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Adaptive Multitone Noise Cancellation from Speech Signals

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Abstract

A common problem in voice communications is the interference of noise from adjacent channels or devices. In this paper an adaptive multitone noise canceller from speech is proposed which does not require a reference of the noise signal. Different multitone noise scenarios were considered to contaminate a speech, while different LMS algorithms have been applied to obtain faster convergence time and highest signal to noise ratio. Simulation results show high reduction percent in the power spectral density of the noise tones in the recovered speech signal when using sign-data LMS algorithm.

Keywords: Adaptive Noise Cancellation, LMS, Sign-data LMS, Multitone Noise, Adaptive filter.

INTRODUCTION

Speech signals in voice communications may be contaminated with noise from different sources such as background voices in crowded environments, echoes due to voice reflections, or signals from adjacent channels or devices. Multitone noise may exist in voice channels which uses digital telephones or mobiles in which the dialed numbers are tone encoded, or due to interferences between different channels.

Extensive research was conducted to reduce different types of noise from speech signals using different adaptation algorithms which ranges from low computational complexity algorithms such as Least Mean Square (LMS) and signed LMS [1,2] to higher computational complexity such as Filtered-X NLMS [3], Normalized LMS [4], Normalized Difference LMS [5], New Variable Length LMS algorithm NVLLMS in which the value of the Mean Error Square (MSE) is used to control N and M and the step size is time varying [6], Intelligent algorithm using the matlab command 'ANFIS' (Adaptive Neuro Fuzzy Inference System) for adaptive noise cancellation [7]. A tradeoff between computational complexity and filter performance (convergence rate and signal to noise ratio (SNR)), is always existing among the mentioned algorithms. All mentioned algorithms except the ones suggested by [1,4] require a priori knowledge of noise characteristics. In this paper a multitone noise cancellation scheme is proposed which does not require a priori knowledge of the noise. The available adaptive algorithms (LMS, NLMS, and the Sign variants of LMS) in matlab platform were used to obtain best convergence time and SNR in the output after applying different scenarios of multitone noisy speech.

ADAPTIVE NOISE CANCELATION

Traditional adaptive noise cancellation (ANC) scheme is shown in Figure 1. Two inputs are required for adaptation. The first one is the desired input for which the contaminated signal d(k) is applied. The signal d(k) contains both the desired signal s(k) and the noise n(k) which are assumed to be uncorrelated with each other. The second input is the main input for which a version of the contaminating noise n'(k) (correlated with n(k)), is applied. The error signal is given by

$$e(k) = d(k) - y(k) \tag{1}$$

e(k)=d(k)-y(k) is minimized through the adaptive algorithm by updating the weights w(k) of the filter.

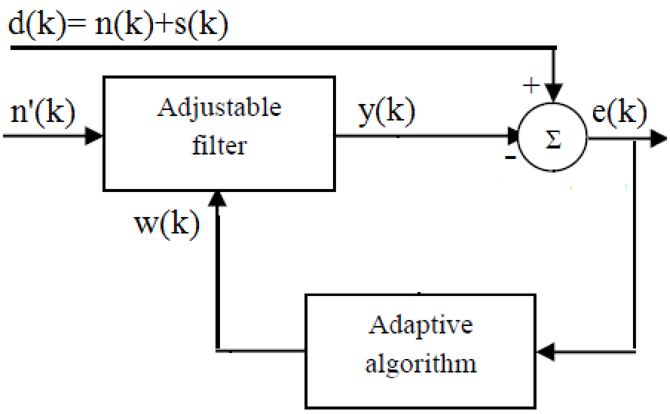


Figure 1. The adaptive noise cancellation

- For LMS algorithm the new weights are given by

$$w(k+1) = w(k) + \mu e(k) n'(k) \tag{2}$$

Where μ is the adaptive step size

- For NLMS algorithm the new weights are given by

$$w(k+1) = w(k) + \frac{\mu e(k) n'(k)}{\alpha + n'(k)^T n'(k)} \qquad (3)$$

Where α is a small constant to avoid

numerical instability of the algorithm.

- For sign-error LMS algorithm the new weights are given by

$$w(k+1) = w(k) + \mu \, sign\{e(k)\} n'(k) \qquad (4)$$

- For sign-data LMS algorithm the new weights are given by

$$w(k+1) = w(k) + \mu \, e(k) \, sign\,\{n'(k)\} \qquad (5)$$

- For sign-sign LMS algorithm the new weights are given by

$$w(k+1) = w(k) + \mu \, sign\{e(k)\} \, sign\{n'(k)\} \qquad (6)$$

Where sign{x} is the signum function

THE PROPOSED NOISE CANCELLATION SCHEME

In the proposed scheme the desired input of the adaptive filter is left unconnected (or connected to ground) and the contaminated signal (desired signal + noise) is applied to the main input of the adaptive filter. Hence the error signal in equation (1) becomes

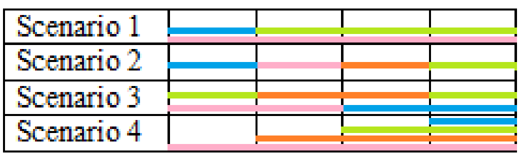
$$e(k) = d(k) - y(k) = 0 - y(k) = -y(k) \qquad (7)$$

That is the error signal is only the inversion of the desired signal which is minimized through the applied adaptive algorithm until steady state convergence is reached.

Noise Contamination Scenarios

The proposed scheme is tested by applying speech signal which is contaminated by multitone in different ways. The first way is to add a number of tones (3, 4, or 5) from the start of the speech until the speech end. In the second way the tones which are added to the speech signal are varied with time. That is the frequencies of the additive noise tones are changed with time. Table 1 shows the four scenarios of this type of noise contamination which have been used.

Table 1. The multitone noise contamination scenarios



Pink = 0.2 kHz, red=1kHz, green=1.5kHz, blue=2kHz

SIMULATION MODELS

The process of noise contamination is implemented using different models on Matlab Simulink (Figure 2). Here the speech signal is read from (.wav) file, and noise tones are added to the speech signal according to the scenarios mentioned before then the resultant speech Contaminated signal is written to a new (.wav) output file. The simulation model of the proposed adaptive noise canceller is also implemented on Matlab Simulink platform (Figure 3). Here the (.wav) file of the contaminated speech signal is read and applied to the main input of the adaptive filter, and the output signal of the filter is written into another (.wav) file.

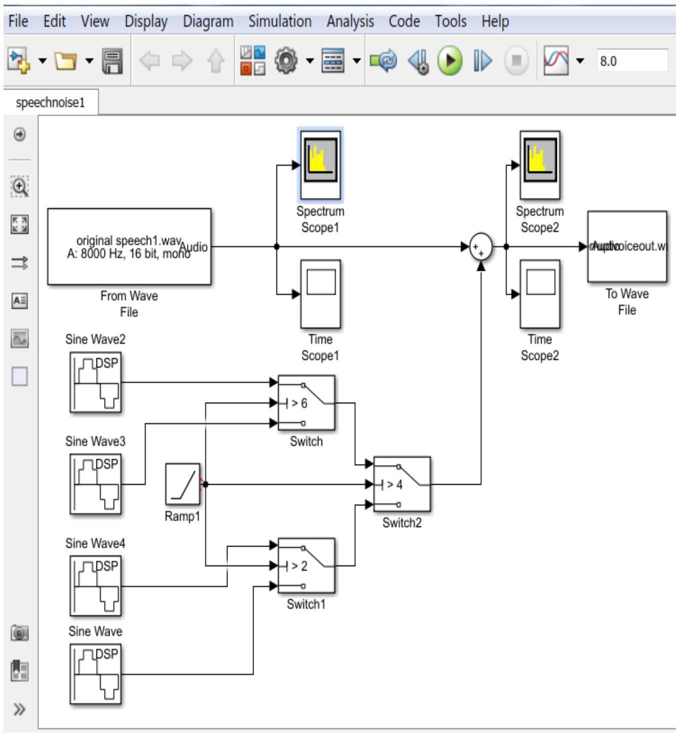


Figure 2. Simulink model of adding multitone noise to the speech using scenario 2

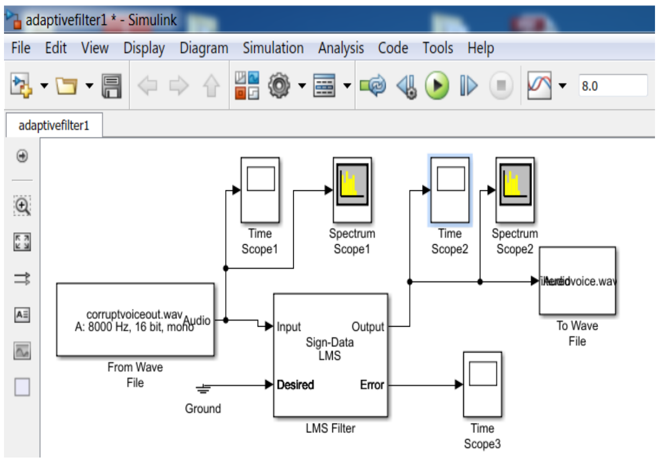


Figure 3. The Simulink model of the proposed ANC

RESULTS AND DISCUSSION

The proposed scheme is tested in three steps as follow

Step 1

The proposed scheme is first tested by changing filter parameters to find the best adaptive algorithm with best filter parameter for filtering the noisy speech signal. For this purpose the original speech signal (lasted for 8 sec. and sampled at 8000 Hz) is contaminated with three tones (0.2 kHz, 1 khz, and 2 kHz). The adaptation step size (μ) and the filter tap are changed using different adaptive algorithm (LMS, NLMS, Sign-Error, Sign-Data, and Sign-Sign). A range of (0.0005- 0.01) is used for (μ), while a range of (15-30) is used for filter tap. It is found that as the adaptation step size is increased the convergence time decreases, but the filtered speech is attenuated for all mentioned algorithms except for the Data-sign algorithm. Hence using the Data-Sign algorithm with adaptive step size of (0.001) and filter tap of (20), gave the shortest convergence time and lowest speech signal attenuation.

Step2

In this test, the speech signal which is contaminated with (3, 4, and 5) tones is applied to the main input of the adaptive filter using the Data-Sign algorithm with a step size of (0.001) and filter tap of (20). The original speech signal in both time domain and frequency domain are shown in Figure 4. The input and output signals of the filter using four tones noise in time domain are shown in Figure 5, while their spectral response are shown in Figure 6. Table 2 shows the Power Spectral Density (SPD) of both the input and the output signals for 3, 4 ,and 5 tones noise. The recovered speech was clearly heard except a peep sound heard at the beginning of the speech which last for less than 0.5 second.

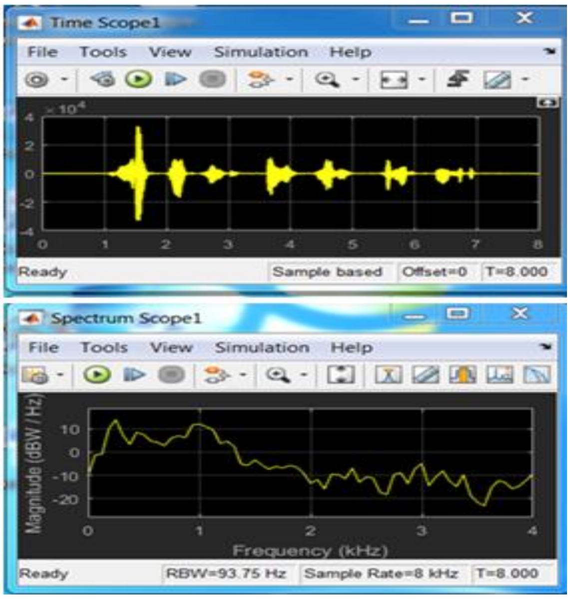


Figure 4. The original speech in time domain (top) and in frequency domain (bottom)

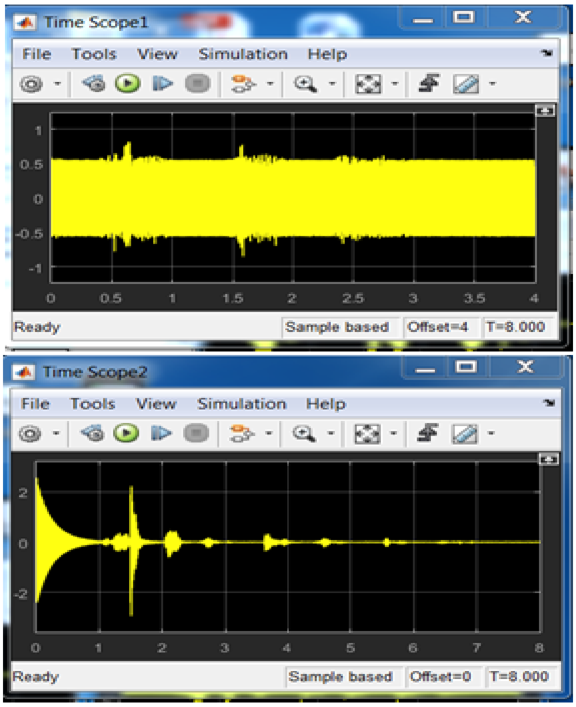


Figure 5. The input (top) and the output (bottom) signals of the filter in time domain using four noise tones

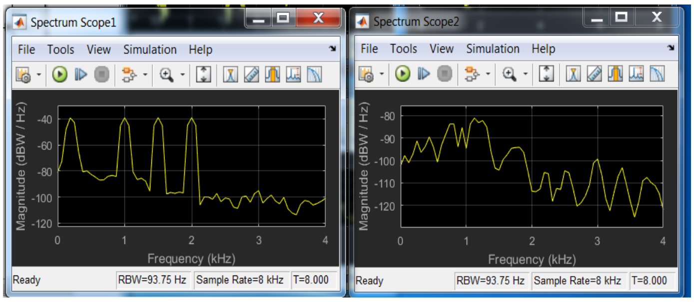


Figure 6. The input (left) and the output (right) signals of the filter in frequency domain using four noise tones

Table 2. The PSD of the input and output for 3, 4, and 5 tones noise

Noise frequency (kHz)	Input PSD (dBW/Hz)	Output PSD (dBW/Hz)		
		3 tones noise	4 tones noise	5 tones noise
0.188	-39.26	-97.50	-97.24	-93.87
0.625	-39.95			-95.94
1	-39.10	-94.51	-94.63	-94.24
1.5	-39.05	-105.22	-104.30	-101.47
2	-39.06		-113.69	-111.36

Step 3

In this test, time variant multitones (see Table1), are added to the speech signal and the resultant contaminated speech signal is applied to the adaptive filter. The adaptive filter parameters are the same as those used in step 2. Figures (7-12) show the Spectral Power Density (PSD) of the input contaminated speech signal (according to the four scenarios shown in Table1) and the output recovered speech signal of the adaptive filter. As scenario 2 represents the most variable tones (each 2 seconds) more detailed results are shown at different times in Figures (8-11).

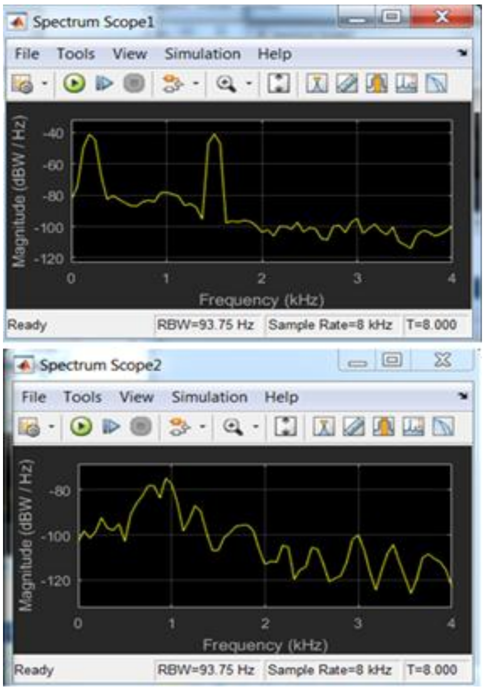


Figure 7. The SPD of both the input (top) and the output (bottom) signals of the proposed adaptive filter using scenario1 of noise contamination at time = 8 seconds.

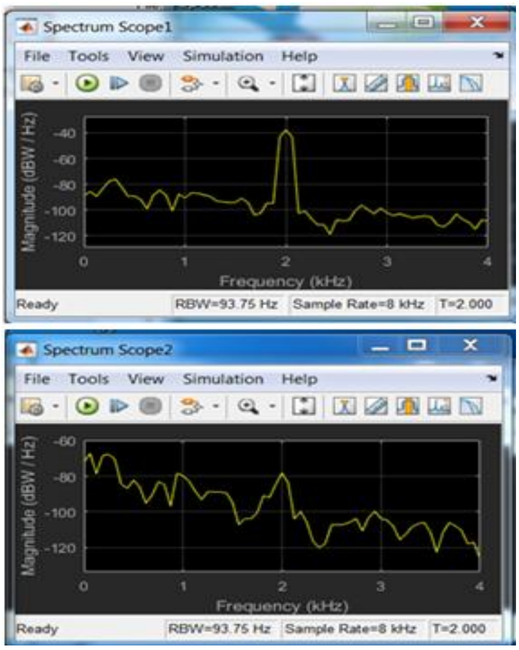


Figure 8. The SPD of both the input (top) and the output (bottom) signals of the proposed adaptive filter using scenario2 of noise contamination at time = 2 seconds.

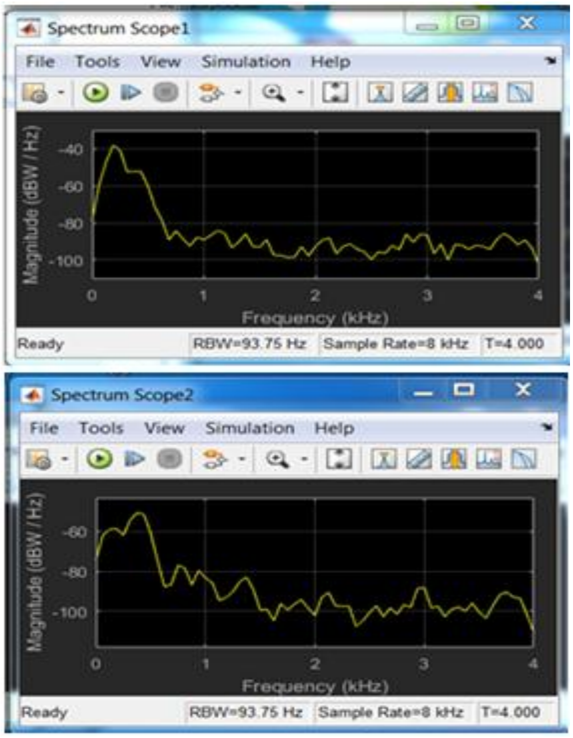


Figure 9. The SPD of both the input (left) and the output (right) signals of the proposed adaptive filter using scenario2 of noise contamination at time = 4 second.

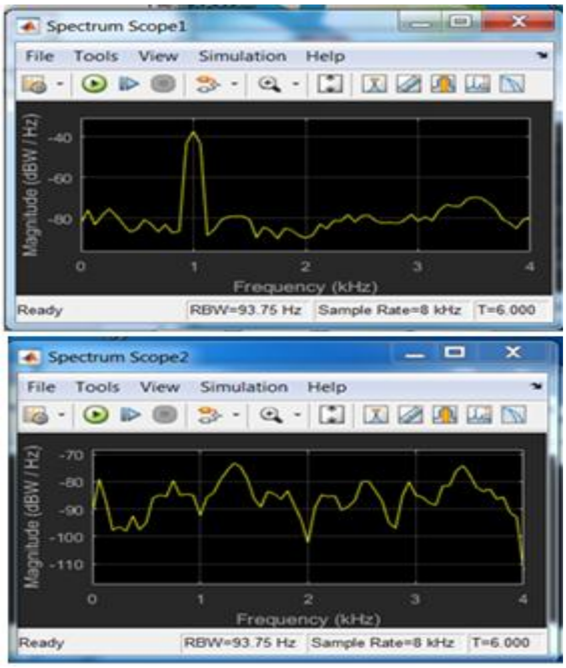


Figure 10. The SPD of both the input (left) and the output (right) signals of the proposed adaptive filter using scenario2 of noise contamination at time = 6 second.

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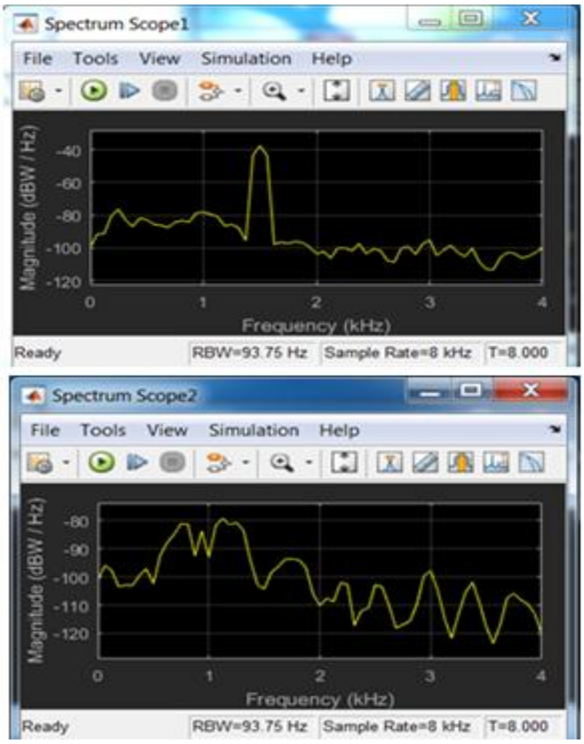

Figure 11. The SPD of both the input (top) and the output (bottom) signals of the proposed adaptive filter using scenario2 of noise contamination at time = 8 second.

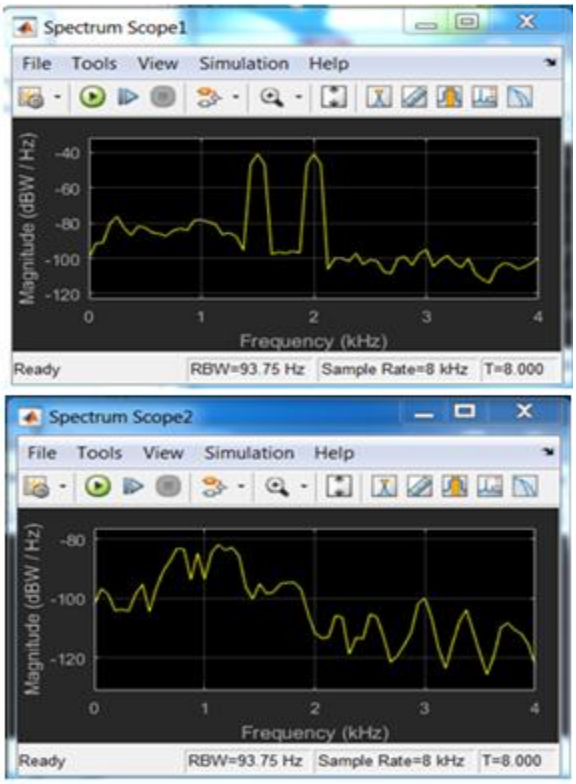

Figure 12. The SPD of both the input (top) and the output (bottom) signals of the proposed adaptive filter using scenario3 of noise contamination at time = 8 seconds.

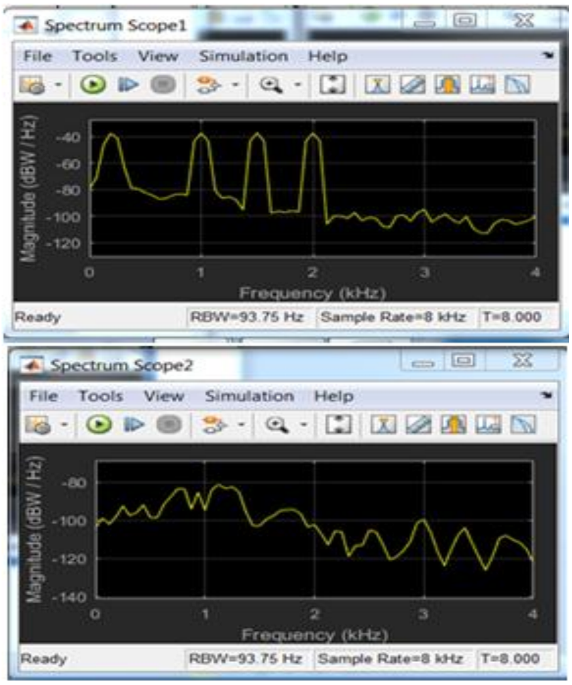

Figure 13. The SPD of both the input (top) and the output (bottom) signals of the proposed adaptive filter using scenario4 of noise contamination at time = 8 second.

The reduction percent in PSD of the used noise tones in the output recovered signal (for the four scenarios) with respect to the PSD of these tones in the contaminated speech signal are summarized in Table 3. The recovered speech signal for the four scenarios is clearly heard except some short peeps (for less than half a second) heard at the beginning of the speech and at instances when tones are changed or added which is due to the required convergence time.

Table 3. Reduction in PSD of the noise tones in the filter output with respect to the PSD of these tones in the filter input.

	Tone frequency	At t = 2 s	At t = 4 s	At t = 6 s	At t = 8 s
Scenario 1	188 Hz	32%	35%	78%	140%
	1500 Hz		72%	120%	173%
	2000 Hz	98%			
Scenario 2	188 Hz		53%		
	1000 Hz			146%	
	1500 Hz				178%
	2000 Hz	108%			
Scenario 3	188 Hz	40%	44%		
	1000 Hz		52%	93%	
	1500 Hz		56%		132%
	2000 Hz			136%	172%
Scenario 4	188 Hz	97%	57%	106%	160%
	1000 Hz		67%	120%	152%
	1500 Hz			135%	174%
	2000 Hz				173%

CONCLUSION

A new adaptive noise canceller scheme for speech contaminated with multitones noise is modeled and simulated using Signed-Data algorithm. The proposed scheme has low computational complexity and does not require a priori knowledge of the noise nor the speech signals. Simulated results show high reduction percent in the PSD of the contaminating tones in the output of the adaptive filter, and clear recovered speech signal is heard at the output (output .wav file). Future work could be considering variable filter parameters algorithms rather than steady state parameters to reduce the short peeps which occur at times when tones are changed.

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