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STUDY ON BEHAVIOR OF THE ADAPTIVE FILTER BASED ON TDBLMS ALGORITHM FOR IMAGE NOISE CANCELLATION

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ABSTRACT- Image denoising is a kind of processing of image which belongs to image restoration, and the ultimate goal of restoration techniques is to improve an image in some predefined sense. This noise gets introduced during acquisition, transmission, reception, storage and retrieval processes. Various image restoration techniques have been developed to restore an image degraded by noise. Up to now, most of the restoration filters have been investigated **.**In order to improve the power of signal to noise ratio of image, a local adaptive Wiener filter with TDBLMS algorithm is applied to a 2-D image. Several 1-D windows could be constructed on the direction character of sub-image in the filter window after wavelet transform.

Keywords:-, De-noising, Power signal to noise ratio, Local adaptive filter, least mean-square

INTRODUCTION

In all practical situations, the received speech waveform contains some form of noise component. The noise may be a result of the finite precision involved in Coding the transmitted waveform (quantization noise), or due to the addition of acoustically coupled background noise. Depending on the amount and type of noise, the quality of the received waveform can range from being slightly degraded to being annoying to listen to, and finally to being totally unintelligible. The problem of removing the unwanted noise component from a received signal has been the subject of numerous investigations. The pioneering work of Wiener and others give an optimum approach for deriving a filter that tends to suppress the noise while leaving the desired signal relatively unchanged. The design of these filters requires that the signal and the noise be stationary and that the statistics of both signals be known a priori. In practice, these conditions are rarely met.

The classical approach to noise cancellation is a passive acoustic approach. Passive silencing techniques such as sound absorption and isolation are inherently stable and effective over a broad range of frequencies. However, these tend to be expensive, bulky and generally ineffective for cancelling noise at the lower frequencies. The performance of these systems is also limited to a fixed structure and proves impractical in a number of situations where space is at a premium and the added bulk can be a hindrance. The shortcomings of the passive noise reduction methods have given impetus to the research and applications of alternate methods of controlling noise in the environment. Various signal processing techniques have been proposed over the years for noise reduction in the environment. The explosive growth of digital processing algorithms and technologies has given an impetus to the application of these techniques to the real world.

Digital Signal Processors (DSPs) have shrunk tremendously in size while their processing capabilities have grown exponentially. At the same time the power consumption of these DSPs has steadily decreased following the path laid down by Gene’s law. This has enabled the use of DSPs in a variety of portable hearing enhancement devices such as hearing aids, headsets, hearing protectors, etc.

There are two different approaches for electrical noise reduction. The first approach is passive electrical noise reduction techniques, such as those applied in hearing aids, cochlear implants, etc. where the signal and ambient noise are recorded using a microphone, noise reduction techniques such as spectral subtraction, the LMS algorithm, etc. are applied and the listener hears only the clean signal. One of the important assumptions of this technique is that the listener is acoustically isolated from the environment. This assumption is however not valid in a large particularly those number of situations where the ambient noise has very large amplitude. In such situations, the second approach of Active Noise Cancellation (ANC) is applicable. ANC refers to an electromechanical or electro acoustic technique of cancelling acoustic disturbance to yield a quieter environment. The basic principle of ANC is to introduce a cancelling “antinoise” signal that has the same amplitude but the exact opposite phase, thus resulting in an attenuated residual noise signal. ANC has been used in a number of applications such as hearing protectors, headsets, etc. The traditional wideband ANC algorithms work best in the lower frequency bands and their performance deteriorates rapidly as the bandwidth and the center frequency of the noise increases. Most noise sources tend to be broadband in nature and while a large portion of the energy is concentrated in the lower frequencies, they also tend to have significant high frequency components. Further, as the ANC system is combined with other communication and sound systems, it is necessary to have a frequency dependent noise cancellation system to avoid adversely affecting the desired signal.

Adaptive filters that use the error signals estimated from the input signals and the expected signal to adjust the coefficients for achieving a better performance are widely used in various applications [1]. The dimension of the adaptive filters varies from application to application.

Image noise is random variation of brightness or colour information in images, and is usually a type of electronic noise. It can be produced by the sensor and circuitry of a scanner or digital camera. Image noise can also originate in film grain and in the unavoidable shot noise of an ideal photon detector. Image noise is an undesirable by-product of image capture that adds spurious and extraneous information.

There are many types of IMAGE noise which are as follows:-
Gaussian noise: caused by poor illumination and/or high temperature, and/or transmission e.g. electronic circuit noise

Salt and paper noise: impulsive noise is sometimes called salt-and-pepper noise or spike noise

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Shot noise: typically that caused by statistical quantum fluctuations, that is, variation in the number of photons sensed at a given exposure level. This noise is known as photon shot noise

Quantization noise: The noise caused by quantizing the pixels of a sensed image to a number of discrete levels is known as quantization noise. Adaptive filters are widely used in various applications for achieving a better performance [1]. An adaptive filter is a filter that self-adjusts its transfer function according to an optimization algorithm driven by an error signal the dimension of the adaptive filters varies from application to application. In the fields of digital signal processing and communication echo cancellation, noise canceling, and channel equalization [2]-[6], the onedimensional adaptive algorithm are generally adopted.

The 1-D adaptive algorithms are usually classified into two families. One is the least-mean-square (LMS) family and the other is the recursive-least-square (RLS) family. The algorithms in the LMS family have the characteristics of easy implementation and low computational complexity [1]. In 1981, Clark [7] proposed the block least-mean-square (BLMS) approach which is an application extended from the block processing scheme proposed by Burrus [8]. In such an approach, the computational complexity is dramatically reduced.

In the applications of digital image processing, two dimensional (2-D) adaptive algorithms such as TDLMS, TDBLMS, OBA, OBAI, and TDOBSG are usually used [8]. Either in TDLMS or TDBLMS, the convergence factors are constant. Instead of the constant convergence factors in TDLMS and TDBLMS, the space-varying convergence factors are used in OBA, OBAI, and TDOBSG for better convergence performance. However, such space-varying convergence factors will increase the computational complexity due to the computations for the new convergence factor of next block. TDBLMS adaptive filter with weighttraining mechanism by finding a suitable weight (coefficient) matrix for the digital filter in advance was proposed by Chuen-Yau Chen and Chih-Wen Hsia [9] .Then, treat this weight matrix as the initial weight matrix for the processing of noise abolition.

The mainly algorithm for noise cancellations which is:-

Two Dimensional BLOCK LMS ALGORITHM

2-D signal is partitioned into blocks with a dimension of LxL for each in the 2-D disjoint block- by-block image processing. An image with R rows of pixel and C columns of pixel partitioned into $\frac{R}{L} \times \frac{C}{L}$ blocks is illustrated in Fig. 1. The block index S and the spatial block index (r, c) is related by [12]

$$s = r - 1 \times (c/L) + c \qquad (1)$$

Where r 1,2 ......R/L and C=1,2,....C/L for convenience, the $(r, c)^{th}$ element $d(r, c)$ of the image can be treated as the rb , cb^{th} element in the $S^{\ th}$ block and denoted as the element $d_s(r_b, c_b)$.

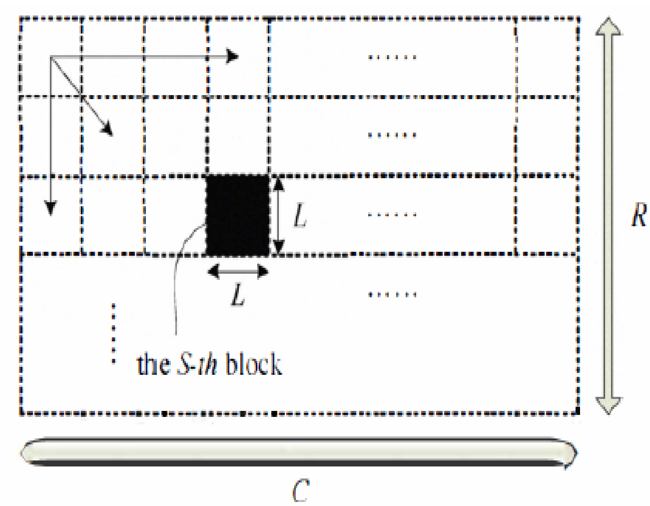

Fig 2D

block by block processing with disjoint square block of dimension LxL

The relationship is:

$$d_s(r_b, c_b) = d[(r-1)L + r_b, (c-1)L + cb \qquad (2)$$

The block processing is started by processing the image block-by-block sequentially from left to right and from top to bottom in which each pixel is convolved the pixel in a filter window with a dimension of M x N. Fig. 2 illustrates this approach which performs the operations from (3) to (5) iteratively [10]. That is,

$$y_s(r_b,c_b) = \sum_{i=1}^{M}\sum_{j=1}^{N} W_s(i,j) \times X_s(rb,cb)$$

$$\sum_{i=1}^{M}\sum_{j=1}^{N} W_s(i,j) \times X[(r-1)L + r_b + (M-1) - i, (c-1)L + c_b + (N-1) - j \quad 3$$

Where $Y_s\ (r_b,\ c_b)$ is the image of the S^{th} block after processing, $W_s\ (i,\ j)$ is the $(i,j)^{th}$ element in the weight matrix $Ws$of the S-th block. The error signal $e_s\ (r_b,c_b)$ is then obtained by subtracting the image $Y_s\ (r_b,c_b)$ from the primary input image $d_s\ (r_b,c_b)$.

$$e_s(r_b,c_b) = d_s(r_b,c_b) - y_s(r_b,c_b) \quad 4$$

The weight matrix W_{s+1} of $(S+1)^{th}$ block is then updated by

$$W_{s+1}(i,j) = W_s(i,j) + \frac{2}{L^2}\boxed{?}\sum_{r_b=1}^{L}\sum_{c_b=1}^{L} e_s\ (r_b,c_b) \times X(r_b + rL - i, c_b + cL - j) \quad 5$$

Where $\boxed{?}$ is convergence factor

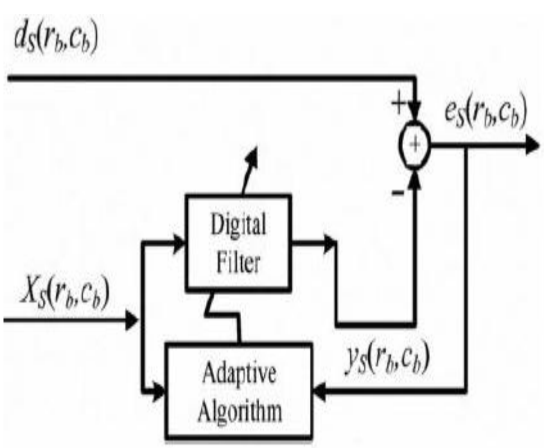

Fig. D adaptive filter for image noise cancellation

CONCLUSION –

The denoising of image is initial step in image processing. The quality of the denoised image depends on the two major parts: wavelet transform for decomposition of image and adaptive wiener filtering in wavelet domain and spatial domain. The block-adaptation phase performs the formal image noise cancellation by taking the suitable weight matrix in the weight- 35 training phase as the initial weight matrix, and performing the 30 TDBLMS algorithm. Hence from this comparative study it can be concluded that the results are improving but need more enhancement at the tackling noise along with blurring, Future endeavors include better expansible proportion of wavelet coefficients in order to get better denoising effects.

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