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KURTOSIS AND NORMALIZED VARIANCE AS MEASURES OF SPEECH SIGNALS CLIPPING VALUE

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***Abstract*—*It is shown that the kurtosis and the normalized variance can be used as a measures of the clipping value of speech signals. The use of the proposed measures makes it possible to significantly simplify and speed up the clipping value calculations compare to the methods where preliminarily estimation of the probability density function of the analyzed speech signal is required. Subjective estimates of the clipped speech signals quality were obtained. Matching maps between the proposed objective measures and the subjective estimates of the clipped speech signals quality have been built. It was shown that the maps can be well approximated by polynomials of small (1st–4th) order. This fact indicates the possibility of construction of simple, in computational sense, algorithms for the control of clipped speech signals quality.*

Index Terms—Clipped speech; speech quality; clipping value; kurtosis; normalized variance; matching map.

I. INTRODUCTION

A ground navigation equipment of unmanned aerial vehicles (UAV) as well as perspective navigation and motion control systems of UAV can be developed with use of automatic speech recognition (ASR) systems. Therefore, the resistance of ASR systems to the effects of noise and reverberation, as well as to the presence of non-linear signal distortions, is an important task.

The maximum use of the speech transmission channel dynamic range is highly desirable, since it allows the highest signal-to-noise (SNR) ratio value. However, there is a risk of non-linear distortions due to signal clipping, in which large instantaneous values of the signal are replaced by some constant values.

At the same time, a small value of clipping is accompanied by small non-linear distortions that may be acceptable for listeners. Therefore, it is advisable to build a clipping detection algorithm so that it is preceded by an algorithm for evaluating the clipping value. A number of known methods of clipping detection are based on this rule and the difference in the shape or parameters of the probability density function (PDF) of the analyzed and undistorted signal can be used as a clipping value measure [1] – [5].

The variance of the analyzed signal was proposed in [1] as a measure of the clipping value. This idea seems reasonable, firstly, because the variance of the signal increases with increasing clipping value, and secondly, because of the simplicity of the calculations. However, the lack of recommendations on the normalization of the variance devalues this idea.

Several modifications for implementing a clipping detection technique are discussed in [5]. In one of these variants, where the PDF estimate is compared with a certain threshold function, a priori information about the reference PDF is required. Other cases are free from the need to have a priori information. In particular, it is proposed to calculate the difference between the PDF estimate and its linear approximation, and then compare this difference with the values of a certain threshold function. A priori information is also not required in the variant consisting in counting the number of peaks on the PDF tail and subsequent comparison of this quantity with the threshold value.

A method for detecting clipping by measuring mel-frequency cepstral coefficients was proposed in [6], but an obvious disadvantage of this method is its relatively high computational complexity that can be justified when using equipment for medical purposes.

In the clipping detection method described in [7], it is required to preliminarily construct a very rough histogram (using about 20 bins) PDF estimate of the analyzed signal. The histogram value for the highest signal values is proposed to use as a parameter, the value of which is used to decide on the presence of clipping.

A clipping detection method using histogram PDF estimate of the analyzed signal was also proposed in [8]. According to this method, two secondary histograms are formed from the primary histogram, one of which is obtained by convolution of the primary histogram with a narrow exponential averaging function, and the threshold histogram is obtained using a broad exponential averaging function. The difference between the two secondary histograms is then used to decide on the presence of clipping.

The SNR as a measure of the clipping value is proposed to be used in [2]. In this case, the signal means instantaneous values of the speech that are within the dynamic range of the transmission channel, and noise means instantaneous values of the speech that go beyond the limits of the dynamic range. Since the instantaneous "noise" values are not known, it is proposed to estimate its power by extrapolating the PDF of the analyzed speech beyond the channel dynamic range.

In papers [3], [4], as in [1], [2], a parametric approach was used, according to which it was proposed to estimate the "clipping coefficient" which characterizes the value of bursts on the tails of the PDF estimate of the analyzed speech. The clipping coefficient has two drawbacks: 1) low ability to evaluate the degree of clipping; 2) the need to pre-evaluate the PDF of the speech signal.

The kurtosis (fourth-order normalized central moment of the analyzed signal [9]) proposed in [10] as a measure of the clipping value of speech signals is free from both of these shortcomings.

At the same time, kurtosis has its own drawback: its range of possible values belongs to the interval [1; ∞], which is inconvenient in practice. The object of this paper is to show the existence of two clipping measures closely related to kurtosis, but free from this drawback. First, it is the reciprocal of the kurtosis, and second, it is the square root of the reciprocal of the kurtosis. Another object of the paper is to experimentally validate the proposed clipping measures.

II. PROBLEM STATEMENT

Signal $y(n)$ as a result of hard clipping of a signal $x(n)$ can be described by the relation

$$y(n) = \begin{cases} |x(n), & |x| < A, \\ |A \cdot \operatorname{sign}[x(n)], & |x| \geq A, \end{cases} \tag{1}$$

where n is the signal sample number; A is clipping threshold $\left(0 < A < C = \max|x(n)|\right)$; $\operatorname{sign}(\bullet)$ is sign function; $|\bullet|$ is module symbol.

The ability to detect clipping using kurtosis

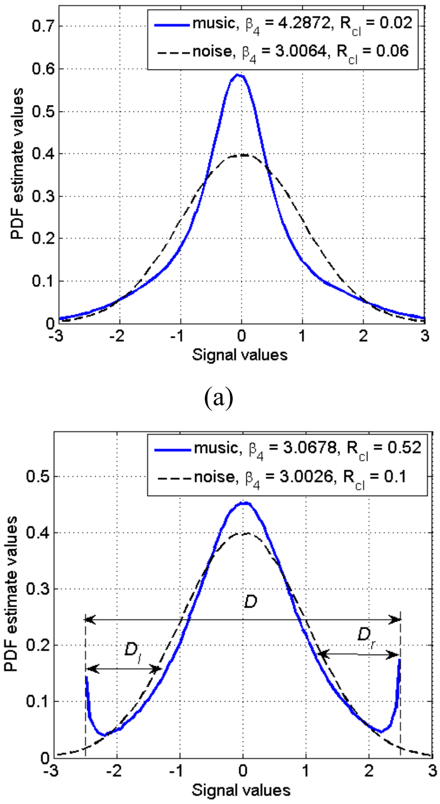
$$\beta_4 = \frac{\mu_4}{(\mu_2)^2}, \tag{2}$$

where μ_k is the central moment of the kth order, first proposed in [10], is shown in Fig. 1, where the solid line shows the graphs of PDF estimates of non-clipped (Fig. 1a) and clipped (Fig. 1b) signal. PDF estimates of nonclipped Gaussian noise are shown in Fig. 1 with dashed lines. As can be seen, the kurtosis β_4 values decrease when signal is clipping.

Estimation results of the proposed in [3] and [4] clipping coefficient

$$R_{cl} = \frac{2\max(D_l, D_r)}{D}, \tag{3}$$

where D is the difference between the maximum and minimum signal values; D_l and D_r are distances between the peaks of the PDF estimate, are also shown in Fig. 1. As can be seen, the values of the clipping coefficient R_{cl} increase when signal is clipping.



(a)

(b)

Fig. 1. Estimates of PDF, β_4 and R_{cl} for unclipped (a) and clipped (b) signals

When comparing (2) and (3), it can be seen that the obvious advantage of measure (2) is the relative simplicity of the calculations, caused by the absence of the need to preliminarily evaluate the PDF of the analyzed signal. As a result, there is no problem of choosing the type of PDF estimator, as well as the related problem of optimizing the parameters of this estimator. Finally, an important advantage of measure (2) is that the matching map between subjective and objective estimates of the clipping value of speech signals is a monotonous function that does not contain sharp kinks, which indicates the suitability of this measure for evaluating the clipping value and speech quality [10].

As regards the clipping coefficient (3), it was noted in [4] that R_{cl} makes it possible to detect strong clipping, but it is of little use for estimating the value of clipping. At the same time, a useful property of a R_{cl} parameter is the finiteness of the interval of its possible values, which is convenient for engineering applications. Indeed, when analyzing (3), it is easy to conclude that the possible R_{cl} values belong to the finite range [0; 1], and the value $R_{cl} = 0$ corresponds to the situation of clipping absence.

Unfortunately, the β_4 parameter does not possess the property of finiteness of the possible values interval. It can be shown that the β_4 values cannot be less than + 1, and the value $\beta_4 = 1$ corresponds to the maximum clipping situation. Indeed, denoting $Z = \left(X - E\{X\}\right) / \left(V\{X\}\right)^{\frac{1}{2}}$ where $E\{\bullet\}$ is the expectation symbol; X and Z are random variables; $V\{\bullet\}$ is the variance symbol, it can be obtained [11] $\beta_4 = V\left\{Z^2\right\} + 1,$ whence we have $\beta_4 \geq 1.$ Thus, the possible values of the parameter β_4 belong to the range of [1; ∞]. For real nonclipped signals, the β_4 values can reach 30-50 in the case of musical signals [12] and 7-12 in the case of speech signals [14], which is not convenient in practice.

III. Problem Solution

For engineering applications, it may be more convenient to use parameters

$$\gamma_4 = \frac{1}{\beta_4} = \frac{\left(\mu_2\right)^2}{\mu_4}, \tag{4}$$

$$\eta_4 = \frac{1}{\sqrt{\beta_4}} = \frac{\mu_2}{\sqrt{\mu_4}}, \tag{5}$$

the possible values of which belong to the finite interval [0; 1], and the values $\gamma_4 = 0$ and $\eta_4 = 0$ correspond to the situation of clipping absence.

Note that η_4 is the normalized signal variance where normalizing is made by dividing by the square root of the central moment of the fourth order. Hence, parameter (4) can be interpreted as the square of the normalized variance of the analyzed process. As noted above, the idea of using signal variance to estimate the value of a signal clipping has been expressed before [1]. Unfortunately, this idea was not worked out to the extent sufficient for technical implementation, since nothing was said in [1] about the need to normalize the signal variance, nor about the type of the normalizing coefficient. The measures γ_4 and η_4 proposed in this paper do not have this drawback.

Thus, one of the problems solved in this paper is the experimental verification of the assumption that γ_4 and η_4 parameters can be used as objective measures of the clipping value of speech signals. Since γ_4 and η_4 parameters can be also considered as objective measures of the quality of clipped speech signals, the second task is to build matching maps between objective and subjective measures of the clipped signals quality. The practical value of such a map is the ability to calibrate objective measures γ_4 and η_4.

IV. Organization of Experimental Studies

Speech signals from 8 speakers (four men and four women), each lasting 15 seconds, were used for experimental studies. All signals were recorded in a muffled room with a sampling rate of 22050 Hz and a bit depth of 16 bits.

The parameter

$$k = 20\lg\left(\max\left|x(n)\right| / A\right), \tag{6}$$

was used for regulating the clipping value [10].

The assessment of the quality of speech signals distorted by clipping was carried out by subjective and objective methods. Subjective assessment of the quality of clipped signals was carried out using the Degradation Mean Opinion Score (DMOS) scale [13], [14]. Six listeners aged 19 to 21 years without hearing impairment participated in the tests. Objective assessment was carried out in Matlab using measures (2) – (5). In estimating the β_4 parameter, the unbiased version of the estimate was used

$$\overline{\beta}_4 = \frac{(N-1)\left((N+1)\overline{\beta}'_4 - 3(N-1)\right)}{(N-2)(N-3)} + 3,$$

$$\overline{\beta}'_4 = \frac{\frac{1}{N}\sum_{n=1}^{N}\left(y(n) - \overline{m}_y\right)^4}{\left(\frac{1}{N}\sum_{n=1}^{N}\left(y(n) - \overline{m}_y\right)^2\right)^2},$$

$$\overline{m}_y = \frac{1}{N}\sum_{n=1}^{N} y(n).$$

Estimation of R_{cl} parameter was carried out in accordance with the algorithm given in [4]. In this case, the histogram PDF estimates with 100 bins have been used.

V. Experimental Results

The result of the subjective assessment $(\mathrm{DMOS}(k)\text{dependence})$ of the clipped speech quality as function of the clipping value k is shown in Fig. 2 [10]. As can be seen, the quality of clipped speech remains subjectively good $(\mathrm{DMOS} \geq 4)$ under the $0 \leq k \leq 8$ dB condition.

The results of objective measures $\beta_4(k)$, $\gamma_4(k)$, $\eta_4(k)$ and $R_{cl}(k)$ dependencies are presented in Fig. 3. As can be seen, the $\beta_4(k)$, $\gamma_4(k)$, and $\eta_4(k)$ dependencies change slowly and monotonously in the range of $0 \leq k \leq 8$ dB, and this behavior is observed both on average and for individual instances of the analyzed signal. In the $8 < k \leq 20$ dB interval, the rate of change of $\beta_4(k)$, $\gamma_4(k)$, and $\eta_4(k)$ dependencies increases but remains approximately the same.

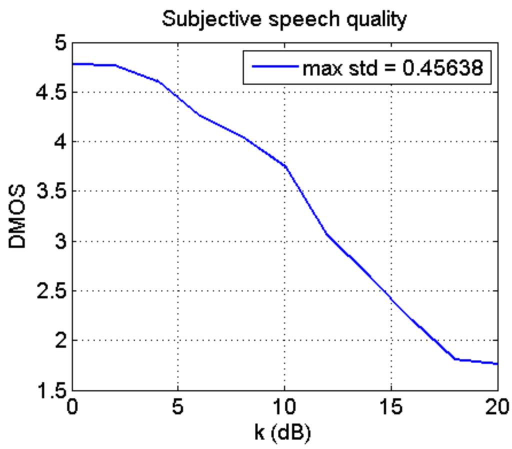


Fig. 2. DMOS(k) dependence [10]

The behavior of $R_{cl}(k)$ dependence is another. The $R_{cl}(k)$ graph practically does not change in the $0 \leq k \leq 3$ dB interval, grows very slowly in the $9 \leq k \leq 20$ dB interval and grows very rapidly in the intermediate $3 < k < 9$ dB interval. Considering the $R_{cl}(k)$ dependence for individual instances of the analyzed signal, we see that in the $0 \leq k \leq 8$ dB interval, a significant violation of monotonicity is possible, and the location of the point from which the rapid increase in $R_{cl}(k)$ dependence is beginning varies widely from $0 \leq k \leq 2$ dB to $0 \leq k \leq 6$ dB.

The preference for the use of $\beta_4(k)$, $\gamma_4(k)$, and $\eta_4(k)$ parameters instead of R_{cl} becomes more evident when building $\mathrm{DMOS}(\beta_4)$, $\mathrm{DMOS}(\gamma_4)$, $\mathrm{DMOS}(\eta_4)$ and $\mathrm{DMOS}(R_{cl})$ matching maps (Fig. 4). Averaged over the speakers $\beta_4(k)$, $\gamma_4(k)$, $\eta_4(k)$ and $R_{cl}(k)$ dependencies were used when building the matching maps. The values of the correlation coefficients between the DMOS estimates and the $\beta_4(k)$, $\gamma_4(k)$, $\eta_4(k)$ and R_{cl} estimates are shown in Table I.

TABLE I. Correlation Coefficients

Objective measure	β_4	γ_4	η_4	R_{cl}
Correlation coefficient	0.985	-0.987	-0.994	-0.885

Values of coefficients of approximating polynomials

$$y = a_0 + a_1 x + a_2 x^2 + a_3 x^3 + \ldots$$

for $\mathrm{DMOS}(\beta_4)$, $\mathrm{DMOS}(\gamma_4)$, $\mathrm{DMOS}(\eta_4)$ and $\mathrm{DMOS}(R_{cl})$ dependencies are given in Table II.

As can be seen, the use of the 4th order polynomial is quite enough to achieve a small approximation error of the $\mathrm{DMOS}(\beta_4)$, dependence. In the case of $\mathrm{DMOS}(\gamma_4)$, dependence, a second-order polynomial can be used, and the $\mathrm{DMOS}(\eta_4)$, dependence is well approximated by a first-order polynomial. As far as $\mathrm{DMOS}(R_{cl})$ dependence, a small approximation error can be achieved only when using a polynomial of at least 6th order. However, even in this case, the quality of the approximation cannot be considered satisfactory due to the nonmonotonic nature of the obtained matching map.

The $\mathrm{DMOS}(R_{cl})$ dependence approximation error can be significantly reduced by using a piecewise linear approximation (Fig. 5):

$$\mathrm{DMOS} = \begin{cases} 4.95 - 1.17 \cdot R_{cl}, & 0 \leq R_{cl} \leq 0.9, \\ 13.6 - 10.72 \cdot R_{cl}, & 0.9 < R_{cl} \leq 1. \end{cases} \quad (7)$$

Relation (7) and the graph of Fig. 5 can serve as a good illustration of the conclusion [4] concerning the low suitability of a R_{cl} measure for estimating the clipping value. Indeed, R_{cl} values belong to the interval (0.05; 0.85) and cover 80% of the possible values range [0; 1] for DMOS $\geq$ 4, i.e. for good quality of clipped signal. There is a sharp drop in signal quality from 4 points to 3.2 points of DMOS scale, i.e. from good quality to very mediocre quality, in a small interval (0.85; 0.95) of R_{cl} values. Thus, we can conclude that the R_{cl} values scale is used inefficiently.

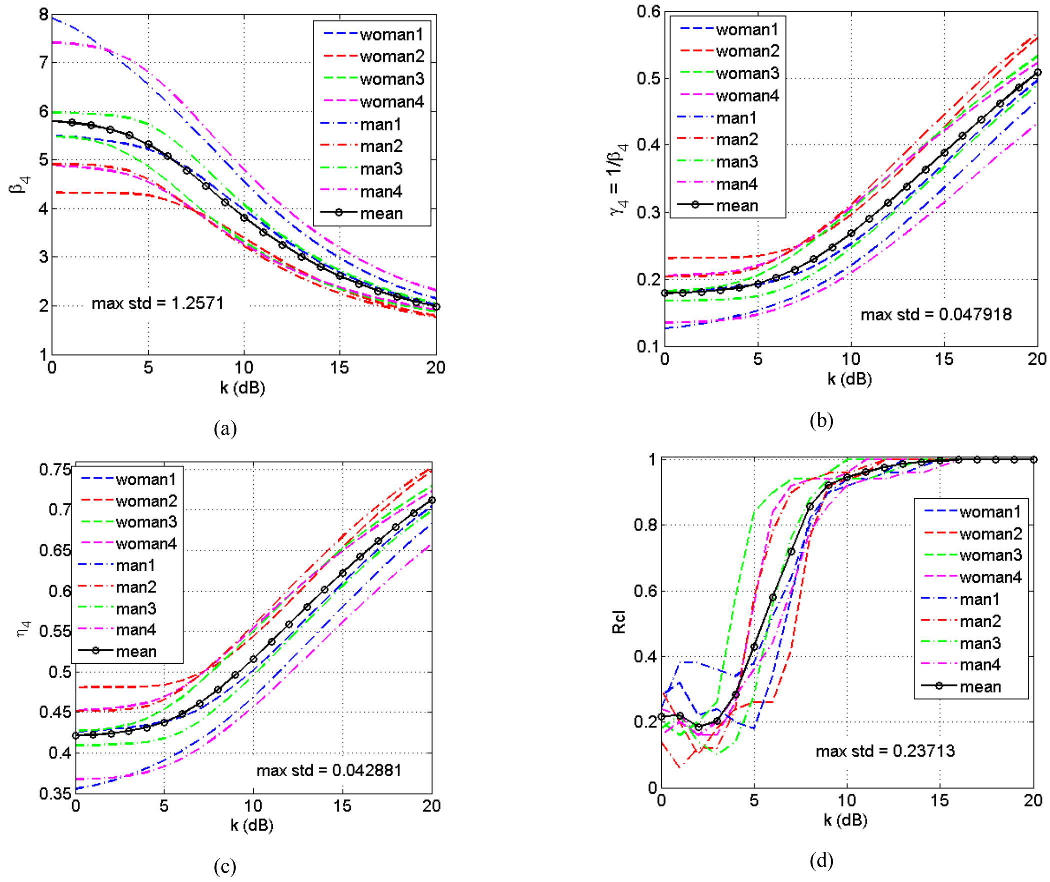


Fig. 3. Estimates of $\beta_4(k)$ (a); $\gamma_4(k)$ (b); $\eta_4(k)$ (c) and $R_{cl}(k)$ (d) dependencies

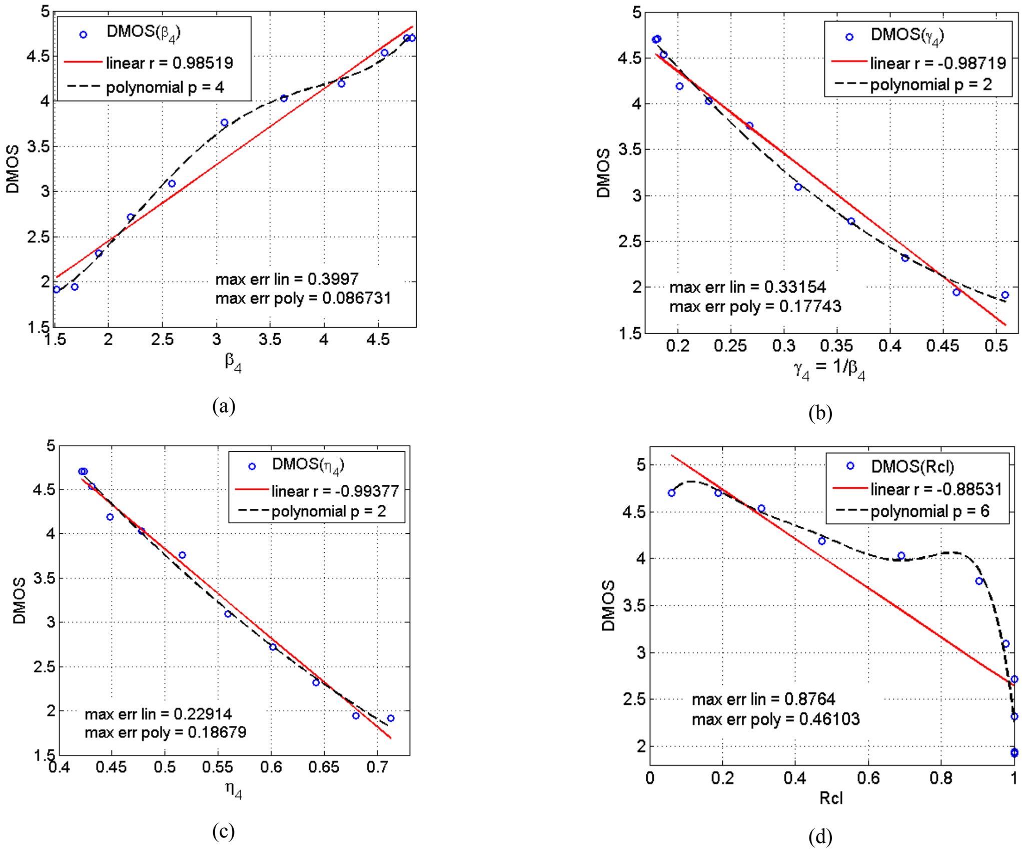


Fig. 4. Matching maps: DMOS(β_4) (a); DMOS(γ_4) (b); DMOS(η_4) (c) and DMOS(R_{cl}) (d)

TABLE II. COEFFICIENTS OF APPROXIMATING POLYNOMIALS

Polynomial coefficient	β_4 $p = 4$	γ_4 $p = 2$	η_4 $p = 1$	R_{cl} $p = 5$
a_0	5.72604	7.52257	8.85577	4.08686
a_1	-7.88317	-18.53578	-10.06244	16.62126
a_2	5.12563	14.48132	-	-132.09174
a_3	-1.20305	-	-	460.86927
a_4	0.09757	-	-	-834.49338
a_5	-	-	-	752.48433
a_6	-	-	-	-265.21887

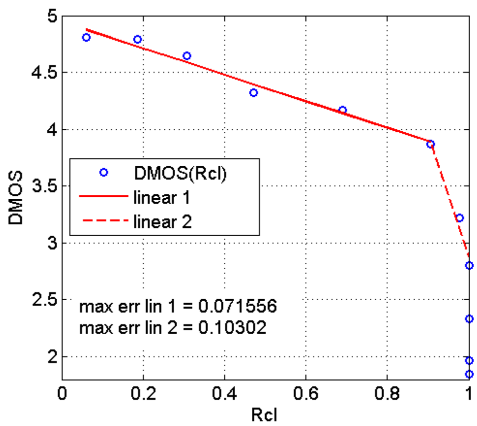


Fig. 5. Piecewise linear approximation of the DMOS(R_{cl}) matching map

On the contrary, the smooth nature of the DMOS(β_4), DMOS(γ_4) and DMOS(η_4) dependencies (Figs. 4a, b, c) indicates the suitability of kurtosis β_4, reciprocal of the kurtosis $\gamma_4 = 1/\beta_4$, as well as the square root of the reciprocal of the kurtosis $\eta_4 = 1/\sqrt{\beta_4}$, for use as clipping value measures. An additional argument in favor of the η_4 and $\gamma_4 = \eta_4^2$ parameters is the almost linear nature of the DMOS(γ_4) and DMOS(η_4) dependencies which makes it possible to significantly simplify the recalculation of γ_4 and η_4 values into the DMOS scale values.

As noted above, the η_4 parameter can be considered as the normalized variance of the process being analyzed, and the normalization is carried out by dividing by the square root of the central moment of the fourth order. This interpretation of the η_4 parameter as a measure of the clipping value agrees with the idea expressed in [1] of using the variance of the process being analyzed as a measure of the clipping value. The variance normalizing proposed in this paper allows one to make this idea suitable for engineering applications.

VI. CONCLUSION

It is shown in this paper that the kurtosis, the reciprocal of it, as well as the square root of the reciprocal of the kurtosis, can be used as objective measures of the clipping value of speech signals. Measure in the form of the square root of the reciprocal of the kurtosis can be considered as the normalized variance of the analyzed process, and the normalizing is carried out by dividing by the square root of the central moment of the fourth order. This measure seems to be the most promising for engineering applications.

The matching maps between the values of the proposed objective measures, on the one hand, and the values of subjective estimate of the of clipped speech quality, on the other hand, are obtained. These matching maps are monotonic functions that do not contain sharp kinks, which allow one to conclude that the proposed parameters are suitable as measures of the clipping speech quality.

An important advantage of the proposed measures is the ability to significantly simplify and speed up the calculations due to the absence of the need to preliminarily estimate the probability density of the analyzed speech signal. Another advantage of the proposed objective measures is that the original signal may be unknown.

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А. М. Продеус, І. В. Котвицький, М. В. Дідковська, К. А. Кухарічева. Ексцес та нормована дисперсія в якості мір ступеня кліпування мовних сигналів

Показано, що ексцес та нормована дисперсія можуть використовуватися в якості мір кліпування мовних сигналів. Використання запропонованих мір дає можливість значно спростити та пришвидшити обчислення значення кліпування порівняно із методами, що потребують попереднього оцінювання функції щільності ймовірності мовного сигналу, що аналізується. Отримано суб'єктивні оцінки якості кліпованих мовних сигналів. Побудовано карти відповідності між запропонованими об'єктивними мірами та суб'єктивними

оцінками якості кліпованих мовних сигналів. Показано, що ці карти можуть бути добре наближені поліномами малого (I–IV) порядку. Цей факт вказує на можливість побудови простих, в обчислювальному сенсі, алгоритмів контролю якості кліпованих мовних сигналів.
Ключові слова: кліпований мовний сигнал; якість мовного сигналу; ступінь кліпування; ексцес; нормована дисперсія; карта відповідності.

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А. Н. Продеус, И. В. Котвицкий, М. В. Дидковская, Е. А. Кухаричева. Эксцесс и нормированная дисперсия в качестве мер степени клиппирования речевых сигналов

Показано, что эксцесс и нормированная дисперсия могут использоваться в качестве мер клиппирования речевых сигналов. Использование предложенных мер позволяет значительно упростить и ускорить вычисления степени клиппирования по сравнению с методами, требующими предварительной оценки функции плотности вероятности анализируемого речевого сигнала. Получены субъективные оценки качества клиппированых речевых сигналов. Построены карты соответствия между предложенными объективными мерами и субъективными оценками качества клиппированых речевых сигналов. Показано, что эти карты могут быть хорошо аппроксимированы полиномами малого (I–IV) порядка. Этот факт указывает на возможность построения простых, в вычислительном смысле, алгоритмов контроля качества клиппированых речевых сигналов.
Ключевые слова: клипированый речевой сигнал; качество речевого сигнала; степень клиппирования; эксцесс; нормированная дисперсия; карта соответствия.

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