THEORY AND METHODS OF SIGNAL PROCESSING

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ASSESSMENT OF CLIPPED SPEECH QUALITY

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***Abstract*—*Clipping of speech leads to the appearance of higher orders harmonics and, as a result, to reducing of accuracy of automatic speech recognition systems used as kind of artificial intellect of aircraft control systems and flight control systems for unmanned aerial vehicles. In this paper, the subjective and objective estimates of clipped speech quality are presented. It was shown that subjective speech quality Degradation Mean Opinion Score scale values are about 4.5, 3.5 and 2.5 for degrees of clipping 5 dB, 10 dB and 15 dB, respectively. Establishing of this rule allows find the boundary permissible degree of clipping based on certain requirements to the speech quality. Dependencies of objective speech quality measures such as segmental signal-to-noise ratio, frequency weighted segmental signal-to-noise ratio, log-spectral distortion, bark-spectral distortion and perceptual evaluation of speech quality on the clipping degree are obtained. It was shown also that kurtosis can be used as clipped speech quality measure. Calculations of correlation coefficients and matching maps which establish relationship between objective and subjective speech quality measures have been made. Obtained results allow concluding that objective speech quality measures can be applied to evaluate both the clipped speech quality and the degree of speech signals clipping.*

Index Terms—Clipped speech signal quality; objective measure; subjective measure; matching map; correlation coefficient.

I. INTRODUCTION

Clipping is a signal distortion kind when signal amplitude is limited by some threshold. Clipping leads to the appearance of harmonics of higher orders and increase in the level of high-frequency components of the signal. As a result, the perceived speech quality and intelligibility are reduced. Obviously, such distortions of speech signals are undesirable both in systems of air navigation and communication and in voice control systems of unmanned aerial vehicles (UAVs). Thus, the task of estimating the clipped speech quality and the allowable degree of speech clipping is quite relevant [1] – [4].

Two main approaches to clipping detecting and to estimation of clipping degree are used: time-domain analysis [1] and probability density functions (PDFs) analysis [3] – [7].

A clipping detecting technique, where a frequency weighted signal matched pseudo-differentiator is used, has been proposed in [1]. The main advantage of this time-domain technique is its relative simplicity. Unfortunately, the accuracy of estimating the degree of clipping of speech signals by this method is low [5].

A method for clipped signals signal-to-noise ratio (SNR) estimation by means of signal amplitude distribution analysis was proposed in [3]. Experiments showed that this method can provide a high accuracy in the clipping detection, and is relatively simple for calculations.

Another technique where calculation of the distances between local peaks of the tails and central part of the PDF estimator is made was proposed in [4]. The advantage of the method is that parameters of the distorted signal (sampling frequency, power and mean value) may vary within wide ranges. At the same time, the technique has some serious restrictions. For example, it is inoperable for signals containing powerful harmonic components and give wrong results for signals with a lot of zero samples.

The influence of signal clipping on quality of speaker recognition was studied in [5] where a simple clipping detection technique based on analysis of the PDF estimator edge values was proposed.

To detect regions of clipping, the signal's samples values PDF estimator and the signal's shape in the time-domain were analysed in [6]. The advantages of proposed algorithm are its computational efficiency and robustness against different types of clipping and to the clipping level.

The Mel-frequency cepstral coefficients as classification signs and special decision making system were proposed in [7] to detect and classify distortions in pathological voice signal.

II. PROBLEM STATEMENT

Unfortunately, well known objective speech quality [8] and speech intelligibility [9] measures, such as segmental signal-to-noise ratio (SSNR), frequency weighted segmental signal-to-noise ratio (fwSNR), log-spectral distortion (LSD), bark-spectral distortion (BSD), and perceptual evaluation of speech quality (PESQ) almost did not occur in the above-mentioned and other similar studies [8] - [18]. This situation can be explained by the fact that initial undistorted signal is usually assumed to be unknown.

Thus, the first objective of this paper is to study the case where the initial undistorted signal is known. An attempt of above-mentioned objective measures using both for clipped speech quality assessment and clipping degree estimation is made.

Second objective of the paper is trying of kurtosis as clipped speech quality measure. This measure seems particularly interesting because it is not required that the original undistorted signal be known. Detection of radio frequency interference in microwave radiometers [19] and distinguishing the seismic signal generated by a person's footsteps from other signals [20] are only few examples of similar tasks solving. Transmission system of an aircraft control system was also studied by means of the kurtosis calculation at different frequency bands in order to detect gearbox failure [21].

III. CLIPPED SIGNAL MODEL

Double-sided hard clipped signal $y(n)$ was formed from clean speech signal $x(n)$ by the rule

$$y(n) = \begin{cases} x(n), |x| < A, \\ A \cdot \mathrm{sgn}[x(n)], |x| \geq A, \end{cases}$$

where n is a discrete time index, A is the clipping threshold ($0 < A < \max|x(n)|$), $\mathrm{sgn}(\cdot)$ is the sign function, $|\cdot|$ is absolute value sign. Thus, the 'clipping degree' can be defined as [5]

$$\gamma = 1 - \frac{A}{\max|x(n)|}.$$

In this paper, another but close to γ clipping degree measure is used:

$$k = 20\lg\left(\max|x(n)|/A\right).$$

IV. EXPERIMENT ORGANIZATION

Subjective and objective evaluations of speech quality were performed for the same of 8 speech signals (4 male and 4 female) as in [10] and [11] where the signals sampling rate 22050 Hz and 16 bits quantization depth were used. Six listeners with an average age 22 years without hearing loss listened to randomly ordered records of distorted signals of 7 -10 s duration. Degradation Mean Opinion Score (DMOS) scale [12] was used upon subjective speech quality assessment.

The objective speech quality assessment was performed for signals of 15 s duration. Six measures were used upon objective speech quality estimation. SSNR [13] is the simplest quality measure for clean $x(n)$ and degraded $y(n)$ signals:

$$SSNR = \frac{1}{R}\sum_{r=1}^{R} 10\lg\left[\frac{\sum_{n=N(r-1)+1}^{Nr} x^2(n,r)}{\sum_{n=N(r-1)+1}^{Nm}[x(n,r) - y(n,r)]^2}\right]$$

where $x(n,r)$ and $y(n,r)$ are nth samples of rth frames of signals $x(n)$ and $y(n)$, R is frames number, N is frame samples number. When calculating SSNR, only those signal segments are considered for which the SSNR value belongs to interval [-10, +35] dB.

Log-spectral distortion is also simple for calculation frequency-domain measure [14]:

$$LSD = \frac{2}{JR}\sum_{r=1}^{R}\sum_{j=1}^{J}\left|G\{X(j,r)\} - G\{Y(j,r)\}\right|,$$

$$G\{X(j,r)\} = \max\{20\lg(|X(j,r)|), \delta\},$$

$$\delta = \max_{j,r}\{20\lg(|X(j,r)|)\} - 50,$$

where $X(j,r)$ and $Y(j,r)$ are rth frames of discrete Fourier transforms of $x(n,r)$ and $y(n,r)$, respectively, j is frequency sample index, J is spectrum samples number.

Frequency weighted segmental signal-to-noise ratio is another frequency-domain measure [8]:

$$fwSNR = \frac{10}{R}\sum_{r=1}^{R}\frac{\sum_{k=1}^{K} W(k,r)\lg\frac{|X(k,r)|^2}{(|X(k,r)| - |Y(k,r)|)^2}}{\sum_{k=1}^{K} W(k,r)},$$

where r and k are indices of the frames and critical bands, respectively, R is the frames number, K is the critical bands number, $|X(k,r)|$ and $|Y(k,r)|$

are the $x(n,r)$ and $y(n,r)$ frames amplitude spectrums, respectively, calculated using a Gaussian window.

Features of the human auditory system are taking in account more thoroughly when BSD measure [12] as perceptual indicator is calculating:

$$BSD=\frac{\sum_{r=1}^{R}\sum_{k=1}^{K}\left[B_x(k,r)-B_y(k,r)\right]^2}{\sum_{r=1}^{R}\sum_{k=1}^{K}\left[B_x(k,r)\right]^2}$$

where $B_x(k,r)$ and $B_y(k,r)$ are bark spectrums of $x(n,r)$ and $y(n,r)$, respectively, k is critical band index.

The algorithm for calculating the PESQ measure is very bulky; its descriptions can be found in [15] and [16]. In this paper, a wideband PESQ version was used.

A kurtosis measure is defined as

$$\text{krts}=\frac{\frac{1}{N}\sum_{n=1}^{N}\left(x(n)-\bar{x}\right)^4}{\left(\frac{1}{N}\sum_{n=1}^{N}\left(x(n)-\bar{x}\right)^2\right)^2}.$$

Unbiased kurtosis estimator was used in the paper

$$\text{kurtosis}=\frac{N-1}{(N-2)(N-3)} \times\left((N+1)\cdot\text{krts}-3(N-1)\right)+3.$$

V. EXPERIMENTAL RESULTS

Averaged, on 6 listeners and 8 speakers, result of subjective speech quality evaluation is shown in Fig. 1.

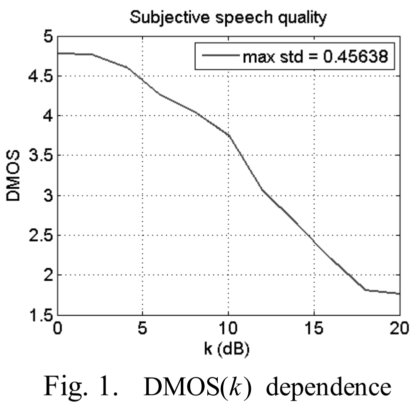
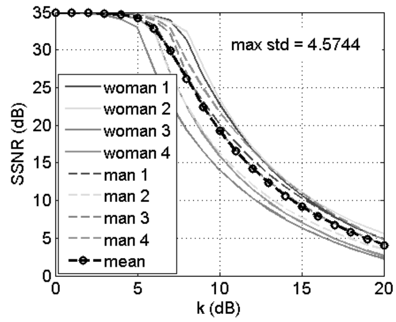


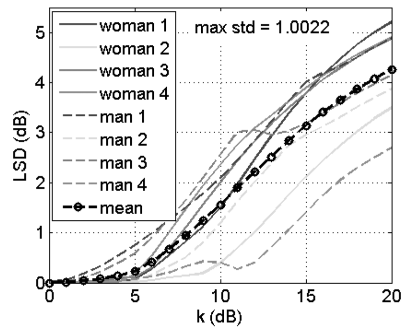
Fig. 1. DMOS(k) dependence

As can be seen from Fig. 1, the clipped speech quality decreases monotonically when clipping is increasing. Thus, increasing the degree of clipping k from 0 dB to 5 dB leads to a decrease in the speech quality from 5 points to 4.5 points on the DMOS scale. Further, the speech quality is reduced to 3.5 points with an increase in k to 10 dB, and reaches 2.5 points with an increase of k to 15 dB.

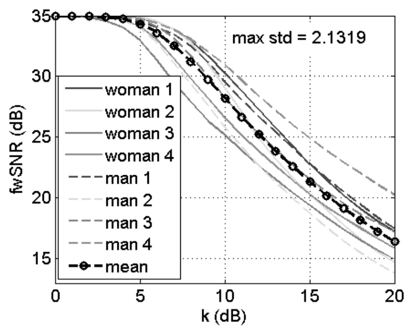
The six objective measure estimates are shown in Fig. 2.



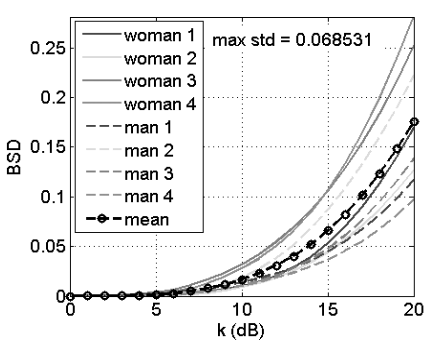
(a)



(b)

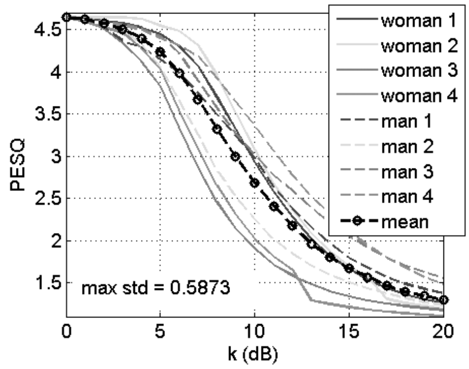


(c)

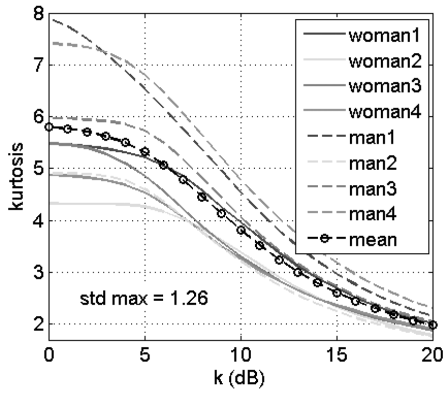


(d)

Fig. 2. Objective measure estimates: SSNR (a), LSD (b), fwSNR (c), BSD (d), PESQ (e), kurtosis (f)



(e)



(f)

Fig. 2. Ending. (See also p. 13)

Correlation coefficients as measures of statistical relationship between averaged objective and subjective estimates are shown in Table I.

Comparison of the graphs in Figs 1 and 2 indicates the performance of studied objective speech quality measures. Coefficients $a_0,\dots,a_4$ values of approximating polynomials

$$y = a_0 + a_1x + a_2x^2 + a_3x^3 + a_4x^4$$

for matching maps (Fig. 3) (the maximum errors of approximation are shown in Table II) are shown in Table III.

Using Fig. 2 graphs allows you to evaluate the degree of clipping by means of objective speech quality measures. As can be seen from Fig. 2b, LSD (k) dependency may be non-monotone for $k \geq 10$ dB, i.e. for strong clipping. Other studied objective measures dependencies are monotonic for $0 \leq k \leq 20$ dB interval.

When comparing correlation coefficients (Table I) and matching maps (Fig. 3), we can see they are in good accordance. In particular, BSD measure has minimum correlation coefficient value -0.908 and maximum polynomial approximation error. At the same time, matching maps are much more useful, because using them allows you to calibrate objective indicators of speech quality.

TABLE I. CORRELATION COEFFICIENTS

Object. measure	**SSNR**	**LSD**	**fwSNR**	**BSD**	**PESQ**	**Kurtos.**
Correl. coeff.	0.983	–0.996	0.996	–0.908	0.976	0.987

TABLE II. APPROXIMATION ERRORS

Polynomial degree p	**SSNR**	**LSD**	**fwSNR**	**BSD**	**PESQ**	**Kurtosis**
1	0.446	0.163	0.198	0.711	0.505	0.403
2	0.284	0.181	0.199	0.303	0.240	0.168
3	0.249	0.187	0.214	**0.205**	0.119	0.166
4	**0.108**	**0.126**	**0.124**	0.210	**0.083**	**0.080**
5	0.096	0.118	0.112	0.203	0.100	0.102

TABLE III. COEFFICIENTS OF APPROXIMATING POLYNOMIALS

Polynom. coefficient	**SSNR** $p = 4$	**LSD** $p = 4$	**fwSNR** $p = 4$	**BSD** $p = 3$	**PESQ** $p = 4$	**Kurtosis** $p = 4$
a_0	2.034	4.768	43.160	4.626	0.167	7.196
a_1	–0.201	–1.167	–7.101	69.61	–0.084	–8.584
a_2	0.039	0.596	0.436	458.25	1.613	4.349
a_3	0.002	–0.247	–0.011	–1177	–0.559	–0.819
a_4	–2e–05	0.031	1.1e–04	0	0.056	0.054

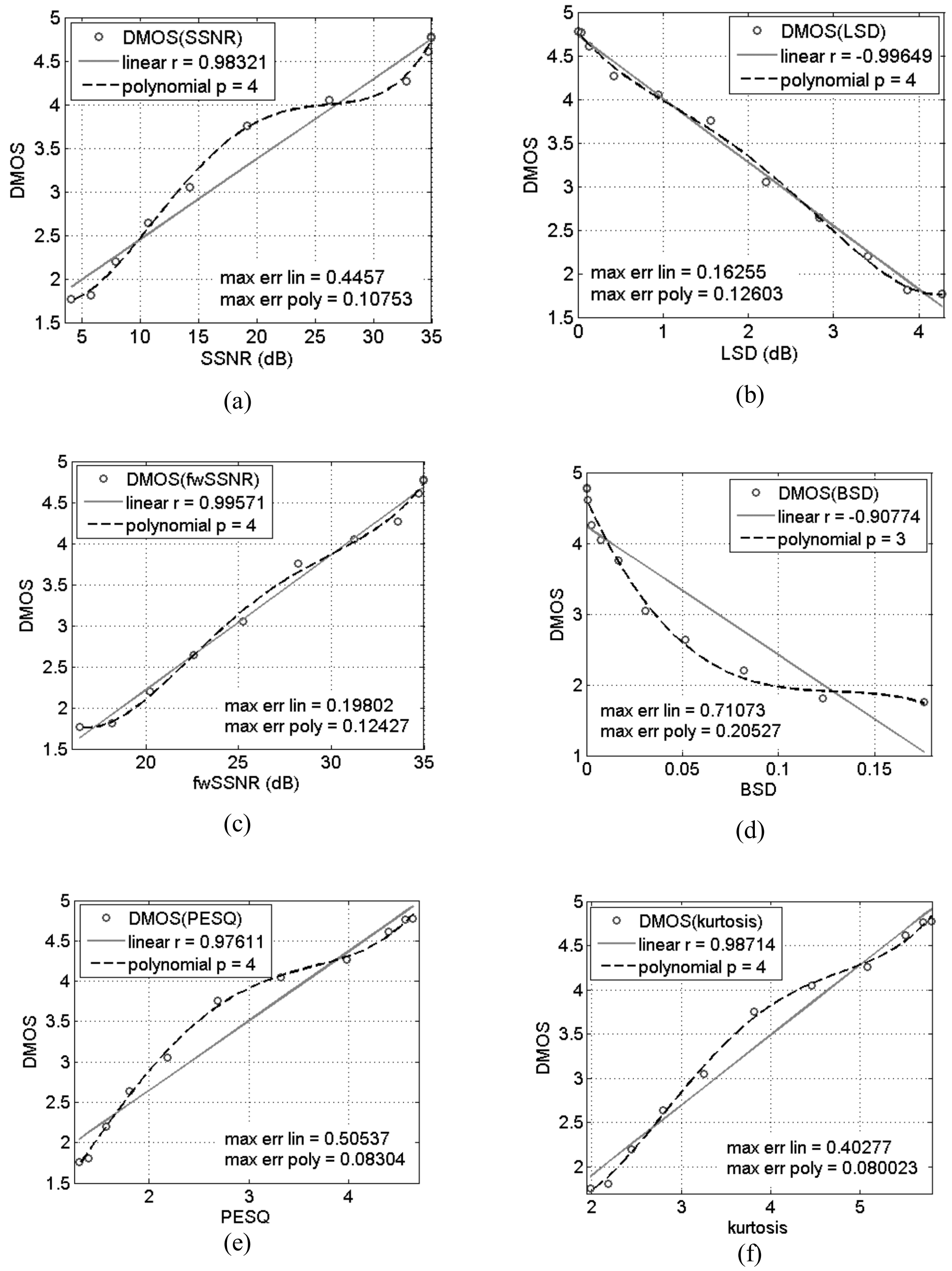


Fig. 3. Matching maps: SSNR (a); LSD (b); fwSNR (c); BSD (d); PESQ (e); kurtosis (f)

VI. CONCLUSION

The case of known undistorted signal is much simpler, from the point of view of measuring the quality of the distorted signal, in comparison with the case where the undistorted signal is unknown. Nevertheless, a reliable solution to this problem should contribute to improving the performance of UAVs voice control systems.

It was shown in this paper that subjective, on DMOS scale, estimates of clipped speech quality decreases monotonically when clipping degree is increasing. DMOS values are about 4.5, 3.5 and 2.5 for degree of clipping 5 dB, 10 dB and 15 dB, respectively. This rule allows establishing the boundary permissible degree of clipping when demands to the speech quality are established.

Dependencies of objective quality measures of a speech signals on their clipping degree are presented. Correlation coefficients and matching maps between objective and subjective speech quality measures have been calculated. The obtained results allow conclude that objective speech quality measures, such as SSNR, fwSNR, LSD, BSD, PESQ, and kurtosis can be applied to evaluate both the clipped speech quality and the degree of clipping. Moreover, kurtosis measure doesn't require that the original undistorted signal be known.

Given this circumstance, in the future it would be necessary to study more thoroughly the potential possibilities of the kurtosis as a clipped speech quality measure.

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А. М. Продеус, І. В. Котвицький, А. А. Дітяшов. Оцінювання якості кліпованої мови

Кліпування мови призводить до появи гармонік вищих порядків і, як наслідок, до зменшення точності автоматичних систем розпізнавання мови, що використовуються як різновид штучного інтелекту систем керування повітряними судами та систем управління польотом для безпілотних літальних апаратів. У даній роботі представлено суб'єктивні та об'єктивні оцінки якості кліпованої мови. Показано, що значення суб’єктивних оцінок якості мовлення становлять 4,5, 3,5 та 2,5 для ступенів кліпування 5 дБ, 10 дБ та 15 дБ, відповідно. Встановлення цієї закономірності дозволяє визначити гранично допустиму ступінь кліпування на основі заданих вимог до якості мовлення. Отримано залежності об'єктивних показників якості мовлення, таких як сегментне відношення сигнал-шум, частотно-зважене сегментне відношення сигнал-шум, логарифмічне спектральне спотворення, барк-спектральне спотворення та перцептуальна оцінка якості мовлення від ступеню кліпування. Показано, що коефіцієнт ексцесу також можна використовувати як міру якості мовленнєвого сигналу. Розраховано коефіцієнти кореляції та карти відповідності, що встановлюють зв'язок між об'єктивними та суб'єктивними оцінками якості мовлення. Отримані результати дозволяють зробити висновок, що об'єктивні міри якості мови можна застосовувати для оцінювання якості кліпованої мови та ступеня кліпування мовних сигналів.

Ключові слова: якість кліпованого мовного сигналу; об'єктивна міра; суб'єктивна міра; карта відповідності; коефіцієнт кореляції.

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А. Н. Продеус, И. В. Котвицкий, А. А. Дитяшов. Оценивание качества клиппированной речи

Клиппирование речи приводит к появлению гармоник высших порядков и, как следствие, к уменьшению точности автоматических систем распознавания речи, использующихся как разновидность искусственного интеллекта систем управления воздушными судами и систем управления полетом беспилотных летательных аппаратов. В данной работе представлены субъективные и объективные оценки качества клиппированной речи. Показано, что значения субъективных оценок качества речи составляют 4,5, 3,5 и 2,5 для степеней клиппирования 5 дБ, 10 дБ и 15 дБ, соответственно. Установление этой закономерности позволяет определить предельно допустимую степень клиппирования на основе заданных требований к качеству речи. Получены зависимости объективных показателей качества речи, таких как сегментное отношение сигнал-шум, частотно-взвешенное сегментное отношение сигнал-шум, логарифмическое спектральное искажение, барк-спектральное искажение и перцептуальная оценка качества речи от степени клиппирования. Было показано, что коэффициент эксцесса также можно использовать как меру качества речевого сигнала. Рассчитаны коэффициенты корреляции и карты соответствия, устанавливающие связь между объективными и субъективными оценками качества речи. Полученные результаты позволяют сделать вывод, что объективные меры качества языка можно применять для оценивания качества клиппированной речи и степени клиппирования речевых сигналов.

Ключевые слова: качество клиппированного речевого сигнала; объективная мера; субъективная мера; карта соответствия; коэффициент корреляции.

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