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ADAPTIVE NOISE SUPPRESSION IN SPEECH SIGNAL TRANSMISSION

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ABSTRACT
A modified LMS algorithm with the weighted sum method is applied to the adaptive noise suppression system with two inputs and one output. The algorithm is superior in solving the excess mean-squared error problem by reducing the step length in weight update equation when the signal noise ratio is positive. The dependence of noise suppression on the coherence characteristics of input signals, i.e. on the relative position and distance between the two microphones, and on the directivity pattern of primary microphone are discussed together with experimental results.

1 - INTRODUCTION

Early in 1975 a modulated cabin noise experiment was conducted by Widrow et al., in which the feasibility of picking up a speech signal from a noise background with an adaptive noise suppression method was demonstrated [1]. Since then on, a series of similar work followed [2-3]. In this method, the basic configuration is a system with two inputs and one output, where the primary input contains a desired signal corrupted by additive noise, the reference input is a noise signal correlated with the noise in the primary input. The reference signal is fed to a digital filter with an adaptive algorithm. By subtracting the filtered signal from the primary input, an error signal output can be given, which is composed of the desired signal and a suppressed noise. The key problem here is that the reference signal should be coherent with the noise in the primary, and at the same time, not with the desired speech signal. In the adaptive digital filter, the LMS algorithm is often applied. Experiments showed that to give a good performance, the filter parameters such as step length, filter length and sampling frequency should be well designed according to a given sound field response reflected in the primary input. Excess mean-square error is the main defect of LMS algorithm, which is generated by the improperly updated weights, especially when the Signal Noise Ratio (SNR) is positive.

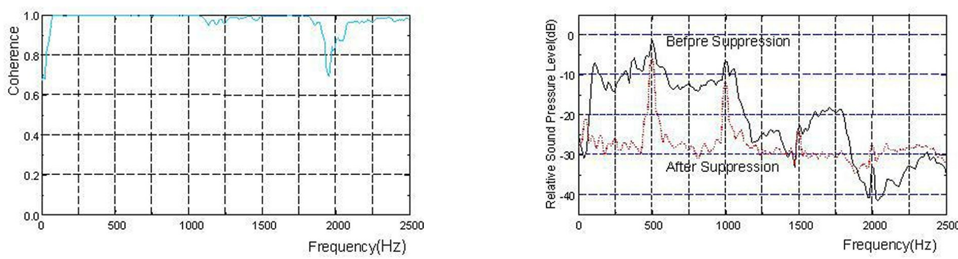
In this paper, the influence of filter parameters and the algorithm itself on the system performance is discussed, based on the coherence analysis of the sound field. A modified LMS algorithm with the weighted sum method is applied, which is superior in solving the excess mean-squared error problem by reducing the step length in the weight update equation when the signal noise ratio is positive. The dependence of noise suppression on the coherence characteristics of input signals, i.e. on the relative position and distance between the two microphones, and on the directivity pattern of primary microphone are discussed together with experimental results. To evaluate the performance of the modification, a simple method of calculating the short-term SNR in speech signal intervals is presented.

2 - EXPERIMENTAL CONFIGURATION

Experiments were conducted in a reverberation room of 5.0 $\times$6.0 $\times$3.3 m^3. A 12" loudspeaker enclosure placed in a corner of the room was used to perform the noise source. Another loudspeaker closed in a box with a ϕ20mm opening was designed for playing the speech signal and to modulate a man's head. Two minor omnidirectional electret condenser microphones of the type of CZII$-$60 are used to pick up the input signals. A TMS320C25 DSP board with 2 12-bit A/D and a 12-bit D/A converter is applied to the digital filter. A dual channel low-pass filter is cascaded in between the preamplifiers and the A/D converters for anti-aliasing. A HP 35670A dual channel signal analyzer is used to give the sound field coherence and the system performance.

3 - INPUT SIGNAL COHERENCE AND NOISE SUPPRESSION

To subtract the noise contained in the primary input, the expected output of digital filter should close to the noise in both amplitude and phase characteristics as near as possible. Two conditions have to be fulfilled: (1) the noise signal coherence between the 2 inputs should be good enough; and (2) the impulse response of digital filter can approximate the transfer function between the reference and the primary channel. Incoherence between the reference input signal and the speech signal in primary input is as important as the noise coherence, for that the speech signal would also be attenuated by the digital filter output and the SNR improvement would be less.
The second condition can be fulfilled by designing the filter parameters according to the transfer function. When the relative positions of the 2 microphones are fixed, the reference signal can be supposed as the input of a system approximated by the digital filter, and the primary as the output. Hence the impulse response of the system determines the filter length. Sampling frequency should be selected as 6 times of the up limit frequency of noise signal. The filter length divided by the sampling period gives the filter order. The weights can be updated automatically by the adaptive algorithm.



(a): Coherence. **(b):** Noise suppression performance.

Figure 1: Coherence and noise suppression when the two microphones are 1 cm apart.

It is obvious that the coherence of 2 input signals is dependent on the distance of the microphones. Figures 1 and 2 give the coherences and noise suppression performances of the system when the distance equals 1 cm and 7 cm respectively. The improvement of SNR for 1 cm is 8.3dB, and for 7 cm is 12.8dB, where the useful signal played by the head simulation is supposed to be the superposition of 4 pure tones of 500Hz, 1000Hz, 1500Hz and 2000Hz as shown in the figures. The results show that the coherence of the 2 inputs of microphones of 1cm apart from each other is much better than that of 7cm. The noise suppression performance of 1 cm is also much better, but the improvement of SNR is not as great as that of 7cm. The reason is just that the coherence of the useful signals in the 2 inputs is almost the same as the noise.
To decrease the coherence of useful signals between the 2 inputs, a CZII−64 directional microphone was used to replace the primary signal pick-up. The sensitivity difference in the front and back directions is more than 8 dB in the frequency range of 300-2000 Hz. When the primary microphone was placed directional to the simulated mouth and 5 cm apart from it, and the reference was 5 cm away from the primary, the situation could be greatly improved. When the useful signal was set to be a pure tone of 340 Hz. The coherences were almost the same in other frequencies, but a difference of about 0.16 was produced at 340 Hz.

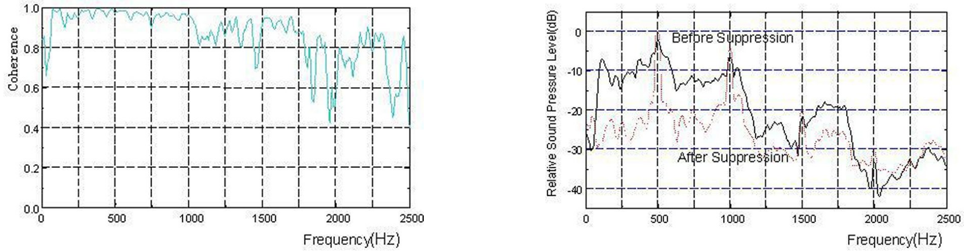
4 - IMPROVEMENT ON LMS ALGORITHM

As we know, the weight update equation for LMS algorithm can be written as

$$W\left(n+1\right)=W\left(n\right)+\mu e\left(n\right)X\left(n\right) \tag{1}$$

where n is the time series, μ is step length, $e(n)$ is the system output and also the error signal for the adaptive algorithm, $W(n)$ is the weight vector at time n.
For a given set of weights, the power output of the system can be derived as

$$\sigma_e^2\left(n\right)=\sigma_t^2+\xi_{\min}+J_{ex}\left(n\right) \tag{2}$$



(a): Coherence. **(b):** Noise suppression performance.

Figure 2: Coherence and noise suppression when the two microphones are 7 cm apart.

where σ_t^2 is the power of incoherent useful signal, $\xi_{\min}$ is the minimum mean-square error. The sum of the two terms is the incoherent power of the two inputs. The other term $J_{ex}(n)$ is excess mean-square error, generated by the deviation of the weights from their optima. The step length in LMS is generally selected as $\mu = \frac{\alpha}{L\sigma_x^2}$, where L is filter order, $\sigma_x^2 = E\left[x^2(n)\right]$ is the mean power of input signal, α is a step length parameter. The excess mean-square error would increase linearly with the useful signal [4-5]. To avoid this situation, the expression of step length can be revised,

$$\mu = \frac{\alpha}{L\left[\sigma_e^2(n) + \alpha\sigma_x^2(n)\right]} \tag{3}$$

where $\sigma_e^2(n)$ is the system output power.
Experiments showed that the modified LMS algorithm performed better than the traditional one, especially when the *SNR* was above 0. Table 1 gives some results for L=64 and the noise is below 1 kHz. SNR_{in} is the original signal noise ratio of the input from the primary microphone. G_{SNR} is the signal noise ratio improvement of the system, which can be generally written as

$$G_{SNR} = SNR_{out} - SNR_{in} \tag{4}$$

where SNR_{out} is the signal noise ratio of the system output. Because that the speech signal can not be separated from the output, we tried to give G$_{SNR}$ approximately by comparing the noise attenuation in speech signal intervals. The presupposition of such a method of calculating SNR improvement is that the speech signal changes little.

SNR_{in}	-10	-3	0	+3
G_{SNR} - Traditional	13.0	12.3	8.2	3.4
G_{SNR} - Modified	15.8	15.0	9.5	7.2

Table 1: SNR improvement for traditional and modified LMS algorithms (dB).

5 - CONCLUSIONS

Coherence analysis is necessary for a well designed adaptive noise suppression system applied in speech communications. The coherence of noise signals and incoherence of speech signals between the primary and reference inputs could be optimized by relative configurations of the two microphones and the directional pattern of the primary signal pick-up. The modified LMS algorithm with step length variant with both input and error signals performs better, especially when the input SNR is positive.

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