ISSN (Print) : 2320 – 3765
ISSN (Online): 2278 – 8875

International Journal of Advanced Research in Electrical, Electronics and Instrumentation Engineering

(An ISO 3297: 2007 Certified Organization)

Vol. 5, Issue 9, September 2016

DCT based Performance Enhancement of Adaptive Filter for Audio Signals

Dr. Madhukar S. Chavan

Associate Professor, Dept. of E & Tc. Engg., P. V. P. Institute of Technology Budhgaon, Sangli, India.

ABSTRACT: Signal degradation due to noise in audio voice systems is today's major concern. The signal enhancement needs high performance adaptive filters. This paper proposes the use of discrete cosine transform for performance enhancement of adaptive filter. The audio signal is converted into DCT domain after applying LMS and RLS algorithms. Output of LMS- DCT combination is compared with output of RLS- DCT combination. Simulation result shows that LMS- DCT gives better performance.

KEYWORDS: Least Mean Square (LMS), Recursive Least Square (RLS), Discrete Cosine Transform (DCT), Adaptive Filters.

I.INTRODUCTION

In real time environment, speech signals are corrupted by several forms of noise like competing speakers, background noise, car noise, distortion caused by communication channels etc. In such situations extraction of high resolution signals is a key task. In this aspect, adaptive filtering comes in to the picture [1]. Adaptive filters have the ability to adjust their impulse response and filter out the correlated signal in the input. They require little or no prior knowledge of the signal and noise characteristics. The author has proposed LMS and RLS algorithms in discrete cosine transform domain for noise cancellation in speech signals. Discrete cosine transform is most favourable for signal compression and feature extraction. Use of DCT improves signal to noise ratio (SNR). If noise is mixed with speech or the signal is highly correlated then DCT exactly de-correlates the signal. The output consists of signal components separated in to individual frequency bands. DCT exhibits excellent energy compaction for highly correlated signals. The energy of the correlated signal is packed into the low frequency region. This improvement helps to reduce the residual noise present in the audio signals.

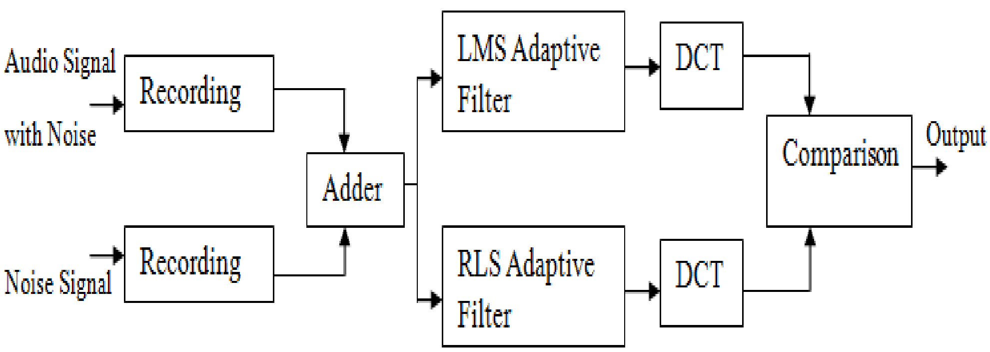


Fig.1 Adaptive filtering using DCT

Fig.1shows that the system requires two inputs: one is noisy audio signal and second input is external noise signal. The adaptive algorithms used are LMS and RLS algorithms. Outputs of LMS DCT and RLS DCT are compared. The result shows that LMS-DCT improves SNR.

ISSN (Print) : 2320 – 3765
ISSN (Online): 2278 – 8875

International Journal of Advanced Research in Electrical, Electronics and Instrumentation Engineering

(An ISO 3297: 2007 Certified Organization)

Vol. 5, Issue 9, September 2016

II.ADAPTIVE FILTER

Filters which adapt its coefficients with variations in input signal are adaptive filters. An adaptive filter makes its transfer function exact following to an optimizing algorithm. It evaluates the instantaneous values of the multiplier coefficients such that error function is minimized [3]. The error function is given by,

$e^{/}(n) = d(n) - y(n)$

Where d (n) is some desired response and y (n) is the filter output.

The design of adaptive filter involves choice of the filter structure, the objective function and adaption algorithm.

The LMS is simple algorithm used in the adaptive structures because it uses the error signal to calculate the filter coefficients. To calculate a signal that changes with time, LMS algorithm requires a gradient descent method. The LMS algorithm approaches the minimum of a function to minimize error by taking the negative gradient of the function. LMS based on steepest descent algorithm in which the weight vector is updated from sample to sample as follows:

$$w_{(k+1)} = w_k - \mu \nabla_k$$

Where w_k and ∇_k are the weight and the true gradient vectors, at the k^{th} sampling instant.

μ controls the stability and rate of convergence. The adaptive LMS equation is given by,

$$w_{(k+1)} = w_k + 2\mu * e_k * x_k$$

Where, $w_{(k+1)}$ are updated adaptive filter coefficients, x_k are the input values, e_k is the error value and μ is the step

size**.** Step size determines the convergence or divergence speed of the adaptive filter coefficients. The filter will converge fast, if step size is more but could diverge if μ is too large. The adaptation process is fast for large step size. The speed convergence and MSE has to be balanced while choosing step size.

Implementation of the Basic LMS algorithm

Step1: Initially, set each weight i = 0, 1… N-1, an arbitrary fixed value, such as 0.

For each subsequent sampling instant = 1, 2… carry out steps (2) and (4) below.

Step2: Compute filter output

$$\hat{n}_k = \sum_{i=0}^{N-1} w_k(i)\, x_{k-1}$$

Step3: Compute the error estimate

$$e_k = y_k - \hat{n}_k$$

Step4: Update the filter weights

$$w_{k+1}(i) = w_k(i) + 2\mu\, e_k x_{k-i}$$

.

For each new set of input and output samples, the LMS algorithm requires approximately 2N+1 multiplications and 2N+1 additions.

RLS adaptive filter algorithm repeatedly finds the filter coefficients that minimize weighted linear least squares cost function relating to the input signals. RLS algorithm requires more computational resources ($2.5N^2$+ 4N). The LMS algorithm has only step-size adjustable parameter that affects convergence rate. The RLS algorithm uses a least square method to estimate correlation directly from input data. All the information which is present in the input signal from the start of the adaptation process up to the present is used by RLS algorithm [4].

Implementation of the Basic RLS algorithm

Step1: Initialize tap weights to zero.

Step2: Find the error signal.

Step3: Compute the gain vector and Update the inverse of the correlation matrix.

Step4: Update the filter weights.

ISSN (Print) : 2320 – 3765
ISSN (Online): 2278 – 8875

International Journal of Advanced Research in Electrical, Electronics and Instrumentation Engineering

(An ISO 3297: 2007 Certified Organization)

Vol. 5, Issue 9, September 2016

III. DISCRETE COSINE TRANSFORM

Discrete Cosine Transform de-correlates signal influenced by noise. After decor relation without losing compression efficiency, each transform coefficient can be encoded independently. For audio signals i.e. one dimension signals redundancy is apparent in the sequence of samples. DCT only uses cosine wave to represent a signal. The DCT coefficients can be expressed as:

$$x_t = \sum_{k=0}^{M-1} Xkcos\, 2\pi kt/M$$

For operating on one-dimensional signals such as speech waveforms, one-dimensional DCT is used.

One-dimensional DCT

Common definition of a 1-D sequence of length M is,

$$c\ (u) = \alpha\ (u) \sum_{x=0}^{M-1} f(x) cos \frac{\pi(2x+1)u}{2M}$$

DCT is Fourier related transform similar to DFT but DCT coefficients are real values. This paper clearly depicts results of SNR calculation. It shows comparison of two algorithms with use of DCT [5], [6].

IV. SIMULATION RESULTS AND DISCUSSIONS

1. Actual signal and signal with noise

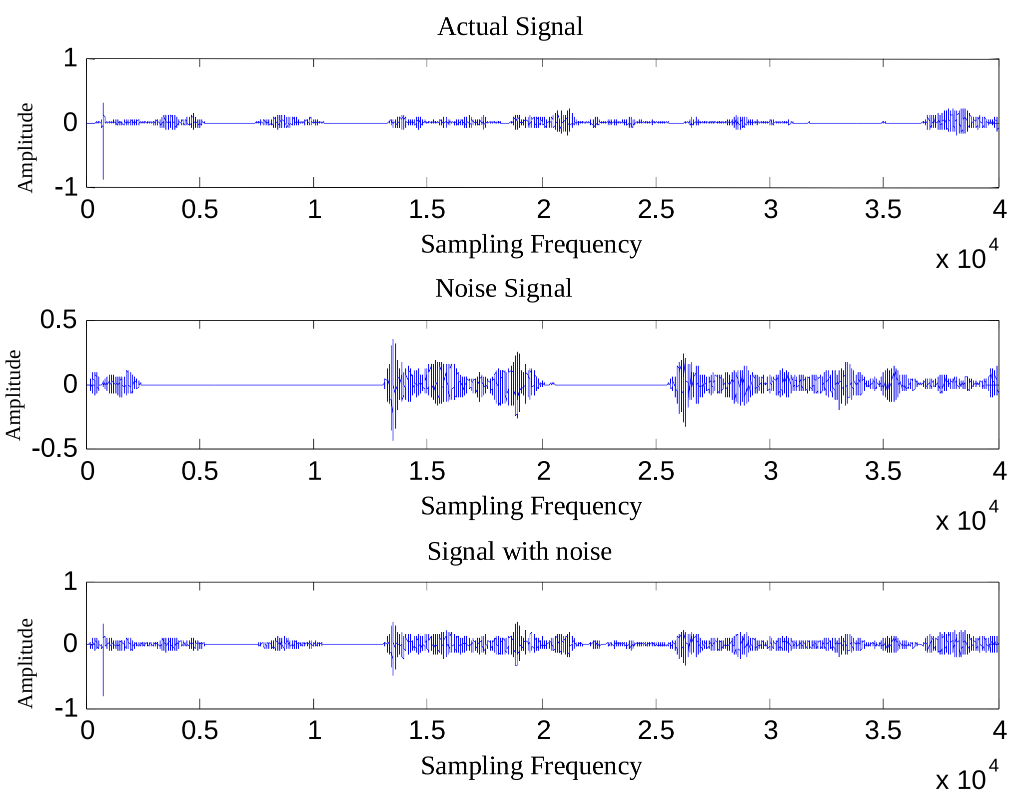


Fig.2 Original signal and signal with noise

ISSN (Print) : 2320 – 3765
ISSN (Online): 2278 – 8875

International Journal of Advanced Research in Electrical, Electronics and Instrumentation Engineering

(An ISO 3297: 2007 Certified Organization)

Vol. 5, Issue 9, September 2016

Fig. 2 Shows plot of actual signal and noise signal generated by using matlab functions. The noise is added to the actual signal and resulting noisy signal is also plotted. Sampling frequency of 8 kHz is used and signals are recorded for 5 seconds.

2. Outputs of LMS and RLS based adaptive filters

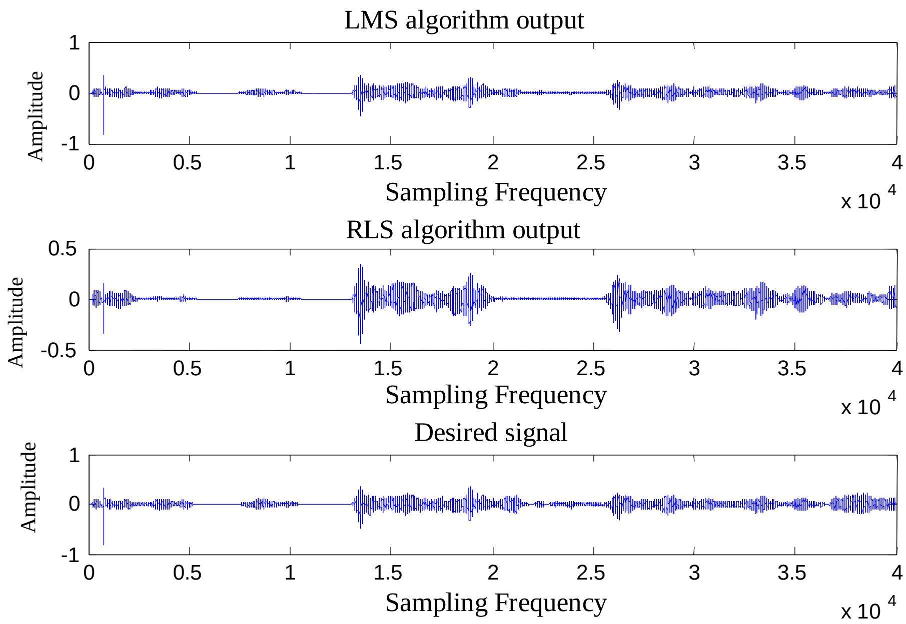


Fig.3 Outputs of LMS and RLS based adaptive filters

Fig.3 shows error output of LMS Adaptive filter and RLS Adaptive filter with desired signal.

3. Comparision of LMS and RLS algorithms

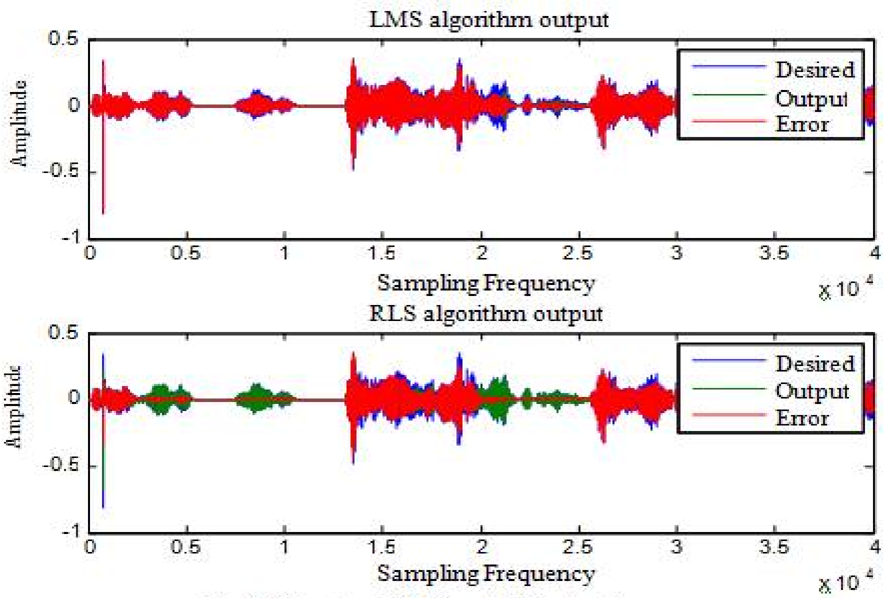


Fig.4 Outputs of LMS and RLS algorithms

ISSN (Print) : 2320 – 3765
ISSN (Online): 2278 – 8875

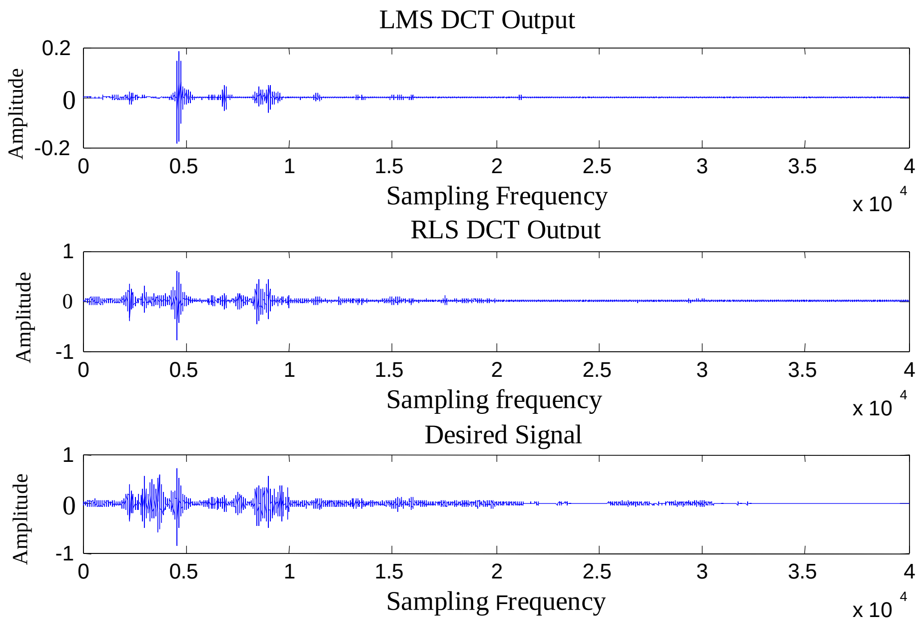
International Journal of Advanced Research in Electrical, Electronics and Instrumentation Engineering

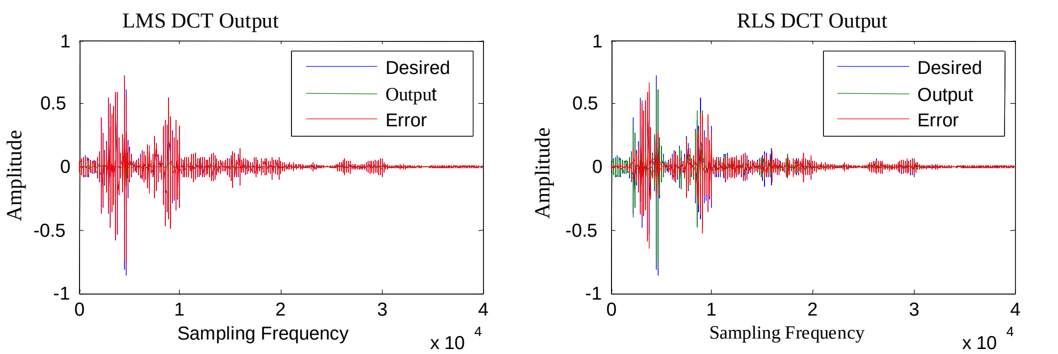
(An ISO 3297: 2007 Certified Organization)

Vol. 5, Issue 9, September 2016

Fig.4 shows outputs of LMS and RLS algorithms. The output of adaptive filter using RLS algorithm is more accurate than adaptive filter using LMS algorithm.

4. Outputs of LMS DCT and RLS DCT algorithms





SNR after RLS : -36.7089 (dB)

SNR after LMS : -20.9372 (dB)

SNR after RLS with DCT: -1.2252(dB)

SNR after LMS with DCT: 0.10627 (dB)

Fig.5 Outputs of LMS DCT and RLS DCT algorithms

Fig.5 shows outputs of LMS DCT and RLS DCT algorithms. SNR estimations are also done. SNR for the output of LMS and RLS adaptive filter is less than the SNR after applying DCT to the output of LMS and RLS adaptive filters. Comparison of outputs shows that LMS- DCT gives better performance.

ISSN (Print) : 2320 – 3765
ISSN (Online): 2278 – 8875

International Journal of Advanced Research in Electrical, Electronics and Instrumentation Engineering

(An ISO 3297: 2007 Certified Organization)

Vol. 5, Issue 9, September 2016

V. CONCLUSION

Simulation results shows that, outputs of LMS and RLS adaptive filter after applying DCT gives better results. All the studied algorithms increase SNR of noisy audio signals giving nearly close estimate of original signal. Use of DCT with adaptive filters is advantageous in SNR improvement. It is concluded that LMS algorithm can be easily used in almost all applications. Combination of LMS with DCT gives SNR improvement than that of individual DCT and LMS. But that is not the case with RLS algorithm. RLS DCT has better performance than only DCT. Due to computational complexity, instability and poor tracking of input signal of RLS algorithm, it may be costly and sometimes may be risky to use it. Comparison of outputs shows that LMS- DCT gives better performance.

REFERENCES

[1] Haykin Simon, "Adaptive Filter Theory", Prentice Hall, Upper Saddle River, New Jersey, 1996.
[2] Jebastine J., Sheela Rani B., "Implementation of block least mean square adaptive algorithm for effective noise cancellation in speech signal", International Journal of Electrical and Electronics Engineering Research Vol.1, Issue.1, 2011.
[3] Draghi ciu Nicolae, Reiz Romulus, "Noise Canceling in audio signal with adaptive filter", Volume 45, Number 6, 2004.
[4] Anjali Sadanand T. **"**Comparative Analysis of Various Adaptive Filters for Noise Cancellation", Proceedings of Fifth IRAJ International Conference, Pune, India, and ISBN: 978-93-82702-29-0, 15th September 2013.
[5] Anand Vartika, Shah Shalini, Sunil Kumar,"Performance Analysis of Various Noise Cancellation Methods", International Journal of Advanced Research in Electrical, Electronics and Instrumentation Engineering, Vol. 2, Issue 5, May 2013.
[6] Ferdouse Lilatul, Akhter Nasrin, and Jaigirdar F. T., "Simulation and Performance Analysis of Adaptive Filtering Algorithms in Noise Cancellation", IJCSI International Journal, Vol. 8, Issue 1, January 2011.
[7] Raghavaraju, Thota Bhavani, "Speech Enhancement using Combinational Adaptive Filtering Techniques", IJECT Vol. 5, Issue Spl-3, Jan-March 2014.
[8] Lakshmikanth S, Natraj K. R., Rekha R., "Noise Cancellation in Speech Signal Processing-A Review International Journal of Advanced Research in Computer and Communication Engineering Vol. 3, Issue 1, January 2014.
[9] Pranjali M. Awachat,.Godbole S.S., "A Design Approach for Noise Cancellation in Adaptive LMS Predictor using MATLAB", International Journal of Engineering Research and Applications (IJERA), Vol. 2, Issue 4, July-august 2012.
[10] Shajeesh. K. U., Soman K. P., "Noise Cancellation Method for Robust Speech Recognition", International Journal of Computer Applications Volume 45, No.11, May 2012.