Noise Cancellation System Based on Frequency Domain Adaptive Filter Using Modified DFT Pair

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Abstract: It is well known that a Frequency Domain Adaptive Filter (FDAF) converges faster than a Time Domain Adaptive Filter (TDAF) even when the input signal is colored such as a speech signal. We have proposed the FDAF using the Modified Discrete Fourier Transform Pair (MDFTP) and its realization and effectiveness has been confirmed through the computer simulations. In this paper, we apply the FDAF using the MDFTP to the noise cancellation system. The proposed system is based on the Adaptive Line Enhancer (ALE) and utilizes single microphone; therefore it is suitable for the portable electronic equipment. Moreover, we propose to utilize the MDFT for detecting of the pitch in the speech because the number of data points in the MDFT must be equal to the pitch to confirmed that the noise can be removed to near the level of SNR.

1. Introduction

The miniaturizing, lightening, and making to low-cost the device are one of the reason of the spread of user of a cellular phone. However, the speech quality is an important factor recently. The cellular phone is used outdoor where various noise sources exist though the conventional telephone is used indoor.

Generally, the speech signal is non-stationary signal. The Adaptive Digital Filter (ADF) which is able to adjust the characteristic of filters adaptively is effective about the processing about non-stationary signal. But, if the input signal is colored, convergence speed at which the adaptive coefficients converges to optimum solution in Time Domain Adaptive Filter (TDAF) based on the Transversal Filters becomes slow because the eigenvalue spread of the input autocorrelation matrix is more than that of white signal. To accelerate convergence speed it is necessary for processing the speech signal which is basically colored.

To improve the convergence speed, the Frequency Domain Adaptive Filter (FDAF) in which the input autocorrelation matrix is approximately digonalized by using the Discrete Fourier Transform (DFT) and Normalized Orthogonal Transform is proposed [1]. We have proposed the FDAF using Modified DFT Pair (MDFTP) [5][6]. The MDFTP can be obtained by simplifying the operation of the DFT and the its inversion. The MDFT can be achieved by real value operation. Especially the Modified Inverse DFT (MIDFT) is obtained only by adding the MDFT outputs [3]. In addition, computational complexity is reduced to 0 (N log N) by using part of signal flow of First Fourier Transform (FFT) instead of block processing. The multiplication in the MDFT can be eliminated because the MDFTP is based on the Finite Impulse Response FIR filter, and cosine function in DFT is approximated using powers of two. Therefore, the FDAF using MDFTP is suitable for processing of colored signal such as speech in real time. And, we have proposed the structure when assuming that the pitch of the speech is constant during processing [9].

In this paper, we propose the noise cancellation system as one of applications based on the FDAF using the MDFTP. The general adaptive noise canceller is unsuitable for the portable electric equipment, because it needs two microphones to remove the noise signal from the speech signal. The proposed noise cancellation System is based on the Adaptive Line Enhancer (ALE). In the detection of the pitch in the speech signal, a simple pitch detection method

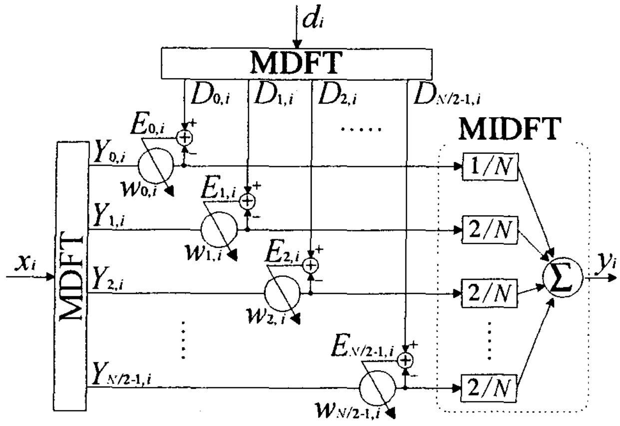

Fig.1.A Structure of FDAF using MDFTP

which utilize the MDFT outputs is proposed.

Section 2 has explanations about the FDAF using MDFT pair. In Sec. 3 we propose the structure of noise cancellation system and the pitch detector utilizing the MDFT. Finally, to verify the effectiveness of the proposed noise cancellation system, we carry out the computer simulation in Sec. 4.

2. FDAF Using MDFTP

By simplifying DFT pair, the MDFTP is defined as [3]

$$Y_{k,i} = \sum_{n=0}^{N-1} x_{i-n} \cos(2\pi nk/N) \tag{1}$$

$$x_i = \frac{1}{N} Y_{0,i} + \frac{2}{N} \sum_{k=1}^{N/2-1} Y_{k,i} \,. \tag{2}$$

Equation (1) is the MDFT and Eq. (2) is the MIDFT. $Y_{k,i}$ indicates the k-th spectrum of the input signal x_i. N is the number of data points in the MDFTP. The subscripts i and n are time indices, and k is a frequency index. From these two equations, it is clear that the MDFTP can be achieved by the real value operation. Moreover, when an input signal consists of multiple harmonic frequency waves, the MDFT decomposes it into component of each harmonic signal keeping their phase difference. Therefore, the kth DFT output corresponds to kth harmonic frequency signal.

A structure of the FDAF using MDFTP is shown in Fig.1. y_i and d_i are the output signal and the desired one, respectively. The input spectra $Y_{k,i}$ and the desired ones $D_{k,i}$ are obtained by applying the MDFT to the input signal x_i and the desired one d_i, respectively. The adaptive weight $w_{k,i}$ is updated to minimize the spectral error $E_{k,i}$ given by

$$E_{k,i} = D_{k,i} - w_{k,i} Y_{k,i} \,. \tag{3}$$

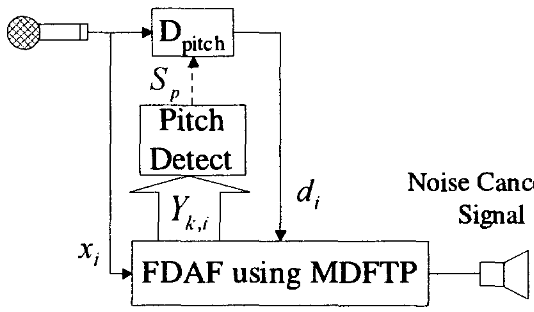

Fig.2.Proposed Noise Cancellation System

The adaptive algorithm is adjusted as

$$w_{k,i+1} = w_{k,i} + 2\mu_{k,i} E_{k,i} Y_{k,i} \,, \tag{4}$$

where $\mu_{k,i}$ is the normalized step size

$$\mu_{k,i} = 0.5 / \left(|Y_k|_{peak} \right)^2 \,, \tag{5}$$

and $|Y_k|_{perk}$ is the maximum of $|Y_k|$ from i to $i - (N - 1)$.

This structure can achieve normalized orthogonal transformation by non-block processing, resulting in the high-convergence speed of FDAF has an adaptive algorithm with scalar operation. The FDAF adjust the amplitude of spectrum but it does not control the phase, hence it can not be applied to application which requires the control of the phase.

3. Noise Cancellation System

In this section, we apply the FDAF using the MDFTP to the noise cancellation system for speech signal.

3.1. Structure of Noise Cancellation System

A speech signal, especially a vowel is assumed stationary in a short interval of time of about 30 milliseconds, and it can be considered as the multi-sinusoidal signal based on the fundamental (pitch) frequency. The noise cancellation system can effectively remove the noise in frequency domain by detecting the pitch period.

The Structure of the proposed noise cancellation system is shown in Fig.2. The proposed system is based on the ALE [4] which is suitable for detecting multiple sine waves buried in a wide-band noise. The general adaptive noise canceller is unsuitable for the portable electronic equipment, because it needs two microphones to remove the noise signal from the speech signal. On the other hand,

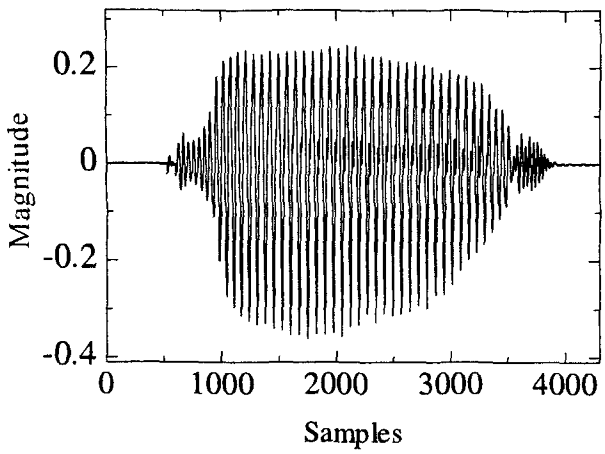

Fig.3. Waveform of /i/

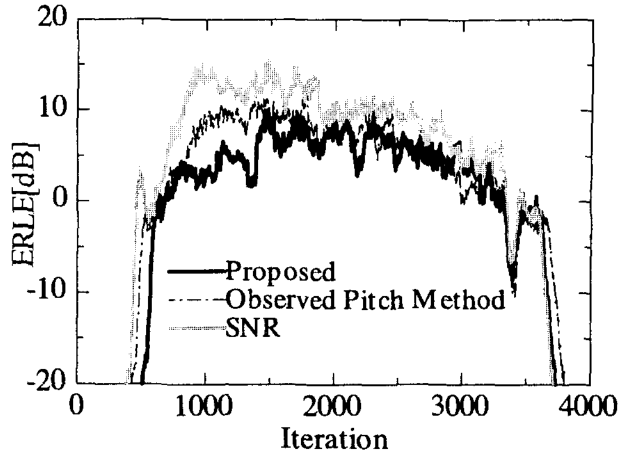

Fig.4. Noise Reduction Performance

the proposed noise cancellation system is capable of achieving this by using a single microphone. Therefore, it is suitable for portable electric equipment.

The noisy speech signal is taken from the microphone into the system as the input signal x_i. x_i which has been delayed by D_{pitch}, it considered as the desired signal d_i of the FDAF. The delay D_{pitch} corresponds to the decorrelation parameter which makes the noise in the input signal decorrelated with that in the desired one in the ALE. If there is the phase difference between the input signal and the desired one, the estimation error cannot be reduced because the FDAF using MDFTP can not control the phase. Therefore, the delay D_{pitch} must be adjusted reduce the phase difference. In this paper, we present a simple pitch detecting method. Then, the delay D_{pitch} is adjusted so as to correspond to the pitch period. The number of samples N in the MDFTP is set to be constant and large enough to detect necessary frequency elements.

3.2. Pitch Detection Method

Many pitch detection methods, for example analyzing the waveform of speech signal and using the autocorrelation parameter etc, have been examined, but a perfect method has not been proposed. In this section, we propose a new pitch detection method, which is simple in comparison to other pitch detection methods, and which utilized the MDFT output.

We assume that the pitch of the speech corresponds to the lowest frequency component, and it has the spectrum peak. When kth MDFT output has a peak in lowest frequency, the number of samples which corresponds to the pitch S_p is estimated by

$$S_p = N/k_p \, . \tag{6}$$

This method is very simple, so that the accuracy of detection may be lower when compared with other detection methods. However, by using this method, the noise cancellation system which includes the proposed pitch detector can be realized without any other additional circuits.

4. Computer Simulation

To confirm the performance of the proposed noise cancellation system, we carried out the computer simulation.

A speech /i/ which has been pronounced by a male was used as the desired signal (16bit data at 11025Hz sampling). Fig. 3 shows its waveform. The white Gaussian noise of variance 0.04^2 was added to the desired signal. N of the MDFTP was 500. The Echo Return Loss Enhancement (ERLE) given by Eq.(7) was used for verification.

$$\mathrm{ERLE} = 10\,\log_{10}\frac{E[d_i^{\,2}]}{E[e_i^{\,2}]} \quad \text{(dB)} \tag{7}$$

where e_i is an error signal between the output signal and the desired one. Moreover, the Signal Noise Ratio (SNR) is defined as

$$\mathrm{SNR} = 10\,\log_{10}\frac{E[d_i^{\,2}]}{E[n_i^{\,2}]} \quad \text{(dB)} \tag{8}$$

where n_i is the additive noise.

Fig. 4 shows the simulation results. For comparison, the result by using for preliminary observation of waveform pitch detection and the SNR curve are also provided. By using the observed pitch method, the noise can be removed effectively though it is a little inferior to SNR. The proposed system also reduces the noise but the performance is a little degraded at the beginning of speech. The reason is due to the low power of the speech signal, hence comparison of each spectrum size becomes difficult. Further study is therefore needed in order to improve on the performance of the pitch detection.

5. Conclusions

We proposed the noise cancellation system based on the FDAF using the MDFTP. In addition, the simple pitch detection method using the MDFT output is applied to the proposed noise cancellation system. We also confirmed the performance and feasibility of the proposed system through the computer simulation using a speech signal. The simulation result showed that noise was reduced in the stationary part of speech while the performance is a little degraded at the beginning of speech. It is therefore necessary to investigate other methods of pitch detection, which are also robust in the non-stationary part of the speech signal.

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