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Quality Assessment of Speech Signals Under a Process of Echo Cancelation in Telecommunications Systems

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Abstract The phenomenon of echo in the telecommunications channels is caused by the reflection of an electrical signal in a long line. In order to improve the quality of the transmitted sound, various adaptive filters are used to remove or at least reduce the level of the reflected delayed signal. However, such a process may result in a degradation in the quality of speech, although its intelligibility may not get worse. The work presents the results of subjective studies on assessing the quality of speech signals under the process of acoustic echo cancellation using different algorithms. The algorithms studied were: LMS (Least Mean Squares), NLMS (Normalized Least Mean Squares) and AP (Affine Projection). The study consisted of assessing the signal quality after applying the echo elimination process using the Degradation Category Rating method. A total of 312 signals were used in the test: 192 male speech and 120 female speech samples. Echo simulation was used using different delay times and levels of echo signal. Both types of speech have signal delay times of 20 ms, 50 ms, 100 ms and 200 ms with echo level values of -6 dB, -12 dB, -18 dB and -24 dB. In addition, for female speech signals, a delay time of 150 ms was introduced. The study involved 14 people aged between 18 and 38, including six women and eight men. All subjects had normal good hearing. Seven listeners had participated in subjective listening tests of the sound quality assessment previously. The listeners' opinions were collected on prepared questionnaire. It was found that the highest ratings were given to the AP filter, while the worst ratings were featured the NLMS. It should also be noted that the range between the results obtained for AP and NLMS for female speech is smaller in comparison to male. It is also interesting that the discrepancy in ratings was greatest for a delay time of 100 ms for the AP filter and 200 ms for the LMS filter. It can therefore be concluded from the obtained results that, in the case of acoustic echo cancelation, AP filter introduced the lowest quality degradation while the LMS achieved slightly worse average ratings when compared to the AP filter . The NLMS filter characterized by the worst ratings, and in some cases received twice the quality degradation compared to the AP filter.

Keywords: DSP, echo cancellation, adaptive filtration, sound perception.

1. Introduction

The echo impression is created when a reflected signal is delayed by at least 100 ms from the direct signal. In the case that the delay of the reflected signal is less than 100 ms, we are talking about reverberation and a listener is not able to distinguish the direct and reflected sound [1]. The phenomenon of echo in the telecommunication channels is caused by the reflection of an electrical signal in a long line and by the processing of sound in telecommunication systems with using hybrides.

The process of eliminating acoustic echo is used in areas where the human voice is used, where echo is not desirable, e.g. telephony. Echo-eliminating systems are used in hands-free mobile phone devices or hands-free devices, where quality and voice understanding play a very important role. In order to improve the quality of the transmitted sound, various adaptive filters are used to remove or at least reduce the level of the reflected delayed signal [2]. However, such a process may result in a degradation in the quality of speech, although its intelligibility may not get worse [2, 3].

The paper presents the results of subjective studies on assessing the quality of speech signals under process of acoustic echo cancellation using different algorithms. The algorithms used in our research were: Least Mean Squares (LMS), Normalized Least Mean Squares (NLMS) and Affine Projection (AP). The study consisted of assessing the subjective signal quality after applying these echo elimination processes.

2. Algorithms of echo cancellation

The application and development of modern technologies results in new and better and faster methods of communication. In recent years, the number of telephone calls made using hands-free sets has increased exponentially [3]. The number of teleconferences and videoconferences is also growing extremely rapidly. Due to the acoustic specificity of this form of communication, which results from poor acoustic isolation between the loudspeaker(s) and the microphone(s), there is a serious problem with acoustic echoes. While for most users of tele- and videoconferencing systems, the presence of their own, slightly delayed voice in the 'handset' is neutral or desirable, delays of several tens of milliseconds [1] make the conversation cumbersome and impossible for delays of about 250 ms [2]. For this reason, it seems extremely important to use an effective method of echo cancellation.

An adaptive filter is a digital filter in which coefficients are calculated by an algorithm to achieve a specific filtration effect. Such a filter is used in many areas where digital signal processing is used. In this case, adaptive filters are used to detect and then eliminate acoustic echo.

The most frequently used adaptive filtration algorithms became the LMS (Least Mean Squares) family. The original LMS algorithm was created by B. Widrow in the late 1950s at MIT [3] . There are different variants of the algorithms from the LMS family. One of them is the NLMS (Normalized Least Mean Square) variant (PNLMS – Proportionate Normalized Least Mean Square) with a time-varying μ-factor, whose value depends on the momentary signal energy. Standardized version of the LMS algorithm (NLMS) is today the most common algorithm used in echo cancelling devices. Since the 1990s there has been an increased interest in research on the problem of acoustic echo cancellation. This is primarily due to the increasing use of hands-free sets in mobile telephony. Teleconferences are also becoming increasingly popular. In both cases, the acoustic isolation between the loudspeaker and microphone is insufficient, which raises the problem of echo cancelling, which in this case has an acoustic basis. A general diagram of the process of echo cancellation including the possibility for both speakers to speak simultaneously is shown in Fig. 1. The speech signal of a distant speaker x reaches the room where the other speaker is located. The signal of the distant speaker is recorded by the microphone, also recording the speech of the near-end speaker $v(n)$ and the noise $w(n)$, where n is sample number (discrete time). In this type of echo cancellation system, there is a DTD (Double Talk Detector) block responsible for detecting situations when two people speak at the same time, as well as an adaptive block updating the echo cancelling filter coefficients of the far-end speaker signal $x(n)$.

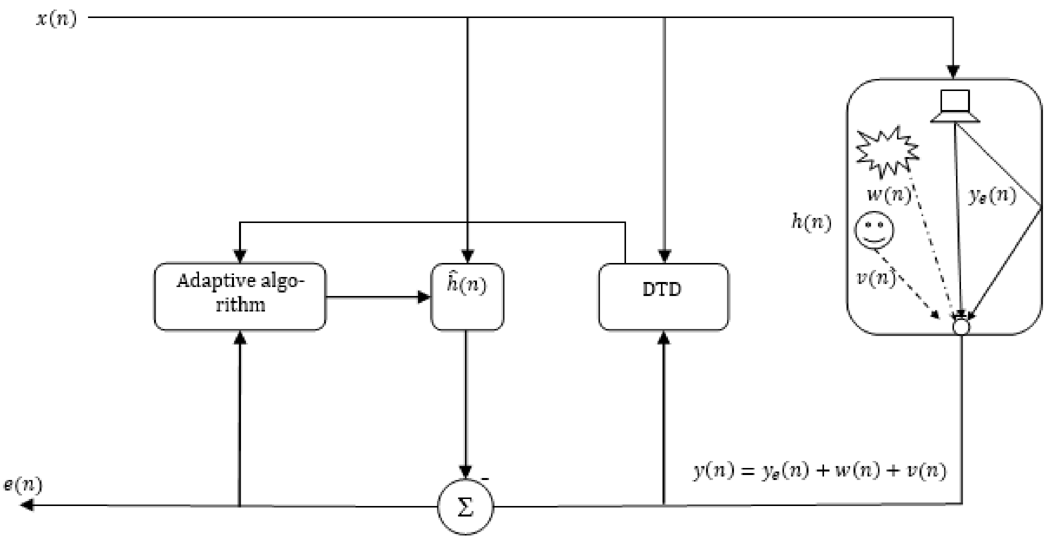

Figure 1. Block diagram of the acoustic echo cancellation system.

Adaptive filters that were used to remove echoes include Least Mean Squares (LMS), Normalized Least Mean Squares (NLMS), and Affine Projection (AP) [2, 3].

1. LMS (Least Mean Squares) is a class of adaptive filters used to mimic the desired filter and by finding the proper filter coefficients that refer to producing the smallest average square of the error signal, that is, the difference between the desired signal and the real signal.

2. Normalized Least Mean Squares (NLMS) is a standardized version of the LMS algorithm. Unlike the LMS algorithm, NLMS uses a variable step size parameter that is divided by the input vector power for each iteration.

3. AP (Affine Projection) is a kind of adaptive algorithm for filtering better convergence with correlated input signals, while providing a computational cost comparable to the LMS algorithm. Affine Projection is a generalization of NLMS algorithms in which projections are performed in multiple dimensions.

Filters from the LMS family are a group of gradient adaptive filters, realizing a strategy of minimizing the momentary (not expected) error value. Hence the cost function takes the form here:

$$J=J(n)=e^2(n), \tag{1}$$

instead of:

$$J = \mathrm{E}[e^2(n)], \tag{2}$$

where $e(n)$ is an error function that can be defined as:

$$e(n) = y(n) - \hat{y}(n). \tag{3}$$

If the echoes are eliminated, the vector $y(n)$ will simply be the signal that comes back to the first speaker. This signal can be presented as a superposition of three component signals according to the following relation:

$$y(n) = y_e(n) + v(n) + w(n), \tag{4}$$

where $y_e(n)$ is an echo of the first speaker, $v(n)$contains the speech signal of the second speaker and $w(n)$ is a signal containing noise. The $v(n)$ component occurs only if both sides of the conversation are speaking at the same time. This kind of communication is called DT (Double Talk). When both sides of the conversation speak at the same time, the correlation between the $x(n)$ and $y(n)$ signals is less than if only the first speaker is speaking. The task of Double Talk Detection (DTD) algorithms is to detect the simultaneous speech situation of both people. When this is detected, the adaptive filter coefficients are 'frozen'. The adaptation process is then continued after the second speaker has finished speaking.

It is assumed that the echo signal of the first person $y_e(n)$ can be presented as follows:

$$y_e(n) = \sum_{i=0}^{N-1} h_i\, x_i(n) = \boldsymbol{h}^T \boldsymbol{x}(n), \tag{5}$$

where N is size of the filter and h_i is the i-th coefficient of the filter.

However, the vector of coefficients $\boldsymbol{h}$ is unknown. Therefore, its estimation is determined by the following equation:

$$\hat{y}(n) = \hat{\boldsymbol{h}}^T \boldsymbol{x}(n), \tag{6}$$

where $\hat{y}(n)$ is an estimation of the echo signal of the first conversation participant.

Adaptive filtration involves the continuous adaptation of the $\hat{\boldsymbol{h}}$ vector to the $\boldsymbol{h}$ vector, so that with each subsequent iteration, the vector $\hat{\boldsymbol{h}}$ was getting closer to $\boldsymbol{h}$. However, as the $\boldsymbol{h}$ vector remains unknown, the matching of $\hat{\boldsymbol{h}}$ and $\boldsymbol{h}$ must be done indirectly. One of the matching methods is to minimize the error function described in formula (3).

In such a case, the adaptation of the $\hat{\boldsymbol{h}}$ coefficients is therefore such that the correction is proportional to the gradient of the cost function, but has the opposite sign. Hence the equation determining the update of the filter coefficients takes the following form:

$$\hat{\boldsymbol{h}}(n+1) = \hat{\boldsymbol{h}}(n) + \Delta\hat{\boldsymbol{h}}(n) = \hat{\boldsymbol{h}}(n) - \frac{\partial J(\hat{\boldsymbol{h}}(n))}{\partial \hat{\boldsymbol{h}}(n)}. \tag{7}$$

In many cases, equation (7) is expanded to form:

$$\hat{\boldsymbol{h}}(n+1) = \hat{\boldsymbol{h}}(n) - \frac{1}{2}\mu(n)\boldsymbol{W}(n)\frac{\partial J(\hat{\boldsymbol{h}}(n))}{\partial \hat{\boldsymbol{h}}(n)}, \tag{8}$$

where the time-varying scaling factor $\mu(n)$ determines the rate of change of the vector $\Delta\hat{\boldsymbol{h}}(n)$, while the $\boldsymbol{W}(n)$ matrix is responsible for the rate of adaptation

For the classic LMS filter, the $\boldsymbol{W}(n)$ weight matrix is replaced by the biased unit matrix $\mathbf{I}$:

$$\boldsymbol{W}(n) = \boldsymbol{I}, \tag{9}$$

while the scaling factor remains unchanged over time:

$$\mu(n) = \mu = const, \tag{10}$$

Finally, the equation determining the value of the filter coefficients in the next iteration takes the following form for the LMS algorithm:

$$\widehat{\boldsymbol{h}}(n+1) = \widehat{\boldsymbol{h}}(n) + \mu \boldsymbol{x}(n) e(n). \tag{11}$$

For a NLMS filter, the scaling factor value is variable over time and is given by the formula:

$$\mu(n) = \frac{\mu_0}{\boldsymbol{x}^T(n)\boldsymbol{x}(n)}. \tag{12}$$

After this change, the relationship determining the value of the filter weighting factors is given by the following relation:

$$\widehat{\boldsymbol{h}}(n+1) = \widehat{\boldsymbol{h}}(n) + \frac{\mu_0}{\boldsymbol{x}^T(n)\boldsymbol{x}(n)} \boldsymbol{x}(n) e(n), \tag{13}$$

The NLMS algorithm, despite its undoubted advantages, which include simplicity of implementation and low computational complexity (and thus the possibility of implementation on less powerful hardware platforms) also has a serious drawback. In the case of input signals, for which the covariance matrix is characterized by a large dynamics of eigenvalues, the convergence speed of the algorithm significantly decreases.

For AP algorithm input is created as follow:

$$\boldsymbol{X}_{AP}(n) = (x(n), \dots x(n-L), \tag{14}$$

where n is discrete time index, L is projection order and $\boldsymbol{X}_{AP}$ is filter input.

The filter output is given by formula and adaptation of filter coefficients are given as below:

$$\boldsymbol{y}_{AP}(n) = \boldsymbol{X}^T{}_{AP}(n)\boldsymbol{w}(n), \tag{15}$$

$$\boldsymbol{w}_{AP}(n+1) = \boldsymbol{w}_{AP}(n) + \mu \boldsymbol{X}_{AP}(n)(\boldsymbol{X}^T{}_{AP}(n)\boldsymbol{X}_{AP}(n) + \varepsilon \boldsymbol{I})^{-\mathbf{1}} \boldsymbol{e}_{AP}(n) \tag{16}$$

Where μ and ε are numerical coefficients, and $\boldsymbol{e}_{AP}(n)$ is error vector given as follow:

$$\boldsymbol{e}_{AP}(n) = \boldsymbol{d}_{AP}(n) - \boldsymbol{y}_{AP}(n). \tag{17}$$

The convergence of LMS, NLMS and AP algorithms has been studied many times by various researchers who confirm faster convergence of AP algorithms relative to NLMS algorithms, which in turn have faster convergence relative to LMS algorithms [4-6].

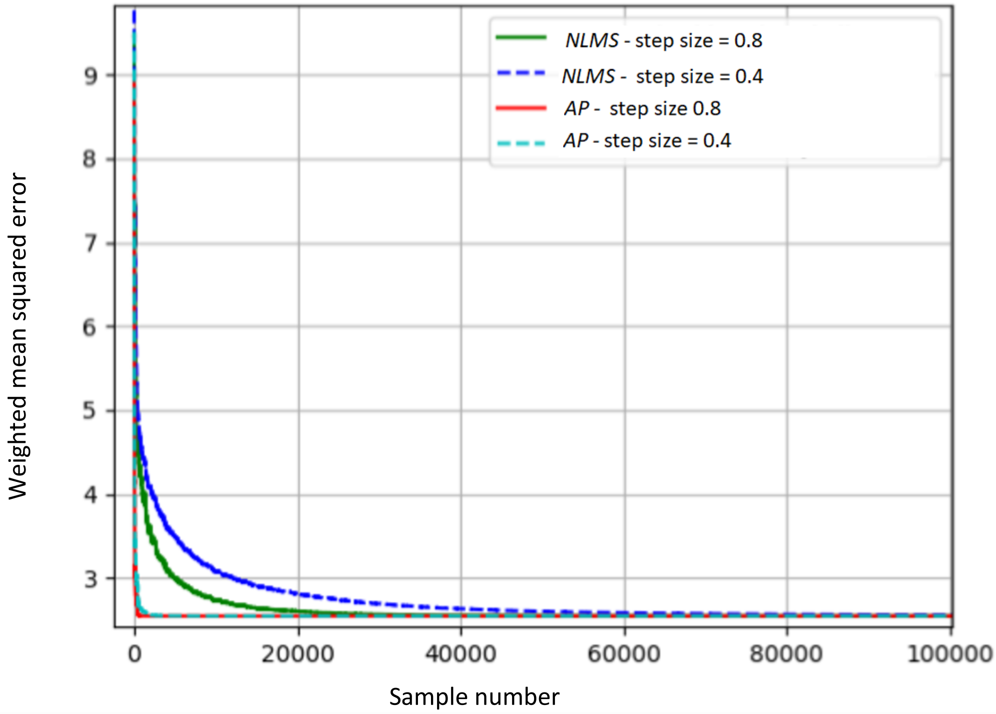

Figure 2. Comparison of the convergence of NLMS and AP algorithms [6].

The work presents the results of subjective studies on assessing the quality of speech signals under the process of acoustic echo cancellation using different algorithms. Experiment was done with a group of 14 listeners with speech sounds recorded by male as well as female voices.

3. Subjective experiment

3.1. Goal

The algorithms studied in the research were: Least Mean Squares (LMS), Normalized Least Mean Squares (NLMS) and Affine Projection (AP). The study consisted of assessing the signal quality after applying the echo elimination process using the Degradation Category Rating (DCR) method.

3.2. Stimuli

A total number of 312 signals were used in the test: 192 male speech and 120 female speech samples. Echo simulation was used using different delay times and levels of echo signal. Both types of speech have signal delay times of 20 ms, 50 ms, 100 ms and 200 ms with echo level values of -6 dB, -12 dB, -18 dB and -24 dB. For each algorithm, filters with lengths of 882, 2205, 4410, 8820, respectively, were used. The value of the step size parameter was set to 0.4.

All the signals used in the experiment were prepared as mono to prevent differences in judgement due to the influence of the spatial impression of the stimuli. The experiment used synthetically produced echo signals containing, in addition to the original signal, its single delayed (20ms, 50ms, 100ms and 200ms) repetition with reduced level (-6dB, -12dB, -18dB and -24dB). Multiple repetitions of the original signal were not simulated in the experiments performed.

3.3. Procedure

For speech quality assessment in this experiment the Degradation Category Rating test methodology defined by ITU [4] was used. The stimuli were presented to listeners by pairs (A-B) where A was the quality reference sample and B the same sample processed by the system under evaluation. Some "null pairs" (A-A), have been included to check the quality of anchoring. It can be noted that using a reference and subjective judgements with respect to that reference is quite a common procedure in psychoacoustics and it tends to result in a good sensitivity for the overall evaluation by listeners. Samples A and B were separated by 0.5 s break. The speech samples were rated using a five-step scale, from 1 to 5 in descriptive intervals: from "Degradation is inaudible" (5), through "Degradation is audible but not annoying" (4), "Degradation is slightly annoying" (3), "Degradation is annoying" (2) to "Degradation is very annoying "(1), according to the listeners' judgments. Test samples were presented with comparison to the pattern, and in the first interval the pattern was always presented while the compared sample was presented in the second interval. This method of presentation affords higher sensitivity of evaluation. The sentence lists of Polish language used in the subjective speech quality measurements were created in the Department of Acoustics, Multimedia and Signal Processing, at Wroclaw University of Science and Technology. Each list was divided into 10 groups, each with 5 tasks. The one listening session took no more than 25 minutes and a 15-minutes break occurred after this. The listeners' opinions were collected on the prepared questionnaire as a part of user's interface of the computer program.

3.4. Listeners and listening conditions

The study involved 14 people aged between 18 and 38, including six women and eight men. All subjects had normal good hearing which had been confirmed by the audiometric tests. It should be also noted that half of those listeners participated in various listening tests to assess the quality of musical signals and speech quality evaluation so one may say that they were experienced in the subjective research. The signals have been stored on a computer and presented with the use of a pair of active loudspeaker-boxes (TLC Pro-AMS 1). The listening sessions were provided in the recording studio of the Department of Acoustics, Multimedia and Sound Processing at Wroclaw University of Science and Technology which fits the recommended listening conditions [7, 8].

4. Results

The statistical treatment by the means of ANOVA testing with the statistical power set on the critical *p* value of 0.05 has shown that the results could be averaged for all listeners in a group referring to tested parameters. Moreover, the variances have been found homogeneous and confirmed by Bartlett test ($\chi^2 < \chi_\alpha^2 = 4.54$ and $\chi^2 < \chi_\alpha^2 = 5.38$ for male and female speech signals, respectively, at $\alpha = 0.05$). Thus, it allowed to average the results over the fourteen listeners in the testing group and their five repetitions. The results of the quality evaluation and their standard deviations for 95% confidence obtained with the use of DCR method for the distinguished echo reduction and delay-time for male speech are presented in Fig. 3 and for female speech in Fig. 4. The results of statistical treatment have indicated a significant effect of the algorithm used ($p < 0.008$ for male speech, and $p < 0.006$ for female speaking).

It was found that for both kinds of speech (male and female) and the short time of delay (20 ms and 50 ms) the highest ratings were given to the AP filter, while the worst ratings featured the NLMS. It should also be noted that the range between the results obtained for AP and NLMS for female speech is smaller in comparison to male one. It is also interesting that the discrepancy in ratings was greatest for a delay time of 100 ms for the AP filter and 200 ms for the LMS filter: for female speech the best rating were given to the LMS algorithm while for male speech the lowest quality degradation has been still observed for AP filtering.

It can therefore be concluded from the obtained results that, in the case of acoustic echo cancellation, the Affine Projection (AP) filter introduced the lowest quality degradation while the Least Mean Squares (LMS) achieved slightly worse average ratings when compared to the AP filter. The NLMS filter is characterized by the worst ratings of sound quality, and in some cases received almost twice the quality degradation compared to the AP filter, especially for female voice.

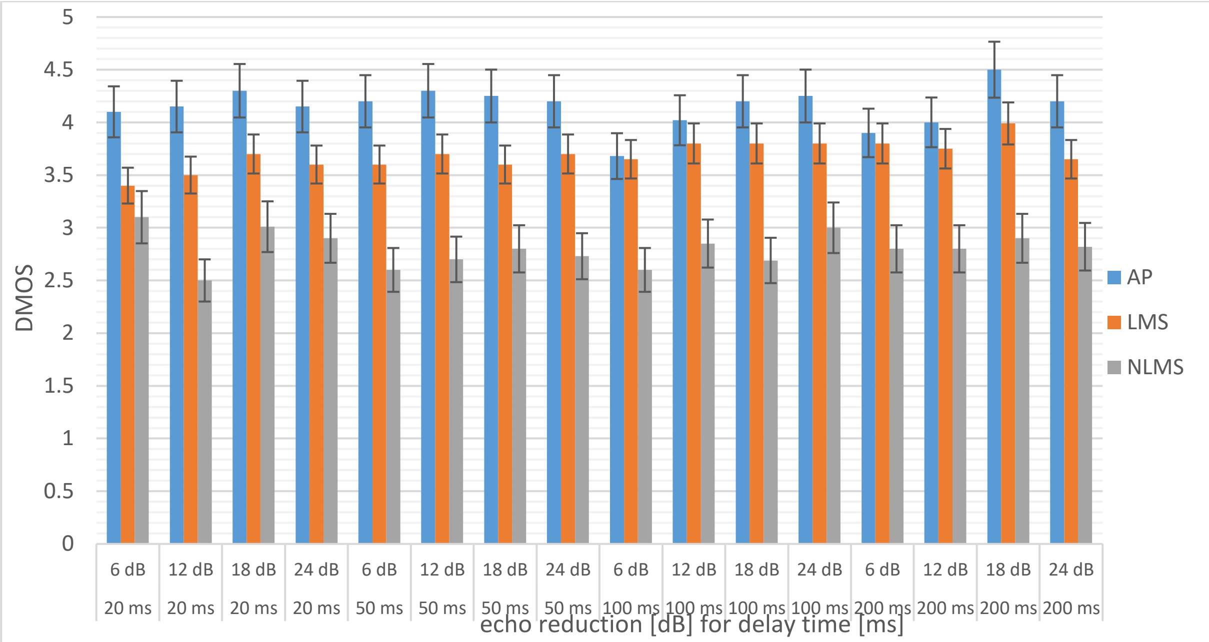

Figure 3. DMOS values for male speech obtained for various levels of echo reduction at particular delay time.

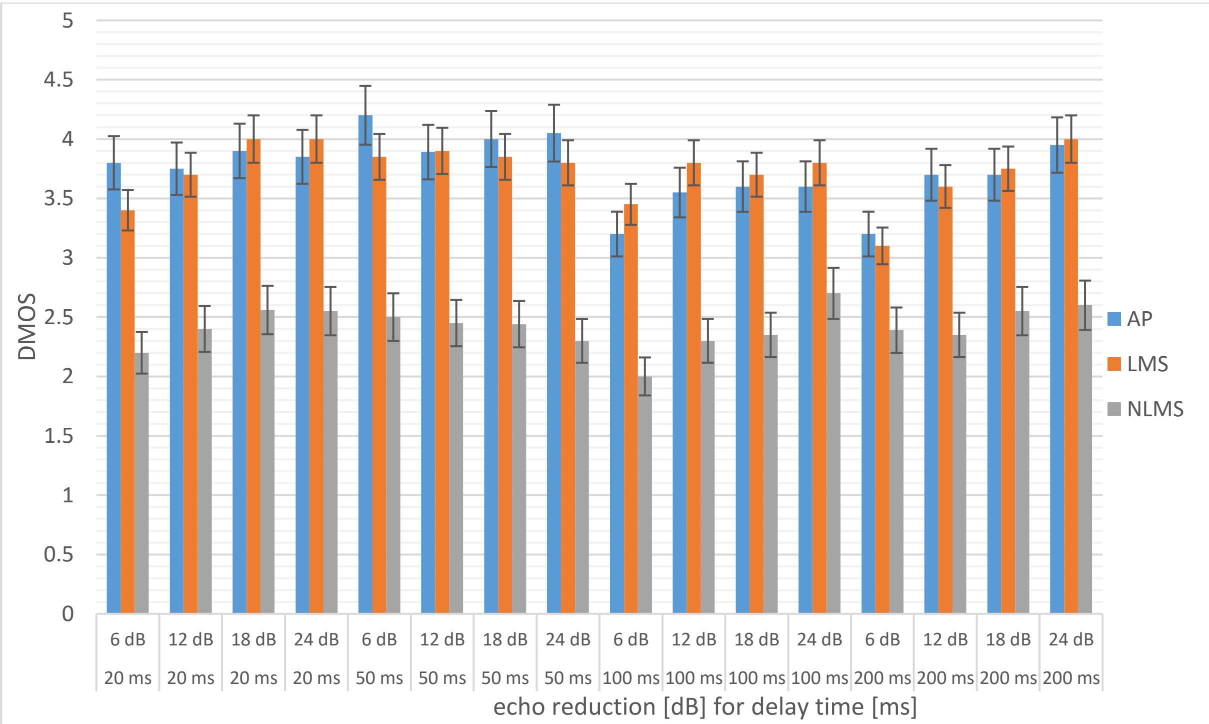

Figure 4. DMOS values for female speech obtained for various levels of echo reduction at particular delay time.

5. Discussion

The results of the tests indicate that the highest quality of speech signal can be achieved using the Affine Projection adaptive filter (AP). It proved to be the best in the echo from human speech signals cancellation process in comparison to LMS and NLMS. The lowest signal degradations have been observed at echo-delay-time of 200 ms and 18 dB reduction for male speakers, and 50 ms of delay-time and 6 dB reduction of echo for female talk. These values are 4.5 and 4.2, respectively, which means, according to the degradation scale, that observed degradation is practically inaudible. In listeners' opinions, the increase in quality degradation observed for female speech samples has been caused mainly by these noise-like signals, or “whispering”, introduced to the original voice after echo cancelation process, and of course the strongest disturbing components have been observed for Least Mean Squares filter. In the authors’ opinion, this fact may be connected to the spectrum of tested signals and their highest components appearing in the time periods referring to sibilants. Nevertheless, this did not significantly degraded the overall quality of the perceived signals. This may indicate a minor influence of the above-mentioned artifacts on the overall quality assessment of speech sounds, as this type of distortion does not cause a significant decrease in the intelligibility of the perceived sentences [9].

The worse speech quality has been obtained for the Normalized Least Mean Squares filtering and the DMOS values never reached the satisfactory quality level of 3.5 [10] for both kinds of speech. That means that this algorithm of echo cancellation is not recommended for the commercial applications. Moreover, the maximum step size of the NLMS algorithm is limited by the power of the reference signal vector, which scales directly by the number of coefficients [3].

The computational complexities of tested algorithms can be also the subject to compare. Let us denote the filter size as L and N as the number of dimensions of the AP algorithm, while k is a constant associated with the matrix inversion operation. As it is shown in Table 1. the AP algorithm has the highest computational complexity among the presented algorithms. In contrast, the classical LMS algorithm requires only $2L$ multiplications and L+1 summations every iteration.

Table 1. The computational complexity of the tested LMS, NLMS and AP algorithms.

Algorithm	Number of additions per iteration	Number of multiplications per iteration
LMS	L+1	$2L$
NLMS	$2L$+1	$3L$+50
AP	-	$2LN+kN^2$ (every N samples) and $2L+kN$ (for other iterations)

Comparing the computational complexity to the results of the subjective quality assessment, it can be seen that the most computationally complex process guarantees the least quality degradation for male speech. It should also be noted that this fact exists for all values of the tested delays and echo signal levels. For female speech, however, and for delay times of 50 ms and greater, the quality assessment results are similar for both: the least and the most computationally complex echo cancellation algorithm. Then, for commercial applications both these algorithms can be used.

6. Conclusions

If one assumed that the overall sound quality of speech signals may be assessed at satisfactory level, with DCR value of 3.5 [10], the required sound quality can be guaranteed in almost every tested condition with the Affine Projection adaptive filter, and for the Least Mean Squares filtering. In typical situations, speech-transmission networks are composed of a chain of various transmission, switching and terminal equipment. To investigate the quality of such networks, it is important to take the whole transmission channel into account. In more complex acoustic echo cancellation systems, non-recursive multi-pass algorithms can be used. In this case, increasingly better acoustic echo cancellation is achieved in each pass. However, this is paid with an increase in the number of performed operations [11-13].

The worse results of the NLMS algorithm in comparison to the LMS may be also caused by a non-optimal choice of the adaptation coefficient, whose value had been set at 0.8. In the authors’ opinion, the influence of the value of the adaptation coefficient on the quality assessment of speech signals under the process of echo cancellation is an interesting research problem for further investigation.

It should be also noted that the manner of speech signal perception is not fully understood and clarified. Such components as linguistic elements, the way of speaking and clarity and speech intelligibility may influence the overall evaluation of speech signal quality [14]. Since the major task of a telecommunication system is to transmit information and from this point of view, if the intelligibility of speech has been maintained at the required level, the degradation of the overall signal quality is of secondary importance. Therefore, when using certain echo cancellation algorithms, both the computational complexity and the degradation of the quality of the transmitted signals must be taken into account.

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