

## **Attenuation of Industrial Noise by FFT based Adaptive MDCT**

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**Abstract**—Industrial noise induced hearing loss is an increasingly prevalent disorder that is the result of exposure to high intensity sounds, especially over a long period of time. Noises of industry can cause partial deafness, interference with communication by speech and annoy. These undesirable effects are best avoided by reducing the noise to acceptable levels. Several investigations on industrial noise proved that industrial workers need at least 10-15 dB higher SNR (Signal to Noise Ratio) than the other places. The objective of this paper is to implement Fast Fourier Transformation (FFT) based Modified Discrete Cosine Transformation Least Mean Square (MDCT-LMS) to reduce the effect of industrial noise and to improve overall sound quality of industrial workers. The computer simulated results show superior convergence characteristics of the adaptive complex transformation algorithm by improving the SNR at least 11dB for input SNR's less than and equal to 0 dB, with excellent convergence ratio, better time and frequency characteristics. These results suggest that a headset with digital signal processing adaptive algorithm are useful for hearing protection in workplaces with high levels of wide band industrial noise.

**Keywords**— Industrial noise, Hearing protection, noise control, adaptive filter, SNR improvement, fast MCT-LMS.

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### **I. INTRODUCTION**

Industrial noise induced hearing loss is an increasingly prevalent disorder that is the result of exposure to high intensity sounds, especially over a long period of time. High-intensity noises are a health hazard for industrial workers, and hearing protection is necessary to prevent hearing loss. Hearing loss caused by occupational noise is one of our biggest industrial diseases. It is a disease that has been recognized since the Industrial Revolution. The conventional passive methods, such as ear muffs, are ineffective against low-frequency noise [3]. This problem can be effectively solved by using the adaptive algorithms for different frequencies [4].

Many researchers has stated that [7] noise can not only cause hearing impairment due to long-term exposures of over 85 dB, but it also acts as a causal factor for stress and raises systolic blood pressure. Additionally, it can be a causal factor in work accidents, both by masking hazards and warning signals, and by impeding concentration [12]. Noise also acts synergistically with other hazards to increase the risk of harm to workers [2]. [10] States that exposure to 85 dB of noise for more than eight hours per day can result in permanent hearing loss. Since decibels are based on a logarithmic scale, every 3 dB sound pressure level increase results in a doubling of intensity, meaning hearing loss can occur at a faster rate. Therefore, gradual developing industrial noise induced hearing loss occurs from the combination of sound intensity and duration of exposure.

Noise induced hearing problems are typically is centered at 4000 Hz. The louder the noise is, the shorter the safe amount of exposure is. Normally, the safe amount of exposure is reduced by a factor 2 for every additional 3 dB. For example, the safe daily exposure amount at 85 dB is 8 hours, while the safe exposure at 91 dB is only 2 hours [8], [9]. Sometimes, a factor 2 per 5 dB is used. Personal electronic audio devices, such as iPods, because iPods often reaching 115 decibels or higher. This can produce powerful enough sound to cause significant hearing loss in the workers, given that lesser intensities of even 70 dB can also cause hearing loss [11]. Different kinds of filtering methods are suggested in the literature for the minimization of noise in industries [5], [6]. However, through the proper use of ear protection, education, hearing conservation programs in the workplace, and audiological evaluations, industrial noise induced problems can be reduced [13].

The DCT is a technique that converts a spatial domain waveform into its constituent frequency components as represented by a set of coefficients. The DCT has good orthonormal, separable, and energy compaction property. Most of the signal information tends to be concentrated in a few low frequency components of the DCT. Although the DCT does not separate frequencies, it is a powerful signal decorrelator. It is a real valued function and thus can be effectively used in real-time operation. It is a close relative of DFT – a technique for converting a signal into elementary frequency components, and thus DCT can be computed with a Fast Fourier Transform. Unlike DFT, DCT is a real valued and provides a better approximation of a signal with fewer coefficients. The DCT is central to many kinds of signal processing. For non-stationary signals the DCT provides good approximation of a signal with fewer coefficients [15]. Hence MDCT-LMS algorithm is suited for non-stationary inputs like industrial noise and the convergence time is also less compare to direct LMS techniques and DFT-LMS algorithms.

### **II. MODIFIED DISCRETE COSINE TRANSFORM**

A block transform based on the DCT or DFT is equivalent to a filter bank consisting of multiple band pass filters. Such a filter bank is called as a Time Domain Aliasing Cancellation Filter bank (TDAC). Additionally this method provides an added advantage in the block processing of discrete time signals such as audio and speech. The next step in designing a fast algorithm based on a TDAC filter bank is to select the transform that will be used. A number of transforms, which have

a meaningful interpretation in the transform domain, are used in the digital processing of signals. More notably are the DFT and DCT. Of these it was found that the DCT offers maximum advantages [1], [15]. The basis set actually used in TDAC systems is slightly modified version of DCT-II and takes on the form:

$$X(k) = 2 \sum_{n=0}^{N-1} z(n) \cos\left(\frac{2\pi}{N}(n+n_0)\left(k+\frac{1}{2}\right)\right), \quad 0 \leq k \leq N/2 \quad 1$$

Where  $z(n)$  is the windowed input sequence  $z(n)$

$$x(n) = \frac{2}{N} \sum_{k=0}^{N/2-1} X(k) \cos\left(\frac{2\pi}{N}(n+n_0)\left(k+\frac{1}{2}\right)\right), \quad 0 \leq n \leq N \quad 2$$

Where,  $X(k)$  are the spectral co-efficient.  $n_0 = (N/2 + 1)/2$  in the equations 1 and 2. Such TDAC with the DCT as the transform is called as the MDCT.

### A. Fast algorithm for MDCT

The entire MDCT requires complexity of  $O(N^2)$ . Ideally we would like computation times logarithmic or at least linear in the size of the input block length to make the use of these transforms feasible in real time signals. This in turn motivates us to look for algorithm, which computes the MDCT and the IMDCT expressions as fast as efficiently as possible. This paper uses FFT based MDCT algorithm as described below.

**FFT based MDCT:** This algorithm is a variation of the algorithm by Lee. Fast algorithm uses an FFT like decimation to compute the MDCT in an efficient way. We start with the IMDCT expression,

$$x[n] = \sum_{k=0}^{(N/2)-1} X[k] \cos(\phi), \quad n = 0, 1, \dots, N-1$$

$$\text{Where } \phi = \frac{\pi}{2N} (2n+1 + N/2)(2k+1). \quad 3$$

The fast algorithm has derived by multiplying both sides of equation 3 by  $C_N[n] = \cos[\pi/2N(2n+1 + N/2)]$ , which is always nonzero for integer n. After some manipulation with the equation and separation of the output into two halves, we get a pair of equations as follows.

$$x[n] = \frac{1}{2C_N[n]} \sum_{k=0}^{(N/2)-1} X[k-1] + X[k]C_{N/2}[n, k], .$$

$$x\left[n + \frac{N}{2}\right] = \frac{1}{2C_N[n + N/2]} \sum_{k=0}^{(N/2)-1} (-1)^k [X[k-1] + X[k]]C_{N/2}[n, k], \quad n = 0, 1, \dots, \left(\frac{N}{2} - 1\right) \quad 4$$

Where  $C_{N/2}[n, k] = \cos[(\pi/2(N/2))(2n+1 + N/2)k]$

Further decomposition requires an additional manipulation similar to that of the decimation in time FFT. The whole process is repeated as many times as necessary to completely decompose the IMDCT into several sets of two point transforms. After applying the transformation, described above, normalizing and by passing through LMS filter gives us fast FFT based MDCT-LMS.

### B. Computational Complexity

Computational complexity of TDLMS is very high because of the complexity of transformation. Reducing the complexity of transformation can reduce this high complexity of TDLMS. This type of fast MDCT algorithm requires

$$\left(\frac{N}{4}(\log_2 N + 1)\right) \text{ real additions and } \left(\frac{N}{4}(3\log_2 N - 3)\right) \text{ real multiplications.}$$

**Table 1. Computational complexity for N=8.**

Type of transformation	Number of real additions	Number of real multiplications	Total complexity
Direct DCT	64	56	O(120)
Modified DCT	Forward MDCT 32 Inverse MDCT 32	Forward MDCT 28 Inverse MDCT 24	O(116)
Fast Factored DCT	13	29	O(42)
Fast Recursive MDCT	8	16	O(24)
Fast MDCT	8	12	O(20)

### III. PERFORMANCE EVALUATION

Performance of the adaptive filters are measured, compared and analyzed with the help of following parameters.

**a.** Convergence rate: The convergence rate determines the rate at which the filter converges to its resultant state. Usually faster convergence rate is the desired characteristic of an adaptive system. Convergence rate is not, however, independent of all other performance characteristics. If the convergence rate is increased, the stability characteristics will decrease, making the system more likely to diverge instead of converge to the proper solution. In this work, convergence rate is measured in terms of eigenvalue ratio.

**b.** Minimum mean square error (MSE): The MSE is a metric indicating how well a system can adapt to a given solution. A small minimum MSE is an indication that the adaptive system has accurately modeled, predicted, adapted and/or converged to a solution for the system.

**c.** Stability: Stability is probably the most important performance measure for the adaptive system. The algorithm convergence time and stability depends upon the ratio of the largest to the smallest eigenvalue associated with the correlation matrix of the input sequence. Therefore, stability of the algorithm is defined in terms of eigenvalue ratio.

**d.** Eigenvalue ratio: Eigenvalue ratio or the eigenvalue spread is the ratio between the maximum eigenvalue and the minimum eigenvalue of the input autocorrelation matrix. The eigenvalue ratio  $r$  can be calculated as

$$r = \frac{\lambda_{\max}}{\lambda_{\min}} \quad 5$$

Where  $\lambda_{\max}$  and  $\lambda_{\min}$  are the maximum and minimum eigenvalues, which found on the main diagonal of the autocorrelation matrix. Then the rate of convergence can be calculated as

$$C.rate = \frac{(r-1)^2}{(r+1)^2} \quad 6$$

From the above equation it is clear that, the convergence time decreases if the eigenvalue ratio increases and vice versa.

**e.** SNR: Amount of noise filtering can be measured from adaptive system with the help of input SNR and output SNR. Input SNR is the ratio between the power of input signal and power of noise at input. Output SNR is the ratio between the power of filtered signal and power of noise at output. In general SNR is defined as

$$SNR = \frac{\sum_n x^2(n)}{\sum_n e^2(n)} \quad \text{and} \quad SNR(dB) = 10 \log_{10} \frac{\sum_n x^2(n)}{\sum_n e^2(n)} \quad 7$$

Where,  $x(n)$  is the input signal and  $e(n)$  is the noise.

The algorithm is evaluated for different types of industrial noises with different SNR. In this work  $x(n)$  is the speech signal and  $e(n)$  is the industrial noise. Results show that, both parameters SNR and eigenvalue ratio are strongly depending on type of noise.

**Table 2. Simulated output of fast DCT-LMS Noise canceller**

SNR of the input signal	SNR of the output signal	Eigenvalue ratio
0 dB	11.0 dB	6.09
+5 dB	11.29 dB	5.44
+10dB	13.20 dB	5.6
-10 dB	10.2 dB	5.5

For different input SNR, the output SNR and eigenvalue ratios are calculated as shown in Table 2. The eigenvalue ratio is calculated to find out how well the algorithm converges to the optimum Wiener solution.

### IV. CONCLUSIONS

The performance of FFT based MDCT-LMS is same as MDCT-LMS and DCT-LMS. But the main advantage of FFT based MDCT is it's less computational complexity. This algorithm is excellent compared to NLMS and DFT-LMS algorithm in terms of convergence performance. The SNR improvement of FFT based MDCT-LMS is same as MDCT\_LMS and DCT-LMS. The eigenvalue ratio is 7 for zero dB and is very less compared to time domain adaptive methods and DFT-LMS noise reduction. Hence, this real transformed adaptive filter can quickly converge to the optimal solution.

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