_i g_i = x_i \,, \tag{11}$$

where $x_i = \left\{ x\left(t_p\right) \in X_i \mid p = 1..l \right\}$ – l is a l-dimensional vector of amplitude values of $x(t)$ speech signal within the interval of X_i; $g_i = (g_{i,1}, \dots, g_{i,l})$ unknown vectors.

The g_i vector is defined as (12) using the (11).

$$g_i = \nabla_i^{-1} x_i \,. \tag{12}$$

The inverse ∇_i^{-1} matrix does not exist because the ∇_i matrix is degenerated ($\det(\nabla_i) = 0$) [3]. Therefore, a method of determining y_i vector [1, 17] is proposed to solve (11) problem using the theorem of residual minimizing $||x_i - \nabla_i g_i||^2$ of the (12) linear system.

$$g_i = \nabla_i^+ x_i + \left(1 - \nabla_i^+ \nabla_i\right) r_i \,, \tag{13}$$

where ∇_i^+ is a generalized inverse Moore-Penrose matrix which is pseudoinverse to ∇_i matrix [1, 17]); $\left(1 - \nabla_i^+ \nabla_i\right)$ is an operator of projection on the kernel of ∇_i operator; r_i is a random vector of l dimension. The first addition from (13) is a pseudoinverse solution and the second is a solution for $\nabla_i g_i = 0$ homogeneous system. A way to determine a vector of characteristics of i-th element of χ coverage that is presented in (13) is possible because $\nabla_i^+ \nabla_i$ matrix is not degenerated according to [1].

The ∇_i^+ Moore-Penrose matrix is determined by singular decomposition of ∇_i matrix in the following way [17]:

$$\nabla_i^+ = V_i \Sigma_i^+ U_i^T \,, \tag{14}$$

where U_i, V_i are unitary matrixes of $l\times l$ order of singular decomposition of the ∇_i; Σ_i^+ matrix of $l\times l$ order which is pseudoinverse to Σ_i diagonal matrix of singular decomposition [3] and ∇_i matrix. The Σ_i^+ matrix is obtained from Σ_i by replacing all nonzero singular values $\sigma_{i,q}$ ($\sigma_{i,1} \ge \sigma_{i,2} \ge \ldots \ge \sigma_{i,l} \ge 0$) to corresponding inverse $1/\sigma_{i,q}$. because Σ_i matrix is also degenerated.

A random vector is determined by the following residual: $r_i^{j+1} = ||x_i - \nabla_i g_i^j||_l$, here $||\cdot||_l$, where l is a norm [1] in the iteration process of finding by g_i^{j+1} (14).

3. SPEECH SIGNAL SEGMENTATION BY THE QUASISTATIONARY INTERVALS

Let the matrix G' is formed with $g'_i = (g'_{i,1}, \ldots, g'_{i,l})$ vectors which were derived from (13)

$$G' = \begin{pmatrix} g'_1 \\ g'_2 \\ \ldots \\ g'_n \end{pmatrix} = \begin{pmatrix} g'_{1,1} & g'_{1,2} & \ldots & g'_{1,l} \\ g'_{2,1} & g'_{2,2} & \ldots & g'_{2,l} \\ \ldots & \ldots & \ldots & \ldots \\ g'_{n,1} & g'_{n,2} & \ldots & g'_{n,l} \end{pmatrix}. \tag{15}$$

Matrix should be normalized by the maximum element: $G = G'/\max(G')$. As a result, we obtain the matrix $G = < g_i >_{i=1..n}$, whose elements are defined as: $g_i = g'_i / \max(G')$.

Let introduce the metric distance of two elements X_i and X_j via Chebyshev distance of corresponding vectors g_i and g_j in the vector space $\{g_i\}$:

$$\mu\left(X_i, X_j\right) = \max_{k\in[1;l]} \left\{\left|g_{i,k} - g_{j,k}\right|\right\}, \tag{16}$$

With the help of metric (16) it is possible to perform an initial segmentation by determining the condition of belonging elementary area X_j to the quasistationar Y'_i

$$X_j \in Y'_i \Leftrightarrow \forall z \in \left[a_i; b_i\right] : \mu\left(X_j, X_z\right) \le \varepsilon \,, \tag{17}$$

where $\varepsilon \in \mathbf{R}^{1,+}$– threshold value, which is an error of referring to the elementary area X_j to the quasistationar Y'_i

The term initial segmentation is undertaken in the context of the fact that in some cases the use of separate metrics (e.g. correlation metric) may need additional procedure. The point of this procedure is to combine the adjacent segments Y'_i and Y'_j if the distance between their starts is equal to l. As a result we will get a set of quasi-stationary areas of $\{Y_i\}$. Obviously, the need for additional procedure should be determined separately for each metric. If such a metric is selected, that there is no need in additional combining so $Y_i = Y_i'$ and additional procedure actually performs as zero additive operator. Its use does not affect the segmentation results, but only slows down the speed of the computational process.

Problem (17) has a solution for a given pre-threshold value ε. However, in case of determination of the criterial feature K the selection procedure of ε can be automated by solving some extremal problem.

Realization of described approach can become a way to determine the value of automatedε, in base of which the criterion K is the optimization of deviation of the results of segmentation Y which are obtained by developed method from results $Y_{ет}$ got by 'reference' method.

$$K : \|Y - Y_{ет}\| \to \mathrm{opt}(\varepsilon). \quad (18)$$

During the practical implementation of the described method the way of automated determination of threshold value ε was based on maximization of the value of the selected coefficient of similarity (similarity measure) [10]. In general case, it could be selected a few coefficients instead of just one and some integral parameter could be determined with their help. For this purpose, from a general formula of continuum similarity measures of Syomkin [11]

$$K_{\tau,\iota}(Y, Y_{ет}) = \left(\frac{K_{\tau,\iota}(Y \mid Y_{ет}) + K_{\tau,\iota}(Y_{ет} \mid Y)}{2} \right)^{\frac{1}{\iota}}; -1 < \tau < \infty, -\infty < \iota < \infty. \quad (19)$$

when τ = 0 (a proximity measure of adjacent objects according to the general formula of average values of Kolmogorov [10, 11]) and ι = +∞, 1, 0, 1, -∞ is selected the set of the most used measures of similarity which are ordered by ι, in particular of Kulchynskiy $K_{0,1}$[15], Ohai $K_{0,0}$[16], Sørensen $K_{0,-1}$[20], Braun-Blanquet $K_{0,-\infty}$[13], Shymkevycha-Simpson $K_{0,+\infty}$[19]. Average value can be determined from the elements of the set of measures

$$K_{\Sigma} = \frac{K_{0,+\infty} + K_{0,1} + K_{0,0} + K_{0,-1} + K_{0,-\infty}}{5}. \quad (20)$$

Then, according to (18), to calculate value ε the problem of search mechanism is solved:

$$\|Y - Y_{ет}\| \to \max_{0<\varepsilon\le 1} K_{\Sigma}. \quad (21)$$

4. THE EXPERIMENTAL RESULTS AND CONCLUSION

An algorithm for the speech signal segmentation was developed in order to verify the theoretical researches based on the proposed method. Software implementation of the algorithm was tested using the initial segmentation of the word "mysha" which wave representation is shown in Figure 1 and 2. The following speech signal characteristics are given: the word duration is 1.03 seconds, sample rate is 11025 Hz, and length of basic intervals is l = 120 samples.

Chebyshev metric (fig. 1 and Table 1) and cosine metric (fig. 2 and Table 1) are used to determine the distance during practical implementations. 0.15 and 0.01 values are obtained in case of Chebyshev and cosine metrics correspondingly as results of the procedure of automatically determining the threshold ε.

Results of the same signal segmentation by DELCO algorithm [7] with 1.6 threshold value in the same elementary area size are selected as an evaluation criteria for automatically determining of threshold deviation ε. The numerical values of the segmentation results by DELCO method are shown in the table 1. In this case the extreme problem was solved in concerning to the integral index (20). Values of all parameters from (20) are shown on the fig. 3. In addition, Yurtsev $K_{0,+\infty}$ [9] and Zhakar $K_{1,-1}$ [14] measures also calculated. They were not included to the integral indicator (2) because Yurtsev measure is dual to Braun-Blanquet measure and Zhakar overlapping measure is equivalent to Sørensen measure. Values of the same coefficients in the case of segmentation based on asymmetric matrix of convergence are given on the fig. 2 with the aim to improve the perception of values of similarity coefficients. Segmentation results by this method are given in the table 1.

The obtained results show that Kulchinsky, Ohai, Sørensen and Braun-Blanquet measures approximately equally represent the similarity nature of the segmentation results according to DELCO method. Chebyshev metric provides better results of proximity measure according to the segmentation results by DELCO method.

Values of all defined coefficients show that segmentation results based on the developed method by Chebyshev metric are most similar to the DELCO results. Herewith the deviation of values is not sufficiently large even when taking into account the value of Simpson's measure. This demonstrates that the developed method based on Chebyshev distance is sufficiently resistant to proximity measures and allows use only the calculation of one of the Kulchinsky, Ohai, Sørensen, or Braun-Blanquet or Simpson coefficients instead of using the integral index (20) in contrast with the case of using matrix convergence.

Measures of similarity values are smaller by values, but deviation is also small in the case of cosine metric. Therefore, it's possible to ascertain the stability of the segmentation results and automatic determination of threshold deviation to selection of proximity measure during the use of cosine metrics.

Developed method with the using of Chebyshev distance gives a possibility to select unique speech units according to the obtained results are shown in Fig. 1-2. At the same time cosine metric allows making of the initial segmentation only. This method is more sensitive within the noisy parts of the speech signal than the DELCO method.

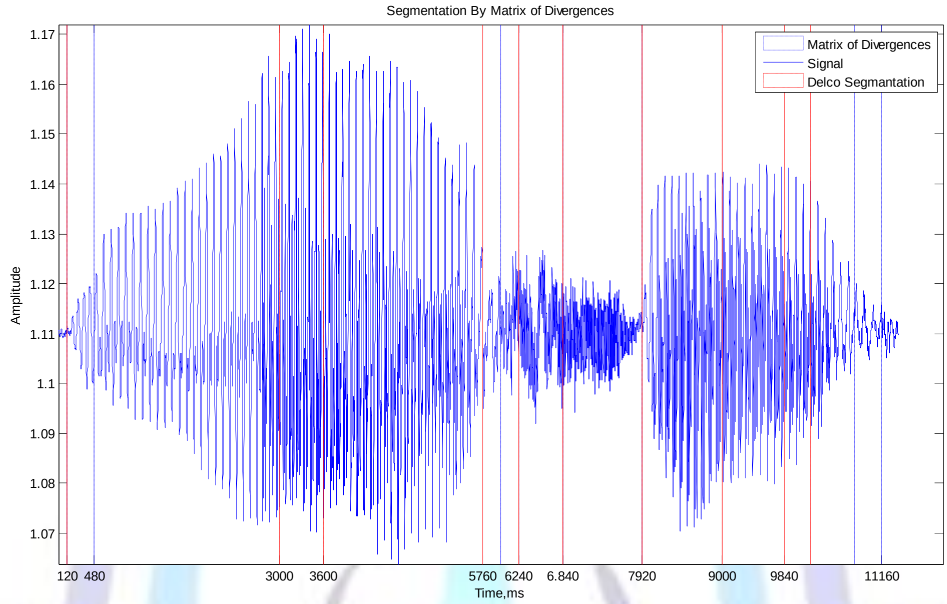

Fig 1: The results of speech signal segmentation (the word "mysha") by DELCO algorithm and the developed method based on Chebyshev metrics

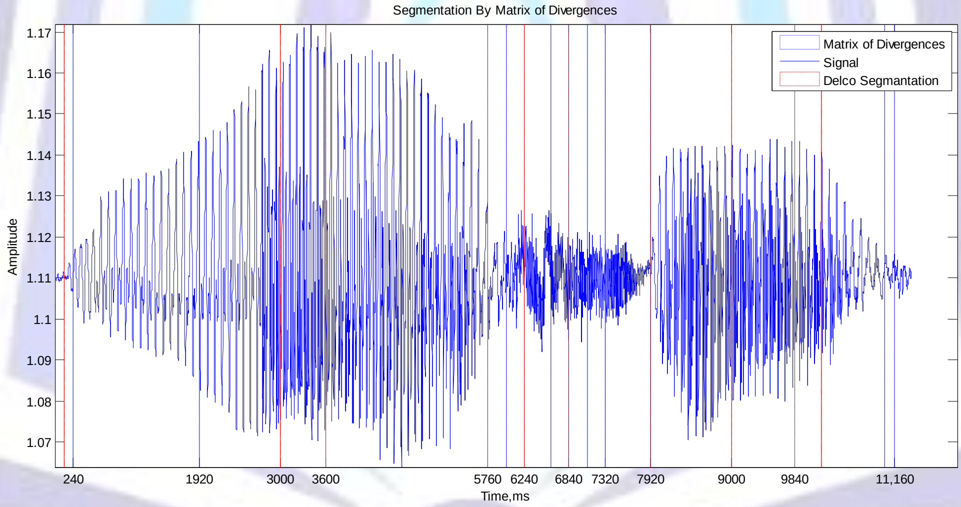

Fig 2: The results of speech signal segmentation (the word "mysha") by DELCO algorithm and the developed method based on using cosine metric

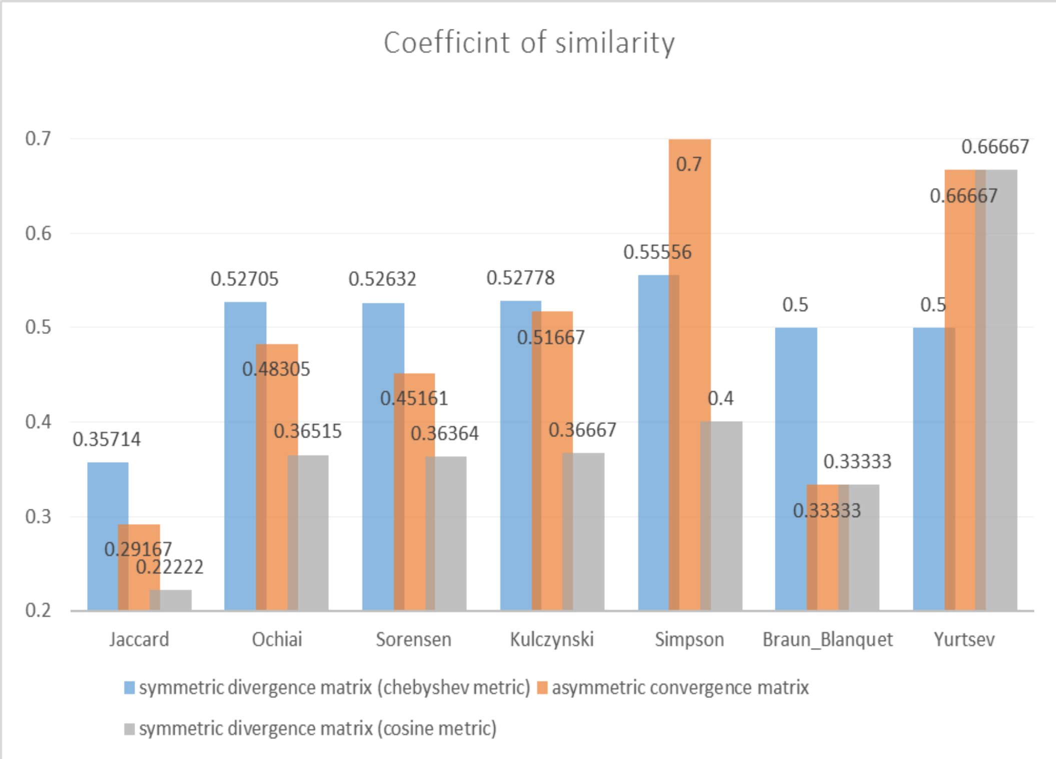

Fig 3: Coefficients of the segmentation results similarity by DELCO algorithm and obtained method

Table 1. The numerical values of the segmentation results by DELCO method and segmentation results based on asymmetric matrix of convergence

Interval (ms:ms)	Segment by index	Count of elements	Interval (ms:ms)	Segment by index	Count of elements
Segmentation By Matrix of Divergences			**Segmentation by Asymmetric Convergence Matrix**		
#Euclidean metric					
(120:480)	[1;4]	3	(480:1200)	[4;10]	6
(480:5760)	[4;48]	44	(1200:1680)	[10;14]	4
(5760:6000)	[48;50]	2	(1680:2160)	[14;18]	4
(6000:6240)	[50;52]	2	(2160:2760)	[18;23]	5
(6240:6840)	[52;57]	5	(2760:3000)	[23;25]	2
(6840:7920)	[57;66]	9	(3000:3600)	[25;30]	5
(7920:10800)	[66;90]	24	(3600:3960)	[30;33]	3
(10800:11160)	[90;93]	3	(3960:4680)	[33;39]	6
#Cosine metric			(4680:5160)	[39;43]	4
(240:1920)	[2;16]	14	(5160:5760)	[43;48]	5
(1920:5760)	[16;48]	32	(5760:6240)	[48;52]	4
(5760:6000)	[48;50]	2	(6240:6600)	[52;55]	3
(6000:6240)	[50;52]	2	(6600:7200)	[55;60]	5
(6240:6600)	[52;55]	3	(7200:7920)	[60;66]	6
(6600:6840)	[55;57]	2	(7920:8400)	[66;70]	4
(6840:7080)	[57;59]	2	(8400:9000)	[70;75]	5
(7080:7320)	[59;61]	2	(9000:9240)	[75;77]	2
(7320:7920)	[61;66]	5	(9240:9840)	[77;82]	5
(7920:11040)	[66;92]	26	(9840:10440)	[82;87]	5
(11040:11160)	[92;93]	1	(10440:11160)	[87;93]	6
Delco Segmentation					
(120:3000)	[1;25]	24	(6240:6840)	[52;57]	5
(3000:3600)	[25;30]	5	(6840:7920)	[57;66]	9
(3600:5760)	[30;48]	18	(7920:9000)	[66;75]	9
(5760:6240)	[48;52]	4	(9000:9840)	[75;82]	7
			(9840:10200)	[82;85]	3

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Authors' Biography with Photo

Dr. Dmytro Peleshko is Professor at Lviv Polytechnic National University, Ukraine. He has published more than 200 papers in international and national scientific issues and journals. His research work based on digital signal processing. In addition, he is an author of monograph by the topic of his doctoral dissertation. He also is a supervisor of four Ph.D. students.

Yuriy Pelekh is a Ph.D. student at Lviv Polytechnic National University, Ukraine. He has B.Sc. degree in Computer Science and M.Sc. degree in Automated Control Systems. His research area is speech signal reconstruction based on singular value decomposition. Currently he has published more than 10 papers in scientific journals and has presented 8 research papers at international conferences. His interests are digital photography and software engineering.

Mariya Rashkevych is a Senior Teacher in the Department of Computer Science at the Lviv Polytechnic National University. She received her Ph.D. in Technical Sciences from Lviv Polytechnic National University. Her research interests lie in the general field of digital signal processing. Dr. Rashkevych.s research has been published in various international journals and conferences. Her thesis work focused on type systems for speech signal reconstruction and value decomposition. Dr. Rashkevych.s interests lie with the design and implementation of programming languages, particularly compiler technology, concurrency and parallelism, type systems, and program semantics.

Yuriy Ivanov is a Ph.D. student at Lviv Polytechnic National University, Ukraine. He has a M.Sc. degree in Automated Control Systems, B.Sc and M.Sc. degree in electrical machines and apparatus. His research work based on tracking of dynamical objects in videostreams of territory-distributed videosurveillance systems. Currently he has published several papers in scientific journals and has presented over 10 research papers at international conferences. His interests are videostreams processing, videoanalythics and software engineering.

Iryna Verbenko is a Ph.D. student at Lviv Polytechnic National University, Ukraine. She has M.Sc. degree in Printing and publishing business. Her research area is fuzzy logic systems. She has published more than 10 papers in scientific journals and has participated in various international and national conferences.