Proceedings of the International Multiconference on
Computer Science and Information Technology pp. 767–774

ISBN 978-83-60810-27-9
ISSN 1896-7094

Application of optimal settings of the LMS adaptive filter for speech signal processing

Jan Vaňuš
VŠB-Technical University of Ostrava, FEECS, Dept. of Electrical Engineering
17. listopadu 15, 70833 Ostrava-Poruba, Czech Republic
Email: jan.vanus@vsb.cz

Vítězslav Stýskala
VŠB-Technical University of Ostrava, FEECS, Dept. of Electrical Engineering
17. listopadu 15, 70833 Ostrava-Poruba, Czech Republic
Email: vitezslav.styskala@vsb.cz

***Abstract*–This paper describes a proposition of the method for optimal adjustment parameters of the adaptive filter with LMS algorithm in the practical application of suppression of additive noise in a speech signal for voice communication with the control system. By the proposed method, the optimal values of parameters of adaptive filter are calculated with guarantees the stability and convergence of the LMS algorithm. The DTW criterion is used for the quality assessment of speech signal processing obtained from output of adaptive filter with LMS algorithm. In the experimental section is described the way of verification of the proposed method on the structure of the adaptive filter with LMS algorithm and on the structure of the adaptive filter with LMS algorithm in application of suppressing noise from speech signal by simulations in MATLAB software and implementation on DSK TMS320C6713.**

***Keywords*–LMS adaptive filter (Least Mean Square), DTW criterion (Dynamic Time Warping), noise canceller.**

I. Introduction

FOR optimum settings of a step size parameter μ and the length M of the adaptive filter with the LMS algorithm is necessary ensuring the stability and convergence of the LMS algorithm. As a result of appropriate setting of the adaptive filter parameters is correct speech signal processing and subsequent correct the isolate words recognition through the use of the DTW criterion.

II. The Adaptive Filter With LMS Algorithm

A. LMS algorithm

Least mean – square (LMS) algorithm was developed by Widrow and Hoff in 1960. This algorithm is a member of stochastic gradient algorithms [2]. The LMS algorithm is a linear adaptive filtering algorithm, which, in general, consists of two basic processes:

a) filtering process, which involves

- Computing the output $y(n)$ of adaptive filter in response to vector input signal $\mathbf{x}(n)$ (1),
- Generating an estimation error $e(n)$ (Fig.5) by comparing this output $y(n)$ with desired response $d(n)$ (Fig.2) (2),

b) An adaptive process (3), which involves the automatic adjustment of the parameters $\mathbf{w}(n+1)$ of the filter in accordance with the estimation error $e(n)$ [3], [1]

$$y(n) = \mathbf{w}^{\mathrm{T}}(n)\,\mathbf{x}(n) \tag{1}$$

$\mathbf{w}(n)$ tap – weight vector,

$$e(n) = d(n) - y(n) \tag{2}$$

$$\mathbf{w}(n+1) = \mathbf{w}(n) + 2\mu e(n)\mathbf{x}(n) \tag{3}$$

$\mathbf{w}(n+1)$ tap – weight vector update,
μ step size parameter.

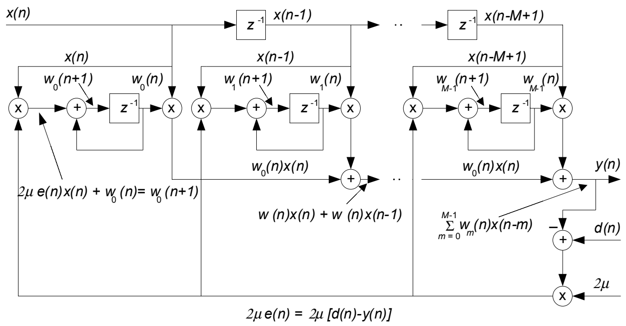

Fig. 1 FIR LMS adaptive filter realization [2].

B. Settings of a step size parameter μ

a) Calculating of a step size parameter μ to ensure the stability of adaptive filter with the LMS algorithm.

For determination, when the LMS algorithm remains stable is necessary find the upper bound of $\mu_{\max}$, that guarantees stability of LMS algorithm (4) [1]

$$\mu_{\max} < \frac{1}{3\,\mathrm{tr}[\mathbf{R}]} \tag{4}$$

tr[**R**] trace of **R**, which mean sum of the diagonal elements of **R**,
R Toeplitz autocorrelation matrix calculated from vector of input signal $\mathbf{x}(n)$, size **R** is M x M.

Toeplitz autocorrelation matrix **R** is calculated by equation (5)[2]

$$\mathbf{R} = \mathrm{E}[\mathbf{x}(n)\mathbf{x}^{T}(n)]\,. \tag{5}$$

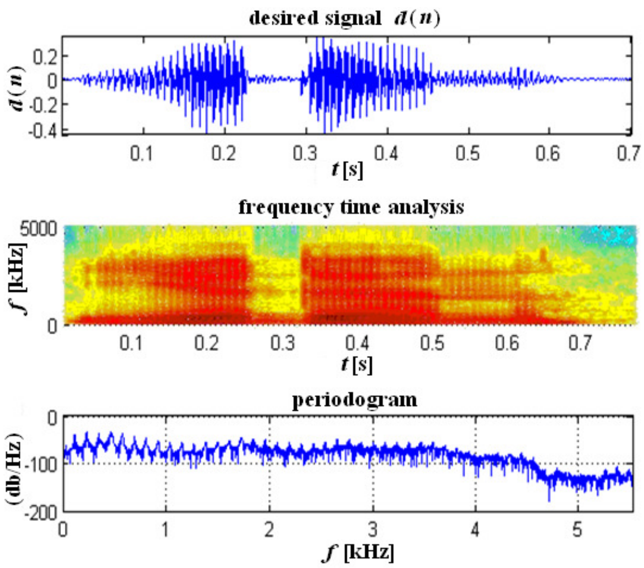

Fig. 2 Waveform, spectrogram (frequency time analysis) and periodogram of power spectral density estimate of desired speech signal $d(n)$ of isolated czech word "jeden" to the input of adaptive filter with LMS algorithm.

The significance of the upper bound of μ, which is provided by (4), is that it can easily be calculated from the filter input samples. Range of μ that is provided by (4) is sufficient for the stability the LMS algorithm, but is not necessary [1].

b) Calculating of a step size parameter μ to ensure the convergence of adaptive filter with the LMS algorithm

Convergence behaviour of LMS algorithm is directly linked to the eigenvalue spread of the autocorrelation matrix **R** and the power spectrum of $\mathbf{x}(n)$. Convergence of the LMS algorithm is directly related to the flatness in the spectral content of the underlying input process. E[$\mathbf{v}(n)$] converges to zero when μ remains within the range of formula (6). E[$\mathbf{v}(n)$] is expectation of weight – error vector $\mathbf{v}(n) = \mathbf{w}(n) - \mathbf{w}_0$.

$$\mu_{\text{conv}} \leq \frac{1}{\lambda_{\max}} \tag{6}$$

λ_{max} maximum eigenvalue of autocorrelation matrix **R** of the input vector $\mathbf{x}(n)$.

The above range does not necessarily guarantee stability of LMS algorithm. The convergence of LMS algorithm requires convergence of the mean of $\mathbf{w}(n)$ towards $\mathbf{w}_0$ and also convergence of the variance of the elements of $\mathbf{w}(n)$ to some limited values [1]. Vector $\mathbf{w}_0$ is calculated by Wiener – Hopf equation and the superscript "$_0$" indicates the optimum Wiener solution for the Wiener filter [2].

c) Calculating of optimal value of a step size parameter μ_{opt} of adaptive filter with the LMS algorithm

Determination of a step size parameter μ_{opt} value is important to conduct an algorithm LMS. When selecting parameter μ_{opt} terms of a compromise between the two aspects. On the one hand, large values μ can leads quickly to the optimal settings the LMS algorithm for speech signal processing. On the other hand, may increase the value μ of a mistake in the speech signal processing in further steps. Small value μ, on the contrary, ensure the stability and the convergence of LMS algorithm [1].

As a result small value μ is the slowdown in the convergence of LMS algorithm and, consequently, increasing the inaccuracies in the filtration non-stationary signals [9]. For the optimal value of the parameter μ_{opt} is the following equation (7) [1]

$$\mu_{\text{opt}} = \frac{M}{(1+M).\text{tr}[\mathbf{R}]}, \tag{7}$$

tr[**R**]trace of **R**, which mean sum of the diagonal elements of **R**, M parameter misadjustment.

Parameter misadjustment M is defined as ratio of the

Table I.

CALCULATED VALUES OF A STEP SIZE PARAMETERS μ_{opt}, μ_{max}, μ_{conv} OF LMS ADAPTIVE FILTER FOR INPUT SIGNAL $\mathbf{x}(n)$ WITH DIFFERENT SSNR VALUES.

$SSNR_a$ = 6,731(dB)	**$SSNR_{w1}$= 18,187(dB)**	**$SSNR_{w2}$= 3,119(dB)**	**$SSNR_{w3}$= –1,783(dB)**
μ_{max}=3,25.10^{-2} (M=10%)	μ_{max}=3,54.10^{-2} (M=10%)	μ_{max}=2,99.10^{-2} (M=10%)	μ_{max}=2,25.10^{-2} (M=10%)
μ_{conv}=1,21 (M=10%)	μ_{conv}=1,221 (M=10%)	μ_{conv}=1,239 (M=10%)	μ_{conv}=1,168 (M=10%)
μ_{opt}=5,9.10^{-3} (M=10%)	μ_{opt}=6,4.10^{-3} (M=10%)	μ_{opt}=5,4.10^{-3} (M=10%)	μ_{opt}=4,09.10^{-3} (M=10%)
μ_{opt}=1,08.10^{-2} (M=20%)	μ_{opt}=1,18.10^{-2} (M=20%)	μ_{opt}=1.10^{-2} (M=20%)	μ_{opt}=7,5.10^{-3} (M=20%)
μ_{opt}=14,99.10^{-3} (M=30%)	μ_{opt}=1,63.10^{-2} (M=30%)	μ_{opt}=1,38.10^{-2} (M=30%)	μ_{opt}=1,04.10^{-2} (M=30%)

steady – state value of the excess mean-square error (MSE) ξ_{excess} to the minimum mean square (MSE) error ξ_{min}.

$$M = \frac{\xi_{excess}}{\xi_{\min}} = \mu\,\text{tr}[\mathbf{R}]. \tag{8}$$

The misadjustment M is a dimensionless parameter that provides a measure of how close the LMS algorithm is to optimality in the mean - square – sense.

The smaller M is compared with unity, the more accurate is the adaptive filtering action being performed by the LMS algorithm. Values of misadjustment M are usually the 10%, 20% and 30% (Tab.I), (Tab.III), (Tab.V), (Tab.VI), (Tab.VII) [1].

A value of M = 10% means, that the adaptive system has an MSE only 10 percent greater than ξ_{min} [8].

III. Determination of the LMS adaptive filter length M by way of Widrow method [8]

Time constant τ_{mse} is calculating

$$\tau_{mse}=\frac{1}{4\mu\lambda}=\frac{M}{4\mu\,\mathrm{tr}[\mathbf{R}]}, \tag{9}$$

λ eigenvalue of autocorrelation matrix **R** of the input vector $\mathbf{x}(n)$,
tr[**R**]trace of **R**, which mean sum of the diagonal elements of **R**.

When the eigenvalues λ are sufficiently similar for the learning curve to be approximately, fitted by a single exponential, its time constant τ_{mse} may be applied to (9) to give an approximate value of M [8].

Values of order M LMS adaptive filter can be calculated from input signal $\mathbf{x}(n)$ to adaptive filter (Tab.III)

$$M=\frac{\mathrm{tr}[\mathbf{R}]}{\lambda}. \tag{10}$$

IV. Using DTW criterion for determination of the adaptive filter length M

The correct determination of the adaptive filter length M is very important. When the length M of the adaptive filter is low, the speech signal processing as a result of a small number of parameters of the adaptive filter is inaccurate. High value of the adaptive filter length M lead to inaccurate speech signal processing by influence of the estimator variance increase. In draft method in this work was used DTW criterion for determining value of length M of the LMS adaptive filter.

By way of DTW criterion is compare two sequences of vectors: reference vector **P** = [p(1), . . . p(P)] of the length P and test vector **O** = [o(1), . . . o(T)] of the length T [6].

Value of the LMS adaptive filter order M is determined by setting values of the order M in interval {0 to 150} and calculating of the minimum distance d (similarity) between the reference vector **P** (desired signal $d(n)$ (Fig.2)) and the test sequence vector **O** (error signal $e(n)$ (Fig.5)). Words are almost never represented by the sequence of the same length P$\neq$T. The distance d between the sequences **O** and **P** is given as minimum distance over the set of all possible paths (all possible lengths, all possible courses) [4]. When the distance d was **$d<0,2$**, the word **was recognized**. This value **$d<0,2$** was determined empirically from the measured results of implemented experiments (Tab.II), (Tab.III), (Tab.V), (Tab.VI), (Tab.VII), (Tab.VIII).

Minimum distance computation

$$D(\mathbf{O},\mathbf{P})=\min_{\{C\}} D_c(\mathbf{O},\mathbf{P}) \tag{11}$$

is simple, when normalization factor N_c is no function of path and is possible write $N_c=N$ for $\forall_c$

$$D(O,P)=\frac{1}{N}\min_{\{C\}}\sum_{k=1}^{Kc} d\left[o\left(t_c(k)\right),p\left(r_c(k)\right)\right]W_c(k) \tag{12}$$

V. Using of the additive noise in experiments with speech signal

For implementation of experiments are used additive noises with calculated segmental SNR (Signal to Noise Ratio) – SSNR (Tab.IV) for speech signal processing [5]

$$\mathrm{SSNR}=\frac{1}{K}\sum_{i=0}^{L-1}\mathrm{SNR}_i VAD_i, \tag{13}$$

L is the number of segments of speech signal,
K the number of segments in speech activity,
VAD_i is information about speech activity (values 1 and 0) in i-th segment, SNR_i is local (short term) SNR.

Table II.
The values of distance d are calculated by comparing of the Czech isolated words "jeden" (one) with the words "dva" (two) for up to "pět" (five) and compared the word "dva" (two) with the words "jeden"(one), "tři" (three) for up to "pět"(five).

jeden–jeden	jeden–dva	jeden–tři	jeden–čtyři	jeden–pět
$d=0$	$d=0{,}713$	$d=1{,}218$	$d=1{,}415$	$d=0{,}552$
dva–jeden	dva–dva	dva–tři	dva–čtyři	dva–pět
$d=0{,}713$	**$d=0$**	$d=0{,}406$	$d=0{,}568$	$d=0{,}373$

Table III.
The values of order M of adaptive filter and distance d between desired speech signal $d(n)$ to adaptive filter and error signal $e(n)$ from adaptive filter calculated by way of widrow methods (simulated in MATLAB).

	$\mathrm{SSNR_a}$=6,731(dB)	**SSNRw1=18,187(dB)**	**$\mathrm{SSNR_{w2}}$=3,119(dB)**	**$\mathrm{SSNR_{w3}}$=–1,783(dB)**
Widrow $\mathbf{M}$=10 %	μ_1=5,9.10^{-3}; **M=19** **d=9,919.10^{-2}**	μ_1=6,4.10^{-3}; **M=17** **d=1,697.10^{-1}**	μ_1=5,4.10^{-3}; **M=21** d=2,919.10^{-1}	μ_1=4,09.10^{-3}; **M=26** d=2,908.10^{-1}
Widrow $\mathbf{M}$=20 %	μ_2=1,08.10^{-2}; **M=19** **d=1,181.10^{-1}**	μ_2=1,18.10^{-2}; **M=17** d=2,54.10^{-1}	μ_2=1.10^{-2}; **M=21** d=4,152.10^{-1}	μ_2=7,5.10^{-3}; **M=26** d=4,011.10^{-1}
Widrow $\mathbf{M}$=30 %	μ_3=14,996.10^{-3}; **M=19** **d=1,357.10^{-1}**	μ_3=1,63.10^{-2}; **M=17** d=3,194.10^{-1}	μ_3=1,38.10^{-2}; **M=21** d=4,907.10^{-1}	μ_3=1,04.10^{-2}; **M=26** d=4,645.10^{-1}

In the paper were used following additive noises for compare of the results of the proposed methods:

- Additive noise $n_a(n)$ (Fig.3) with the ratio of the desired signal $d(n)$ (Fig.2) of isolated czech words "jeden" to noise $n_a(n)$ **SSNR$_a$=6,731(dB)** (Tab.IV).
- White noise $n_{w1}(n)$ with the ratio of the desired signal $d(n)$ (Fig.2) of isolated czech words "jeden" to noise $n_{w1}(n)$ **SSNR$_{w1}$=18,187(dB)** (Tab.IV).
- White noise $n_{w2}(n)$ the ratio of the speech signal of isolated words "jeden" (Fig.2) to noise $n_{w2}(n)$ **SSNR$_{w2}$=3,119(dB)** (Tab.IV).

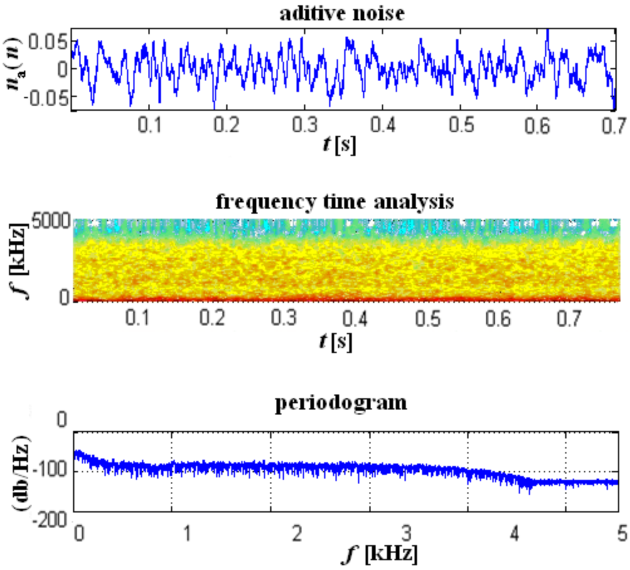

Fig. 3 Waveform, spectrogram (frequency time analysis) and periodogram of power spectral density estimate of desired speech signal $d(n)$ of additive noise $n_a(n)$ with SSNR$_a$=6,731(dB).

TABLE IV.
SEGMENTAL SIGNAL TO NOISE RATIO VALUES CALCULATED FOR THE SPEECH SIGNAL WORD "JEDEN" (FIG.1) TO ADDITIVE NOISE AND TO ADDITIVE WHITE NOISE.

Noise signification	Calculated values of ratio signal to noise
additive noise $n_a(n)$	SSNR$_a$=6,731(dB)
additive white noise 1 $n_{w1}(n)$	SSNR$_{w1}$=18,187(dB)
additive white noise 2 $n_{w2}(n)$	SSNR$_{w2}$=3,119(dB)
additive white noise 3 $n_{w3}(n)$	SSNR$_{w3}$= –1,783(dB)

- White noise $n_{w3}(n)$ the ratio of the speech signal of isolated words "jeden" (Fig.2) to noise $n_{w3}(n)$ **SSNR$_{w3}$=-1,783(dB)** (Tab.IV).

VI. DRAFT DTW METHOD

Draft method for optimal adjustment of a step size parameter μ_{opt} and the length M of the LMS adaptive filter was applied in next steps [7]:

1. Calculation of a step size parameter μ_{opt} optimal value (7) from input signal $x(n)$ (with SSNR) to the LMS adaptive filter (M=10%, M=20%, M=30%) (Tab.I).
2. For reference vector **P** is used desired signal $d(n)$ (Fig. 2) to the LMS adaptive filter.
3. As a test vector **O** was chosen error signal $e(n)$ (Fig.5).
4. Next was calculated the distance d (11), (12) between the signals $d(n)$ and $e(n)$ for sets values of LMS adaptive filter lengths M in interval {1 to 150}.
5. As the optimal value of the LMS adaptive filter order M was chosen value of the adaptive filter length M for minimum distance d (Fig.4) between two compared signals $d(n)$ and $e(n)$.

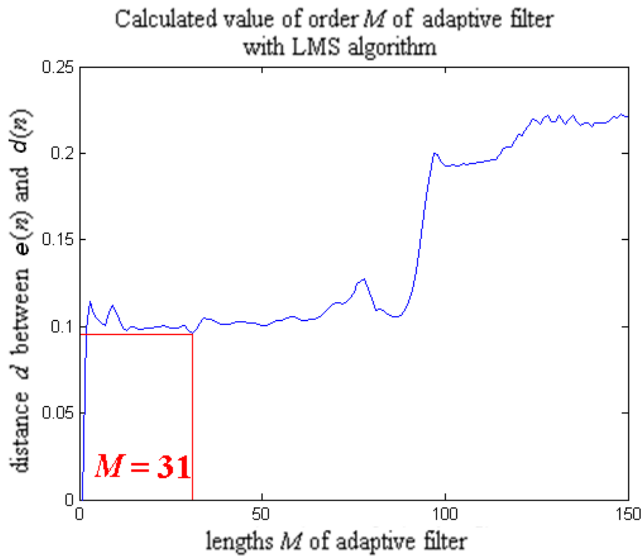

Fig. 4 Calculated values M=31 and d=9,61.10^{-2} of the LMS adaptive filter (μ=5,9.10^{-3}, SSNR=6,731(dB), M =10%).

TABLE V.
THE VALUES OF ORDER M OF ADAPTIVE FILTER AND DISTANCE d BETWEEN DESIRED SPEECH SIGNAL $d(n)$ TO LMS ADAPTIVE FILTER AND ERROR SIGNAL $e(n)$ FROM LMS ADAPTIVE FILTER CALCULATED BY WAY OF DRAFT METHOD WITH DTW CRITERION (SIMULATED IN MATLAB).

	SSNR$_a$=6,731(dB)	**SSNR$_{w1}$=18,187(dB)**	**SSNR$_{w2}$=3,119(dB)**	**SSNR$_{w3}$=–1,783(dB)**
M=10 %	μ_1=5,9.10^{-3}; ***M*=31** ***d*=9,61.10^{-2}**	μ_1=6,4.10^{-3}; ***M*=16** ***d*=1,691.10^{-1}**	μ_1=5,4.10^{-3}; ***M*=17** d=2,902.10^{-1}	μ_1=4,09.10^{-3}; ***M*=88** d=2,784.10^{-1}
M=20 %	μ_2=1,08.10^{-2}; ***M*=12** ***d*=1,124.10^{-1}**	μ_2=1,18.10^{-2}; ***M*=16** d=2,517.10^{-1}	μ_2=1.10^{-2}; ***M*=107** d=3,996.10^{-1}	μ_2=7,5.10^{-3}; ***M*=88** d=3,877.10^{-1}
M=30 %	μ_3=14,996.10^{-3}; ***M*=12** ***d*=1,249.10^{-1}**	μ_3=1,63.10^{-2}; ***M*=37** d=3,142.10^{-1}	μ_3=1,38.10^{-2}; ***M*=107** d=4,614.10^{-1}	μ_3=1,04.10^{-2}; ***M*=88** d=4,463.10^{-1}

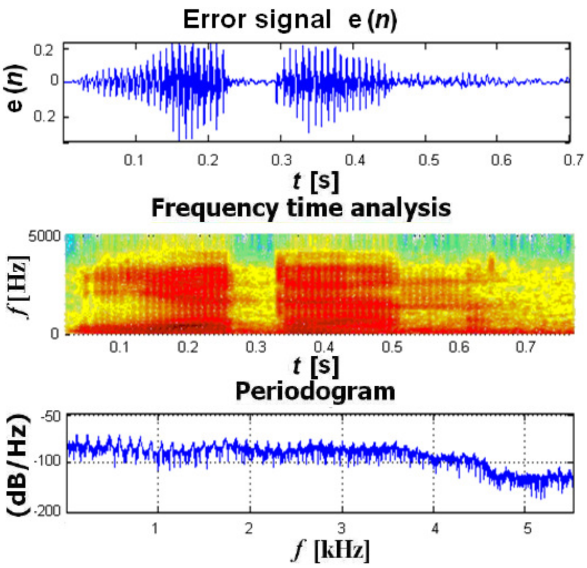

Fig. 5 Waveform, spectrogram (frequency time analysis) and periodogram of power spectral density estimate of error signal $e(n)$ (LMS adaptive filter – first iteration, M=31, μ=5,9.10^{-3}, simulated in MATLAB) [7].

In Table V can be seen, that the speech signal $e(n)$ at the output of the LMS adaptive filter **<u>was recognized</u>** (**d<0,2**) from first iteration for SSNR$_a$=6,731(dB) (μ_1=5,9.10^{-3}; M=31, M=10%), (μ_2=1,08.10^{-2}, M=12, M=20%), (μ_3=14,996.10^{-3}, M=12, M=30%) and for SSNR$_{w1}$=18,187(dB) (μ_1=6,4.10^{-3}; M=16, M=10%). When the additive noise values SSNR$_w$ in speech signal were higher, the speech signal was not recognized.

VII. Using of the Draft Method with DTW criterion for LMS adaptive Noise Canceling from speech signal

B. Matlab simulation

The draft method with DTW criterion was used for the LMS adaptive noise canceling from speech signal, simulated in MATLAB in two channel structure of the adaptive filter with LMS algorithm in an application for the suppression of additive noise (Fig.6). A primary input contains desired signal $d(n)$, and an additive noise $n(n)$. A noise reference input is assumed to be available containing $n''(n)$, which is correlated with the original corrupting noise $n(n)$. As shown figure 6 the LMS adaptive filter receives the reference noise, filters it, and subtracts the result from the primary input. From the point of view of the adaptive filter, the primary input ($d(n)$+$n(n)$) acts as its desired response and the system output acts as its error. The noise canceller output $e(n)$ (Fig.7) is obtained by subtracting the filtered reference noise $n(n)$ from the primary input. Adaptive noise canceling generally performs better, than the classical approach since the noise is subtracted out rather than filtered out [8].

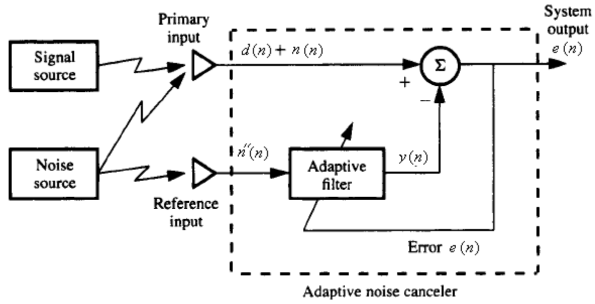

Fig. 6 Separation of signal $d(n)$ and noise $n(n)$ LMS adaptive noise-canceling approach [8].

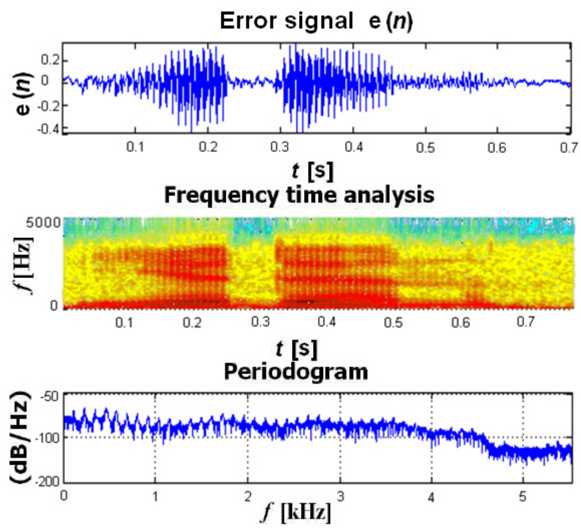

Fig. 7 Waveform, spectrogram (frequency time analysis) and periodogram of power spectral density estimate of error signal $e(n)$ (LMS adaptive noise canceller – first iteration, M=17, μ=5,9.10^{-3}, simulated in MATLAB) [7].

The draft DTW method was used for optimal settings values of the adaptive filter length M and a step size factor μ of

Table VI.
The optimal values of order M of filter and distance d between desired speech signal $d(n)$ and error signal $e(n)$ from LMS adaptive Noise canceller calculated by way of draft method with DTW criterion (simulated in MATLAB).

	SSNR$_a$=6,731(dB)	**SSNR$_{w1}$=18,187(dB)**	**SSNR$_{w2}$=3,119(dB)**	**SSNR$_{w3}$=–1,783(dB)**
M=10 %	μ_1=5,9.10^{-3}; ***M*=17** **<u>*d*=9,9.10^{-2}</u>**	μ_1=6,4.10^{-3}; ***M*=43** ***d*=5,421.10^{-1}**	μ_1=5,4.10^{-3}; ***M*=149** ***d*=9,142.10^{-1}**	μ_1=4,09.10^{-3}; ***M*=74** ***d*=1,473**
M =20 %	μ_2=1,08.10^{-2}; ***M*=10** **<u>*d*=9,8.10^{-2}</u>**	μ_2=1,18.10^{-2}; ***M*=99** ***d*=5,409.10^{-1}**	μ_2=1.10^{-2}; ***M*=103** ***d*=1,127**	μ_2=7,5.10^{-3}; ***M*=74** ***d*=1,42**
M =30 %	μ_3=14,996.10^{-3}; ***M*=7** **<u>*d*=9,87.10^{-2}</u>**	μ_3=1,63.10^{-2}; ***M*=99** ***d*=5,405.10^{-1}**	μ_3=1,38.10^{-2}; ***M*=103** ***d*=1,111**	μ_3=1,04.10^{-2}; ***M*=74** ***d*=1,374**

the adaptive filter with LMS algorithm in the application of the suppression of additive noise from the speech signal. Calculated optimal values – the order M of the LMS adaptive noise canceller and distance d between desired speech signal $d(n)$ (Fig.2) and error signal $e(n)$ (Fig.7) from the LMS adaptive noise canceller are calculated in Table VI. The speech error signal $e(n)$ from the output of LMS adaptive noise canceller **was recognized** (**d<0,2**) from first iteration only for the $SSNR_a$=6,731(dB) (μ_1=5,9.10^{-3}; M=17, M =10%), (μ_2=1,08.10^{-2}, M=10, M =20%), (μ_3=14,996.10^{-3}, M=7, M =30%).

When additive noise values $SSNR_w$ in speech signal are higher, speech signal was not recognized.

C. Implementation LMS adaptive noise canceller on DSK TMS 320C6713

The draft method with DTW criterion for determining of the order M of the adaptive filter with LMS algorithm was used in an application to suppressing noise $n(n)$ from the speech signal $x(n)$ in implementation of two channel structure of LMS adaptive noise canceller on DSP (Digital Signal Processor) Starter Kit (DSK) TMS320C67113 (Fig.10) [11].

Input signal $x(n)$ (Fig.8) is composed from desired signal $d(n)$ (Fig.2) + additive noise $n(n)$. The segmental signal to noise ratio of input signal $x(n)$ (Fig.8) is SSNR=6,676(dB).

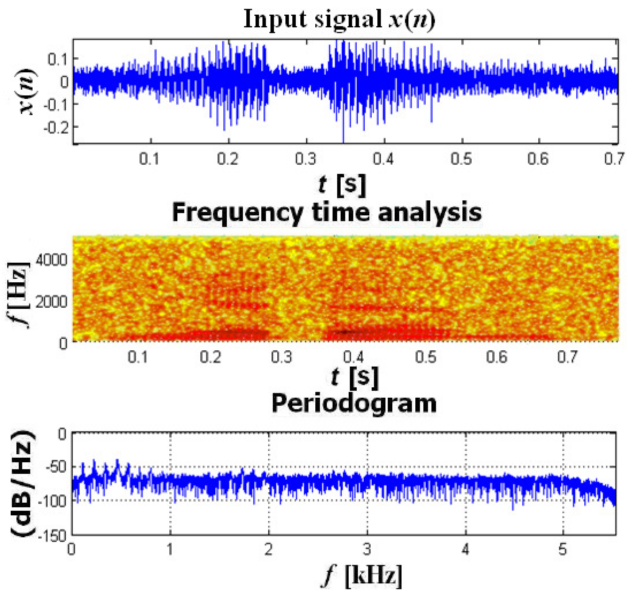

Fig. 8 Waveform, spectrogram (frequency time analysis) and periodogram of power spectral density estimate of input signal $x(n)$ (desired signal $d(n)$ + additive noise $n(n)$ - SSNR=6,676(dB)) to LMS adaptive filter in an application to suppress noise from the speech signal – first iteration, implemented on DSK TMS320C6713) [7].

Applications of the draft method with DTW criterion was carried out in several steps:

1. step - calculation of a step size parameters μ (M =10%, M =20%, M =30%) (Tab.VII) with guarantees the stability and convergence of the LMS algorithm by using of input signal $x(n)$ (Fig.8) SSNR=6,6756(dB) to LMS adaptive noise canceller (simulated in MATLAB).

2. step - the calculation values of the LMS adaptive filter order M (Tab.VII) for sets a step size parameters μ (M =10%, M =20%, M =30%) by using of input signal $x(n)$ (Fig.8) SSNR=6,6756(dB) to the LMS adaptive noise canceller (simulated in MATLAB).

In the Table VII are calculated values d between output signal $e(n)$ (Fig.9) from adaptive filter with LMS algorithm and desired signal $d(n)$ (Fig.2) for the values parameters set of the LMS adaptive filter order M.

The calculated values of distance d (Tab.VII) in MATLAB shows, that an isolated word "jeden" (Fig.9) from adaptive filter output **was recognized** **d=0,184** (**d<0,2**), when optimal set parameters of adaptive filter with LMS algorithm are M=21, μ_1=0,103 for M =10%.

TABLE VII.
THE CALCULATION VALUES OF DISTANCE d, LENGTH M AND A STEP SIZE PARAMETER μ OF THE LMS ADAPTIVE NOISE CANCELLER FOR (M =10%, M =20%, M =30%) SSNR=6,676(dB), (SIMULATED IN MATLAB).

M	M =10%	M =20%	M =30%
μ	μ_1=0,103	μ_2=0,188	μ_3=0,26
M	**M=21**	M=40	M=99
D	**d=0,184**	d=0,265	d=0,307

3. step - empirically was found, that the parameter μ for the LMS adaptive noise canceller, implemented on the DSK TMS320C6713 allow set only in the range μ=1.10^{-8} to μ=1.10^{-12} . The length M of the LMS adaptive noise canceller can be set only in the range M=16 to M=52.

Optimal settings values of a parameter μ and the order M was M=21 and μ=1.10^{-8} for the LMS adaptive noise canceller implemented on the DSK TMS320C6713 (Tab.VIII).

The speech error signal $e(n)$ from the output of LMS adaptive noise canceller (implemented on DSK TMS320C6713) **was recognized** (**d<0,2**) from first iteration for the $SSNR_a$=6,676(dB) (μ=1.10^{-8}; M=21).

TABLE VIII.
THE CALCULATION VALUES OF DISTANCE d FOR SETTINGS OF LENGTH **M=21** AND PARAMETERS μ ($x(n)$ (Fig.8) WITH SSNR=6,676(dB)) (LMS ADAPTIVE NOISE CANCELLER WAS IMPLEMENTED ON DSK TMS320C6713).

Settings of parameter μ	μ=1.10^{-12}	μ=1.10^{-10}	μ=1.10^{-8}
Calculated value of d	d=4,266.10^{-1}	d=3,475.10^{-1}	**d=8,97.10^{-2}**

VIII. USING LMS ADAPTIVE NOISE CANCELLER IN VOICE COMMUNICATION WITH CONTROL BUS SYSTEM NIKOBUS

The draft method with DTW criterion was used for optimal settings parameters of LMS adaptive noise canceller, implemented on DSK TMS320C6713, applied in voice communications with control BUS system NIKOBUS (Fig.10). System NIKOBUS was implemented in simulation of visualization operational control of the technical features of the building through visualization software Promotic. For speech recognition in voice communication with control BUS system has been used software My Voice (Fig.10)

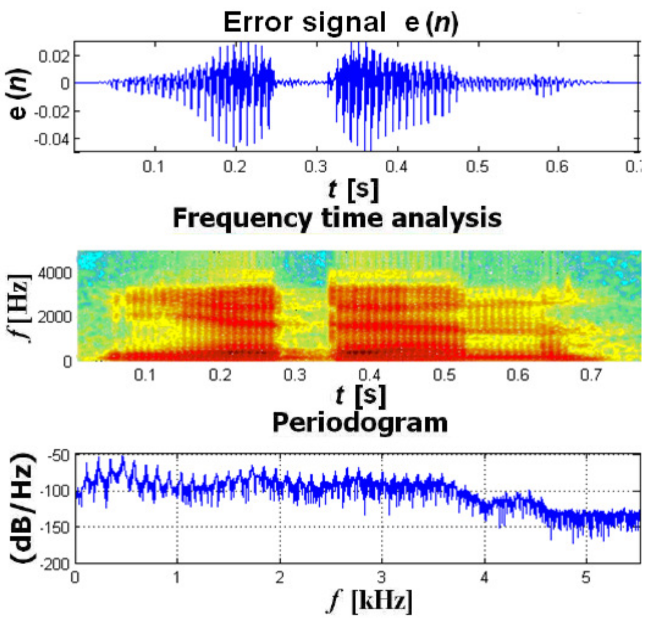

Fig. 9 Waveform, spectrogram (frequency time analysis) and periodogram of power spectral density estimate of error signal $e(n)$ (LMS adaptive Noise Canceling from the speech signal – first iteration, M=21, μ=1.10^{-9}, implemented on the DSK TMS320C6713) [7].

linked with software Promotic. By using of software My Voice is done voice control of operational technical functions in the buildings.

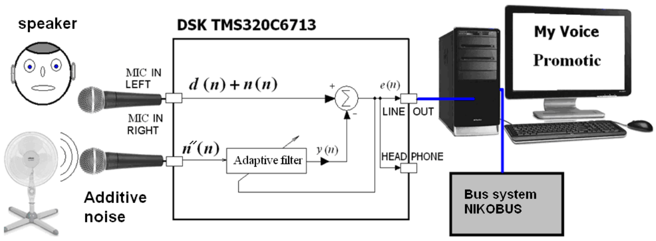

Fig. 10 Implementation of the LMS adaptive noise canceller for voice communications with control system NIKOBUS (Xcomfort) [7].

The aim of the experiment was to determine the success of the detection of selected voice commands. A microphone for capturing speech was located at a distance of about 5 cm from the mouth, according to the manufacturer's instructions. The second microphone was directed to the source of additive noise.

As source of additive noise were used the blower noise and loud radio on (radio station in D major, with classical music). The fan was placed in a distance of 25cm from the microphone. Radio speakers were placed approximately 70 cm from the microphone.

One order was one-word command "boiler" (wash-boiler) is for switching on and off the boiler.

Conditions for the experiment were the following (Fig. 11):

1. 100 x spoken command “boiler” without the LMS adaptive noise canceller:
• Measure 1 without additive noise – 99% successfully speech recognition.
• Measure 2 with additive noise – 81% successfully speech recognition.

2. 100 x spoken command “boiler” with the LMS adaptive noise canceller implemented on DSK TMS320C6713:
• Measure 3 without additive noise – 99% successfully speech recognition.
• Measurement of 4 with additive interference – 99% successfully speech recognition.

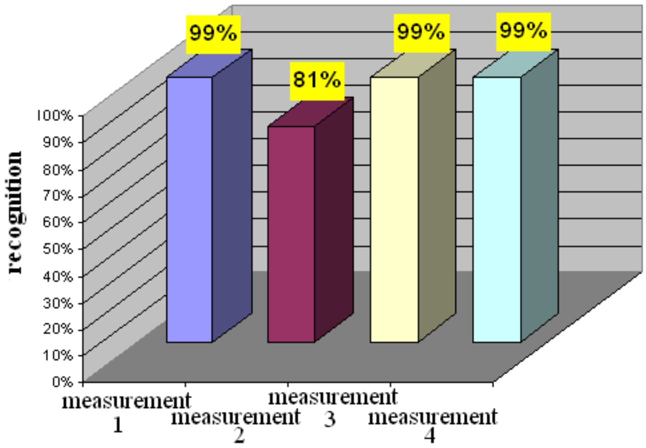

Fig. 11 Evaluation of recognition of isolated czech word - the command "boiler" (wash - boiler) by way of recognition software MyVoice.

Conclusion

In this paper was described the way of verification of the proposed method on the structure of adaptive filter with LMS algorithm in application of suppressing noise from speech signal by way of simulation in MATLAB software.

Through the use of DTW criterion was obtained a tool for determining the quality of speech signal processing using in optimal settings step size parameter μ and order M of the LMS adaptive filter or the LMS adaptive noise canceller

The proposed method was verified by way of the practical realization of the structure of the LMS adaptive noise canceller in the application for suppressing additive noise from speech signal by implementation on the DSK TMS320C6713 (Fig.12). This implementation was used for voice communication with control BUS system NIKOBUS for simulation controlling of operating technical functions in buildings.

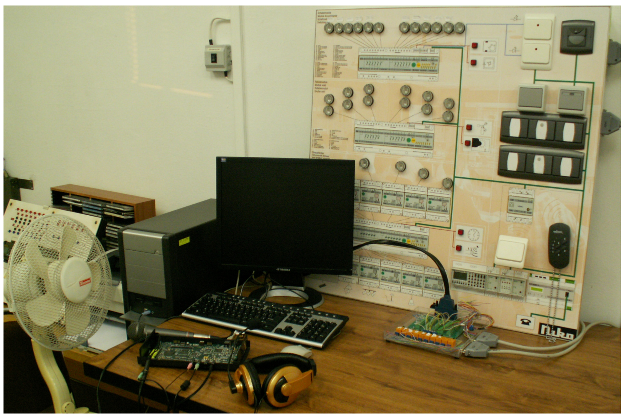

Fig. 12 Experimental workplace with DSK TMS 320C6713 and control panel with BUS system NIKOBUS.

Acknowledgments

This paper has been supported by the VŠB TU grant No. SV/4200011. Authors thanks for the support.

References

[1] B. Farhang-Borounjeny, *"Adaptive Filters, Theory and applications,"* John Wiley & Sons, Chichester, 2005, ISBN 0–471–98337–3, pp. 139-168.

[2] D. A. Poularikas, M. Z. Ramadan, *"Adaptive filtering primer with MATLAB",* Taylor & Francis Group, 2006, ISBN 0–8493–7043–4, pp. 101-122.

[3] S. Haykin, *"Adaptive filter theory,"* PRENTICE HALL, New Jersey 2002, ISBN 0-13-090126-1, pp. 231-319.

[4] J. Uhlíř, P. Sovka, P. Pollák, V. Hanžl, R. Čmejla, *"Technologie hlasových komunikací,"* nakladatelství ČVUT Praha 2007, ISBN 978–80–01–03888–8, pp. 161-165.

[5] P. Sovka, P. Pollák, *"Vybrané metody číslicového zpracování signal,"* vydavatelství ČVUT, Praha, 2003, ISBN 80–01–02821–6, pp. 89-97.

[6] J. Černocký, *"Zpracování řečových signalů"* – studijní opora, http://www.fit.vutbr.cz/.cernocky, VUT Brno, 2006

[7] J. Vaňuš, *"Hlasová komunikace s řídícím systémem",* ("Voice communication with control system"), Dissertation thesis, VŠB TU Ostrava, 2010

[8] B. Widrow, E. Walach, *"Adaptive Inverse Control: A Signal Processing Approach,"* Published by John Wiley & Sons, Inc., Hoboken, New Jersey 2008. ISBN 978-0-470-22609-4, pp. 59-87.

[9] J. Jan, *"Číslicová filtrace, analýza a restaurace signal,"* nakladatelství VUTIUM, Brno, 2002, ISBN 80-214-1558-4, pp. 287-308.

[10] J. Vaňuš, *"Implementation of the adaptive filter for voice communications with control systems,"* TSO 2009 Proceedings, Prešov, Slovakia, 2009 ISBN 978-80-553-0312-3, pp. 144-147.

[11] Chassaing R., Reay D.: *Digital Signal Processing and Applications with the TMS320C6713 and TMS320C6416 DSK,* John Wiley & Sons, Inc. New Jersey 2008, ISBN 978–0–470–13866–3, pp. 319-353.