VILNIUS GEDIMINAS TECHNICAL
UNIVERSITY

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RESTORATION OF NOISE DISTORTED ACCENTS AUDIO RECORDS

Summary of Doctoral Dissertation

Technological Science, Electrical and Electronic Engineering (01T)

Vilnius "Technika" 1999

The work was carried out at the Vilnius Gediminas Technical University.

Right for doctoral studies granted by resolution No. 457 of Lithuanian Republic Government at 1998 04 14.

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One can get acquainted with the dissertation in library's scientists reading room of the Vilnius Gediminas Technical University (Saulėtekio al. 11, Vilnius, Lithuania).

RESTORATION OF NOISE DISTORTED ACCENTS AUDIO RECORDS

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VILNIAUS GEDIMINO TECHNIKOS
UNIVERSITETAS

ŠARŪNAS PAULIKAS

TRIUKŠMO SUGADINTŲ PRIEGAIDŽIŲ GARSO ĮRAŠŲ RESTAURAVIMAS

Daktaro disertacijos santrauka

Technologijos mokslai, elektros ir elektronikos inžinerija (01T)

Vilnius “Technika” 1999

Darbas atliktas 1994-1999 metais Vilniaus Gedimino technikos universitete.

Doktorantūros teisė suteikta 1998 04 14 Lietuvos Respublikos Vyriausybės nutarimu Nr. 457.

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Disertacijos santrauka išsiųsta 1999 m. liepos mėn. 27 d.
Su disertacija galima susipažinti Vilniaus Gedimino technikos universiteto bibliotekos mokslininkų skaitykloje (Saulėtekio al. 11, Vilnius, Lietuva).

TRIUKŠMO SUGADINTŲ PRIEGAIDŽIŲ GARSO ĮRAŠŲ RESTAURAVIMAS

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THE GENERAL CHARACTERIZATION

The Problem.

In our country is a lot of old Lithuanian folk speech record archive. Part of them belongs to vanished dialects. These records have been disappearing due to time influence, old recording technique and record carrier imperfection. To preserve this cultural heritage, the old records must be restored and transferred to save digital media.

Modern techniques of speech signals restoration employ Linear Prediction [1], Adaptive filtering [2, 3], Hidden Markov Models [4], Artificial Neural Networks [5], various Bayesian techniques [6]. However, there are cases when all methods fail. It happens, when sufficiently long segments of speech signals constituting essential information are lost. In order to restore these segments additional *a priori* information must be available.

Lots of existing languages and dialects introduce certain differences in speech signals [7]. One of the main differences in Lithuanian language and its dialects is stressing [8 - 10]. Lithuanian language has free stress that does not belong to the specific syllable The Lithuanian language also differs in accents (raising or falling) of stressed syllables. Other languages with free stress have not this characteristic feature.

Restoring old folk records these prosodic elements as stress and accent must be preserved, because they enable us to distinguish meaning of words, especially, homographs – words which meaning depends on the place of stress or accent type [11].

Relation with Scientific Projects.

Participation in scientific project supported by Lithuanian State Scientific and Study Fund: “Investigation of Lithuanian Folk Dialects Speech Records”.

Objectives.

1. Create methods for filtering of non-stationary acoustic signals with non-linear change of pitch and intensity.
2. Demonstrate suitability of created method for Lithuanian language records with accent restoration.

In order, to accomplish this task:

- the speech signal characteristics, which determines type of accent, were investigated;
- the mathematical accent model was created;
- the algorithm with incorporated accent model and software for restoration of accent in speech signals were created.

Investigation Methods.

Researches in dissertation are based on fundamentals of mathematical modeling, digital speech signal processing, speech signal models, knowledge of experimental phonetics and prosodic.

Scientific Novelty.

The creation of mathematical accent models and constructing of speech signal restoration methods based on these models is scientific novelty of dissertation.

Defending Results.

1. The polynomial mathematical model of accent, that encounters the change in time of pitch and intensity of speech signal.
2. The new speech signals restoration method, that takes into account non-linear changes of pitch and intensity of speech signals.

Practical Benefit.

1. The accent restoration method that uses polynomial accent model with original intensity and pitch characteristics results in 1.7 time smaller mean square restoration error.
2. Restoration of accent can be improved by 1.3 times, when accent model with average intensity and pitch characteristics is used.

Approbation of Dissertation Results.

The main dissertation results was presented in conferences "Elektronika" 1995, 1996, 1997 and 1998 years, 18^{th} International Symposium of Students and Young Scientists at Zielona Gura 1996, Baltic Electronics Conference at Tallinn 1996, 1^{st} International Conference Digital Signal Processing and its Applications at Moscow 1998.

Published Works.

7 scientific publications were published on the dissertation topics: 2 articles, 4 thesis of international conferences and 1 scientific report.

Dissertation Structure and Volume.

Dissertation is written in Lithuanian language. It consists of 98 pages, main text 71 pages. Dissertation has 32 illustrations and 1 table and 6 chapters. Afterwards follows references citations list (86 items) and appendix. Appendix contains MATLAB 5.2 scripts and functions that were used for restoration experiments and CD-ROM with accent restoration examples.

DISERTATION THESIS

First chapter – Introduction. This chapter introduces to problem of restoration of old folk speech records. Presents such specific features of Lithuanian language as stress and accent. Shows importance of these features in distinguishing words – homographs. Further describes accents types and its characteristics. States that accent can be fully described by such speech signal characteristics as duration, intensity and pitch. Shows usefulness of averaged and normalized intensity and pitch in general accent description, because these characteristics depend from specific speaker.

Second chapter – Noise Suppression in Speech Signals. This part devoted to review of existing methods of noise suppression in speech signals. Some presented methods are new other well known but still is under improvement and have receive new attention.

This chapter is divided into two main sections according to character of suppressed noise. First section reviews existing methods of suppressing of background noise. Widely known and used are spectral subtraction method and its modifications. This method takes into account only noise characteristics and ignores signal properties.

Other described noise suppression methods are based on the periodicity or linear model of speech signal.

Second section deals with removal of impulsive noise. The classical methods are based on Median filters. They can remove only short duration impulsive noise. However, at the same time, they also corrupt useful signal.

Advanced methods of impulsive noise removal are based on linear prediction. These methods can restore speech signals affected by long duration impulsive noise. However, the duration of impulsive noise must be shorter than stationary time of speech signals. The best methods are adaptive and manage to model linear change of speech signals in time.

Important problem of noise suppression in speech signals is model non-linear change of speech signal characteristics. This is necessary in restoration of old Lithuanian folk records, especially in homograph case, because in dialects are dominating such features of Lithuanian language as stress an accent. These features are described by non-linear change of speech signal characteristics.

Third chapter – Accent Restoration. In this chapter is described restoration of homograph, i.e. accent type, which is essential in distinguishing these words meaning. In order to accomplish this task the accent model must be created.

Lets consider two words spoken by male speaker of ~ 400 ms duration, recorded with sampling rate of 44.1 kHz (Fig. 1). They are typical representatives of homographs. They are spelled identically, but posses different meaning – $k\tilde{a}ltas$ -"chisel" for Fig. 1 (a) and $kal'tas$ -"guilty" for Fig. 1 (b). Main difference between these sounds is in their accent – the first one has *falling accent*, while another has *raising accent*. In order to characterize differences of accent we calculate such speech signal characteristics:

a) Pitch (see Fig. 1 c, d), according to modified autocorrelation method using clipping [12 – 14], with 30 ms window length and 1/3 overlapping;
b) Intensity (see Fig. 1 e, f), that could be expressed by:

$$I_l = \frac{1}{\max\limits_{r \in \rho} I_r} \sum_{k=1}^{T_l} \left| s(m-k) \right|, \qquad (1)$$

where $s(m)$ is a speech signal, T_l is a length of particular pitch period *l,* and *r* is a set of all intensity values.

From the characteristics of homographs becomes clear that in the case of falling accent – pitch and intensity curves after short rising period has long and steep falling stage, while in the case of rising accent – pitch and intensity curves alternation limits are 2 - 3

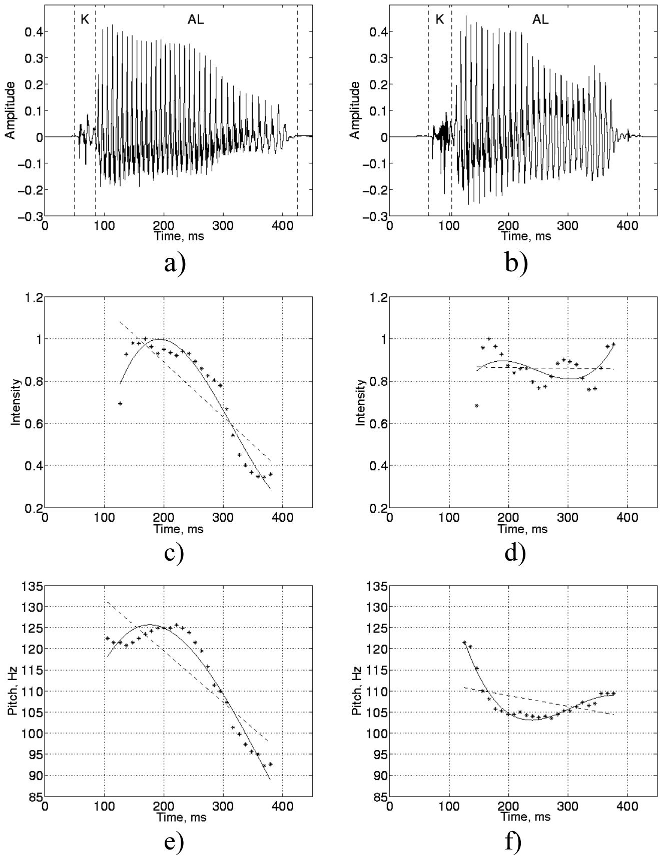


Figure 1. Characteristics of two homographs: (a, b) – waveforms; (c, d) – pitch; (e, f) – intensity. Characteristics in (c – f) are shown as: bullets – measured, dashed line – approximated by line, solid line – approximated by polynomial.

times smaller and curves has tendency to grow. It well confirms to the result in [8 - 10], stating that these two characteristics together with accent duration are main in the description of accent.

In our problem setup accent duration must be tailored to a real speech signal duration, that is why we do not include it into development of accent model.

In order to describe accent in terms of its characteristics lets implicitly express voiced speech signal s_v as a periodic, with constant period T_0 and intensity I_0, signal by:

$$s_v\left(I_0, T_0, m\right) = I_0 \cdot s_v\left(m - T_0\right). \tag{2}$$

Such speech signal definition does not formalize speech signal in detail, however enables to control it through chosen parameters. That is very suitable in non-stationary signal cases, when some speech signal characteristics could be determined easier than the speech signal model. Thus, taking into account that pitch $T(m)$ and intensity $I(m)$ in our case are time varying we could rewrite previous equation as [15]:

$$s_v\{I(m), T(m), m\} = I(m) \cdot s_v\{m - T(m)\}. \tag{3}$$

Now, accent model development squeezes into modeling of pitch and intensity.

The absolute values of pitch and intensity characteristics are very spread and depend on the particular speaker. Thus major modeling objective is not the modeling precision but rather preservation of the character of pitch and intensity variations, which is almost the same for a particular type of accent.

In accordance with principle “try simple things first”, we approximate pitch and intensity curves linearly, ending in such expression:

$$s_v^{lin}\left\{I(m), T(m), m\right\} = \left(i_1 \cdot m + i_0\right) \cdot s_v^{lin}\left(m - t_1 \cdot m - t_0\right), \tag{4}$$

where i_1, i_0 and t_1, t_0 are coefficients of linear approximation of intensity and pitch respectively.

Results of linear approximation in a sense of least squares are shown in Fig. 1 (c – f) by dashed lines. Obviously such approximation is inaccurate, however, it enables to considerably simplify calculations during restoration of homographs.

More accurate description of pitch and intensity could be made employing polynomial approximation by:

$$s_v^{poly}\{I(m),T(m),m\} = \sum_{r=0}^{N^i} i_r \cdot m^r \cdot s_v^{poly}\left(m - \sum_{r=0}^{N^t} t_r \cdot m^r\right), \tag{5}$$

where i_r and t_r are coefficients of polynomial approximation of intensity and pitch respectively.

Results of polynomial approximation in a sense of least squares are shown in Fig. 1 (c – f) by solid lines.

As was mentioned earlier, we assume that in a place of homograph speech signal is completely destroyed. However, there exists *a priori* information (extracted, e.g., from context) about its type and corresponding intensity and pitch characteristics. Application target is not real-time processing. Thus in the restoration of homograph we employ forward together with backward processing in time, more precisely:

$$\hat{s}_v(m) = w^f(m) \cdot \hat{s}_v^f(m) + w^b(m) \cdot \hat{s}_v^b(m), \tag{6}$$

where each restored signal is weighted with corresponding weighting function [16]:

$$w^{f/b}(m) = \frac{1}{2} \cdot \left\{1 \pm \cos\left(\frac{m \cdot \pi}{M_2 - M_1}\right)\right\}, \quad \forall\, m \in [M_1,\, M_2], \tag{7}$$

where M_1 and M_2 are limits of the restoration.

Signal restoration carried out from different directions inheritable is the same, the main difference being in the time direction, i.e. indexes. Thus further we will discuss processing of signal only in the forward direction.

It is obvious, that it is unnecessary to take into account variation of intensity and pitch at each time step, thus we use only their variation with pitch, in the following expressions introducing new variable l as pitch index. Therefore, expression (3) could be rewritten by:

$$\hat{s}_v(m) = \frac{I(l)}{I(l-1)} \hat{s}_v\left(\left\lfloor (m - T(l-1)) \frac{T(l-1)}{T(l)} \right\rfloor\right), \quad \forall \sum_{j=1}^{l-1} T(j) < m \leq \sum_{j=1}^{l} T(j), \tag{8}$$

where now intensity $I(l)$ and pitch $T(l)$ are varying with pitch and expression is valid for particular speech signal period. Note, that in the calculation of time indexes we must use operation of rounding ($\lfloor \bullet \rfloor$ to minus infinity) as ratio of periods could be non-integer number.

Substituting previously developed accent models (4) and (5) into (8) we get such expressions:

$$\hat{s}_v^{lin}(m) = (1 + i_1 / I(l-1)) \cdot \hat{s}_v^{lin}\left(\left\lfloor \frac{m - T(l-1)}{1 + t_1 / T(l-1)} \right\rfloor\right), \ \forall \sum_{j=1}^{l-1} T(j) < m \leq \sum_{j=1}^{l} T(j), \tag{9}$$

$$\hat{s}_v^{poly}(m) = \frac{\sum_{r=0}^{N^i} i_r \cdot l^r}{\sum_{r=0}^{N^i} i_r \cdot (l-1)^r} \cdot \hat{s}_v^{poly}\left(\left\lfloor \left(m - \sum_{r=0}^{N^r} t_r \cdot (l-1)^r\right) \cdot \frac{\sum_{r=0}^{N^t} t_r \cdot (l-1)^r}{\sum_{r=0}^{N^t} t_r \cdot l^r} \right\rfloor\right), \tag{10}$$

$$\forall \sum_{j=1}^{l-1} T(j) < m \leq \sum_{j=1}^{l} T(j)$$

Note, that we could interpret expressions (9) and (10) flowingly: particular sample restoration is based on *measured* values of intensity and pitch of previous period, knowing approximation coefficients and preceding period signal sample. In that way we end up with *recursive* restoration procedure, when current pitch period is formed from previous pitch period samples.

Fourth chapter – Experimental part. This chapter describes simulations of the restoration of homographs (Fig. 1 a, b).

Restoration is carried out employing (9), (10), (6) and (7) expressions. Example of both restorations of 9 pitches length together with original waveform are shown in Fig. 2. It is evident from figure that restored waveforms are close enough to the original. Listening to restored fragments also confirms that:

a) introduced distortions are inaudible,
b) during restoration accent information was reinstated.

In order to estimate mean bounds for the restoration, firstly we define MSE for restored pitch period:

$$\varepsilon = \frac{1}{(M_2 - M_1)} \sum_{m=M_1}^{M_2} \left(s_v(m) - \hat{s}_v(m)\right)^2, \tag{11}$$

where $M_1 = \sum_{j=1}^{l} T(j)$ and $M_2 = \sum_{j=1}^{l+1} T(j)$.

Then MSE of whole restoration length of M pitches, is:

$$e_M = \frac{1}{M} \sum_{l=1}^{M} \varepsilon_l \tag{12}$$

However such calculations do not take into account restoration segment position and thus results will be dependent on it. Lets

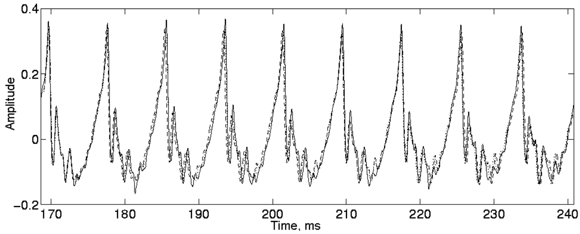


Figure 2. Homograph (Fig. 1 a) restoration results. Waveforms are shown as: solid line - original, dashed line – restored with linear accent model, dotted line - restored with polynomial accent model.

rewrite (12) introducing time shift variable Δ by:

$$e_{M,\Delta} = \frac{1}{M}\sum_{l=1}^{M} \varepsilon_{l-\Delta}\,. \tag{13}$$

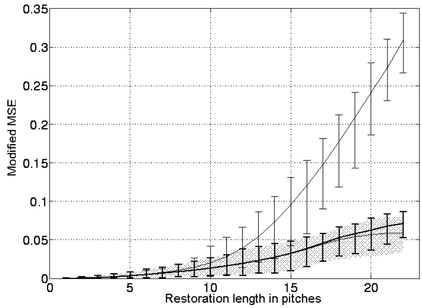
Finally, mean in a sense of data points sum square error of restoration length of M pitches and invariant to restoration segment position, is expressed by:

$$E_M = \frac{1}{\Delta_{max}}\sum_{\Delta=0}^{\Delta_{max}} e_{M,\Delta} = \frac{1}{M\cdot\Delta_{max}}\sum_{\Delta=0}^{\Delta_{max}}\sum_{l=1}^{M}\varepsilon_{l-\Delta} = \frac{1}{M\cdot\Delta_{max}}\sum_{l=1}^{M}\sum_{\Delta=0}^{\Delta_{max}}\varepsilon_{l-\Delta}\,, \tag{14}$$

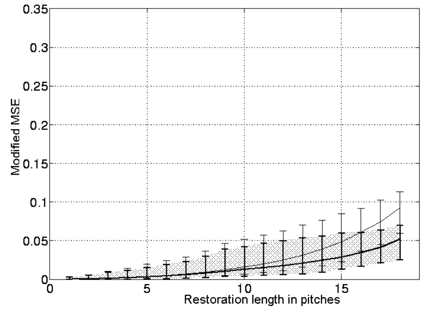
while its limits are:

$$E_M^{\min/\max} = \min_{l\in[1,N]}/\max\left(\frac{1}{\Delta_{\max}}\sum_{\Delta=0}^{\Delta_{\max}}\varepsilon_{l-\Delta}\right). \tag{15}$$

Results of expressions (14) and (15) for restoration with linear and polynomial approximation of characteristics for two homographs are shown in Fig. 3. Additionally there are shown results of our restoration method using original pitch and intensity characteristics.



a)

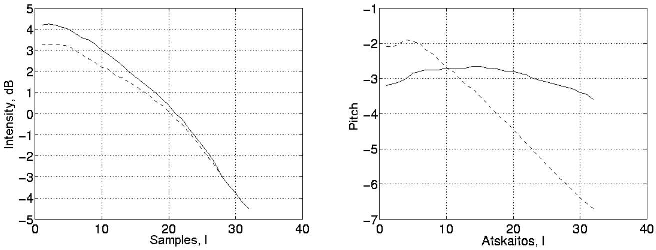


b)

Figure 3. Modified MSE and its limits for both homographs (*kã̃ltas* - a, *kal'tas* - b) restoration: thin line - restored with linear accent model, normal line - restored with polynomial accent model, thick line and shaded area - restored with original characteristics.

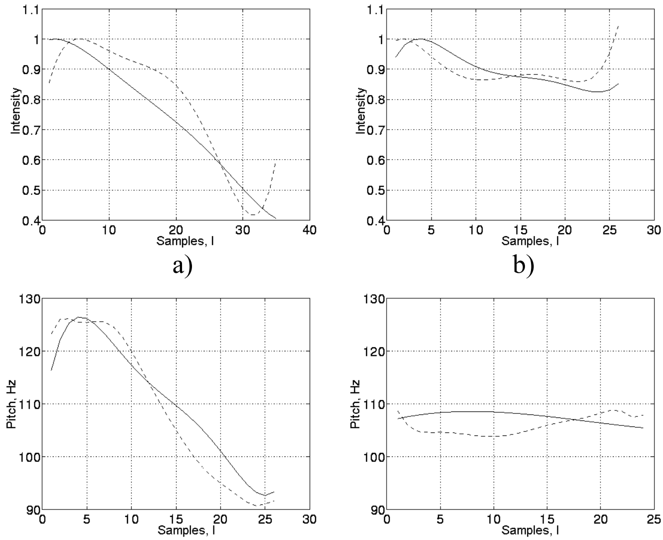
They indicate the lowest bounds possible to achieve during restoration employing the best approximation of characteristics.

From Fig. 3 (a), (b) is evident, that in both homographs restorations linear accent model employment as was expected results in biggest deviation from the original when restoration length increases. Employment of only 3rd order polynomial accent model sufficiently (~ 70 % - 80 %) improves restoration quality.



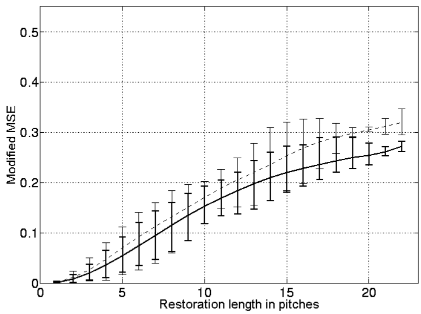
a) b)

Figure 4. Averaged and normalized intensity (a) and pitch (b) characteristics of both homographs (*kãltas* - solid, *kal'tas* - dashed lines).

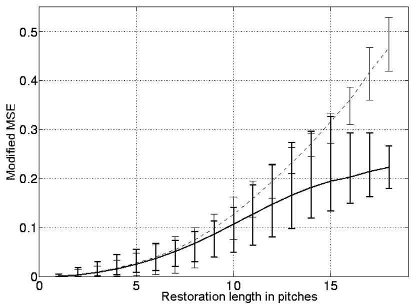


c) d)

Figure 5. Recalculated intensity (a, b) and pitch (c, d) characteristics of both homographs (dashed lines shows original characteristics).



a)



b)

Figure 6. Modified MSE and its limits for both homographs (*kãltas* - a, *kal'tas* - b) restoration when are used averaged and normalized intensity and pitch characteristics: solid line - restored with linear accent model, dashed line - restored with polynomial accent model.

In case, when original pitch and intensity characteristics are not available, the averaged and normalized characteristics should be used (Fig. 4). These characteristics should be adjusted according to speech signal pitch and intensity (Fig. 5). Further restoration is carried out in the same way as described above.

The modified MSE of restoration is shown in Fig. 6. As we can see, the restoration error increased. However, it is still about 20 % - 30 % less, than other known methods, because application of the linear accent model in restoration of speech signal could be viewed as modified method of restoration of lost speech samples described in [16].

Fifth chapter – Concluding remarks. In this chapter obtained results are summarized and conclusions are made. Possibilities of restoration of old Lithuanian folk records were analyzed.

Accent is one of the most important features of Lithuanian language. It does distinguishing function in common Lithuanian language (homographs) and dialects. That is way, in restoration of old folk records is very important to preserve these features as precisely as possible. With that end in view, characteristics that describe accent were analyzed. Also shown that existing noise suppression methods do not take into account non-stationary of speech signals or model only linear its change in time. However, accent distinguishes non-liner change in time of intensity and pitch characteristics. In order to restore these speech signal features:

1. Polynomial mathematical model of accent was created. It takes into account non-linear change in time of intensity and pitch.
2. Proposed new recursive method based on accent model for restoration of speech signals, which models non-linear change in time of intensity and pitch of speech signals.

Experimental results show that:

1. Application of proposed restoration method with polynomial accent model results in reduction by ~ 70 % and ~ 30 % of mean square error comparing to method proposed by Etter, when, respectively, original and averaged characteristics of intensity and pitch were used.
2. Experimental results confirm suitability of Polynomial Accent Model in accent restoration.

RESULTS AND CONCLUSIONS

1. An existing method of suppression of background noise improves signal to noise ratio, but reduces intelligibility.
2. An existing method of suppression of impulsive noise could be used than segment of distorted signal is stationary or change of its pitch and intensity is liner.
3. Polynomial mathematical model of accent was created. It takes into account non-linear change in time of intensity and pitch.
4. Proposed new recursive method based on accent model for restoration of speech signals, which models non-linear change in time of intensity and pitch of speech signals.
5. Application of proposed restoration method with polynomial accent model results in reduction by ~ 70 % and ~ 30 % of mean square error comparing to method proposed by Etter, when, respectively, original and averaged characteristics of intensity and pitch were used.
6. Experimental results confirm suitability of Polynomial Accent Model in accent restoration.

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TRIUKŠMO SUGADINTŲ PRIEGAIDŽIŲ GARSO ĮRAŠŲ RESTAURAVIMAS

Reziumė

Šiame darbe aprašomas triukšmo sugadintų priegaidžių garso įrašų restauravimas. Apžvelgiami senų įrašų etiniai restauravimo aspektai, skiriamieji lietuvių kalbos bruožai, priegaidžių tipai bei jas aprašančios charakteristikos.

Disertacijoje pateikta egzistuojančių foninio ir impulsinio triukšmų slopinimo kalbos signale metodų apžvalga. Svarbi problema slopinant triukšmus kalbos signale – modeliuoti netiesinį kalbos signalo kitimą laike. Tai svarbu restauruojant senus lietuvių liaudies tarmių įrašus, ypač homografus, nes juose svarbų vaidmenį atlieka priegaidės, kurios yra aprašomos netiesiniu kalbos signalo intensyvumo ir pagrindinio tono charakteristikų kitimu laike.

Priegaidės, restauravimui sukuriamas polinominis priegaidės modelis, įvertinantis kalbos signalo pagrindinio tono ir intensyvumo netiesinį kitimą laike. Pateikiamas naujas kalbos signalo restauravimo metodas, kuris, skirtingai nuo kitų žinomų metodų, naudodamas sukurtą priegaidės modelį, įvertina signalo nestacionarumą.

Atlikti eksperimentai parodo, kad priegaidės atstatymui naudojant pasiūlytą restauravimo metodą, kuriame panaudojamas sukurtas priegaidės modelis su tiksliomis priegaidės intensyvumo ir pagrindinio tono charakteristikomis, atstatyto signalo vidutinė kvadratinė paklaida apie 70 % mažesnė negu naudojant kitus žinomus metodus. Kai priegaidės modelyje naudojamos vidutinės jos intensyvumo ir pagrindinio tono charakteristikos restauravimo tikslumas pagerėja apie 20 – 30 %.

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Summary of Doctoral Dissertation
Technological Science, Electrical and Electronic Engineering (01T)

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SL 136. 1999 06 18. 1,39 apsk. leid. l.
Tiražas 100 egz. Užsakymas
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