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# Implementation of Blind Signal Dereverberation Of Speech Signals using Cuckoo-Independent Component Algorithm

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***Abstract***

Blind signal processing of convolutive combinations of unknown time series is asignificant building block in modern schemes connecting broadband signal acquisition bysensor arrays in multipath. Reverberation, a section of any sound produced in a natural atmosphere, can reduce speech intelligibility or more usually the quality of a signal formed within a room.The process of recovering the source signal by removing the unwanted reverberation is called dereverberation.Speech dialog systems interact with the user by recognizing and interpreting the meaning of the received commands.The ICA goes to the category of blind source severance (BSS) and the ICA prominently determined by the key assumption of the physical world character. The BSS estimates the original signal using the mixed-signal information observed from the input channel. The proposed method avoids the drawback of separated sounds with improper localization, directivity and spatial quality of separate sources. The wavelet filter and multi-step linear prediction coding (mLPC)for extraction of coefficients in the late reverberation. The reverberated signals are eliminated using backward differentiation. Cuckoo is an optimization technique is used to improve the effectiveness of ICA. To reduce the redundant bit and hardware cost the new FPGA was proposed to improve the reliability. Hence the reliability and SNR values are increased. Various methods are compared with proposed Cuckoo search algorithm to demonstrate the efficiency of frequency, time delay and power consumption with reduced area utilization.

***Keywords;*** *Blind Speech Signal Dereverberation, CUCKOO-ICA Algorithm, power consumption, SNR values, and FPGA Implementation.*

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# 1. INTRODUCTION

Blind source separation using different methods is an important and trending technology due to its good budding tenders in signal processing such as medical image processing, telecommunication and speech signal processing. One of the methods is Independent Component Analysis (ICA) in which the reflected components of all multi-source signals are separated into additive sub-elements. Basically, a receiver may receive multiple source components with the linear mixture of reflected signals. When all signals are mixed with each other it is critical to predicting the actual speech signal, which can be handled by using ICA algorithm. On the other hand, ICA reduces the higher-order static components and always try to recover the independent components from the mixed signal. Effective outcomes in EEG, fMRI, discourse acknowledgment and confront acknowledgment frameworks demonstrate the control and idealistic trust within the modern worldview. Along with this method, an optimization algorithm called as Cuckoo search algorithm was proposed in this chapter to upgrade the selection strategy of individual components. Concept behind this Cuckoo algorithm is to choose the best coefficients from the adaptive equalization unit.

Acoustic disturbances on the speaker's side or a poor transmission path may severely degrade the intelligibility of the received signal. Most digital devices hence rely on signal processing algorithms to enhance the speech quality for the listener. The natural ability of humans to express themselves with their voice has further triggered the development of human-machine interfaces, which allow the control of technical devices using voice commands. Speech dialog systems interact with the user by recognizing and interpreting the meaning of the received commands. With the increased computational power of today's mobile devices, these systems are now finding their way into everyday applications. In practice, the recognition performance severely suffers in difficult acoustic conditions. The received signals hence need to be processed with speech enhancement algorithms to ensure accurate and reliable recognition results. While such an operation largely increases the usability and acceptance of these systems, the received speech signals at the distant microphone are usually subject to various disturbances caused by the acoustic environment.

As a result, the received signals do not only comprise direct-path components but also time-delayed and decreased replications of the source signal, which are perceived as room reverberation. The effects of reverberation are usually negligible when the distance among the source and the microphone is small, but they become more pronounced as the source-microphone distance increases. Dereverberation is generally considered to be a difficult task due to the fact that neither the source signal nor the transmission paths between the source and the microphones are known. As a result, the clean speech signal has to be blindly estimated solely from the available microphone observations. Such estimation is complicated by the nature of the acoustic transmission paths, which are usually described by room impulse responses that may comprise several thousand coefficients. Since the corresponding coefficients quickly vary in the case of moving sources or if the acoustic environment changes, they have to be continuously tracked. In noisy conditions, tracking of the acoustic environment is particularly challenging since the noise, which may change its characteristics as well, usually causes identification errors.

The major limitation associated with the speech signal is its improvement to attain better clarity of voices. The major factor that affects the speech signal is reverberation. The reverberation is caused in a confined space due to the obstacles and walls. The reverberation is a preliminary factor that affects and degrades the speech signal quality when it is recorded or heard from a distance in a closed space. It also degrades the intelligence of the speech and makes limited applicability in speech applications [3, 2].

A simple multi-microphone speech dereverberation system is that the delay-and-sum beamformer [1, 2]. The dereverberation is achieved


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by a straightforward be an average via the detector outputs, deferred thus on focus within the way of the required speaker. The way of entrance is mostly tailored employing a 2nd-order datum method.

The main objective of the study is to improve the blind speech dereverberation over the reverberated speech signal, which affects the quality of the signal. The technique used in the present study is used for improving the multi-channel blind speech dereverberation for late and early reflections. This work proposed to eliminate the effects of early reverberation in speech signals using blind signal separation. The ICA is carried out to eliminate the effects of echo cancellation and blind dereverberation at similar instances. To eliminate the effects of late reverberation in the speech signal, a blind speech dereverberation framework is proposed. Here, wavelet filters are used to avoid whitening problem and backward differentiation technique is used to improve the segregation quality of the image. To provide a comparison between the proposed techniques and conventional technique associated with the speech quality parameters.

FPGA implementation of this technique offers additional upshot on this chapter. The ICA and Cuckoo search algorithms are utilized to implement the source separation from the mixture of nonlinear signals. Resource utilization, time delay, and power consumption parameters are weighted in the final outcomes of this chapter. FPGA is implemented in XILINX Virtex-7.

## 2. LITERATURE REVIEW

In [1], proposed that to remove reverberation by improving the robustness for automatic speech recognition. Humans infrequently experience difficulties in speech reverberant surroundings but the automatic speech identification system leads to increased error rates. This survey the part of perfect acoustic structures driven by human speech observation for creating ASR schemes. The 2-stage scheme to contract with distracting properties of noise and reverberation unconnectedly. The distortions are derisible to speech quality and it reduces the functioning of automatic speech appreciation schemes. The denoising and dereverberation are showed by using DNN, and two DNN subsystems are used.

In [2], proposed a 2-stage approach to improve corrupted speech. In practical, speech reaching our ears is spoiled by room consequence and background sounds. These disturbances are harmful to speech transparency and superiority, also which supposed to have a critical issue to many applications. In this algorithm improved the unbiased metrics of speech comprehensibility and superiority expressively. The technique for adaptive speech dereverberation and speaker-position change detection. These techniques have two strategies that are followed which are weighted recursive least squares process which helps to estimate RRCs and the second technique is used to which have not previously been addressed. The second strategy recognizes variations in speaker location. Location is comprehended by identifying time outlines where DE reverberated discourse control is atypically amplified. The test comes about appeared that the proposed strategy achieved merging in 5 sec and effectively-identified variations in speaker location.

In [3], proposed noise dismissal execution of a speech-activity-based Wiener channel capable to manage with coherent and diffuse noise. Tests are achieved with CMU genuine multichannel database, incorporates a noiseless speech reference over the head-mounted receiver. The reference flag is moreover utilized to complete re-enactment tests in controlled situations. Broad comes about have been gotten, both LAR and cepstral separations, and with seg SNR changes, surveying the capacities of the modern framework to manage together with resonation and coherent and diffuse commotion totally several acoustic situations. The dereverberation system that uses two microphones and GSC. The GSC is used to create the two signals. Standard delay yields the first signal and the second signal is constructed by blocked the direct speech signal which is called a reference signal. To evaluate the outstanding reverberation in the delay output and delay output are compared. The yield of delay and sum beamformer is improved by means of a spectral improvement technique. Researchers are developed in room impulse responses and simulation was taken to show significant reverberation reduction with low noise.

In [4], proposed an innovative microphone array technique to improve speech signals in a deafening reverberant atmosphere. Speech signal localization has some features such as time delay estimation. The localization strategy strength in tall commotion stages is given by sub-band Kurtosis-weighted edifice. The assessed inter-sensor time-delays are straightforwardly utilized in a versatile soft-constrained sub-band beamformer. Assessment in the mimicked atmosphere through genuine discourse groupings appears promising outcomes. A conventional based linear prediction method, the coefficients are not clearly chosen when there is a noisy background, so an optimized linear coefficient is used in the microphone signal. The signal and their constraints are efficient automatically using the maximization algorithm. The standard Kalman smoother employments a time-invariant covariance framework as a state-transition covariance lattice. So, this strategy employments this lattice and empower it to reach the time-varying discourse features. The constraints are upgraded so that Q work is expanded within the expansion stage. Investigational comes about to appear that this strategy prevalent in ordinary strategies beneath boisterous conditions.

In [5], proposed a Shrinkage based empirical mode decomposition which processing a sub-band and to remove the reverberation. The algorithm here used is a multistage procedure that has one microphone. Initially, EMD calculations have been utilized to break down a noisy reverberant speech flag into its oscillatory portions adaptively coming about in apparatuses termed as Intrinsic Mode Functions (IMF). At that point, EMD based contraction strategy has been utilized to IMF in arrange to decrease commotion, taken after thorough dereverberation of these denoised IMF machinery utilizing Ghostly deletion. In conclusion, it will accomplish an upgraded flag through the reproduction instrument from the prepared IMFs. On the estimation premise of SNR, results are compared to the relaxed state of craftsmanship approach and improvement within the quality of discourse signal was watched.

In [6], proposed a multichannel wideband signal dereverberation and improvement technique uses blindly assessed IRs connecting the signal source and Multiple Input Inverse Filter (MINT) is formed by a sensor and enhancement with stretched GSC to cancel the unwanted mixed signal. The extended GSC plays an important role, it not only removes the reverberation signal present in the input signal also removes ISI and noise. The resultant GSC beamformer not only has faultless dereverberation but also extreme noise suppression execution with a minor no of sensors. Implementation outcomes display efficiency of the GSC technique. An auto-relation supported MIMO converse filtering calculation for the converse filtering of room acoustics in discourse dereverberation. In A-RAM, we unused the connection among gotten reverberant signals and the acoustic motivation reactions. This comes about in an auto-relation which is at that point utilized as an imperative for the A-MINT calculation. Recreation comes about, utilizing both manufactured and recorded room motivation reactions, appear that proposed A-RAM accomplishes quick meetings related to A-MINT.

In [7], proposed 6 VARIOUS single-channel dereverberation procedures in the casual environment to account for various kinds of distortions. The final result is analyzed based on speech excellence and speech fluency. To determine the excellence and intelligibility, the correlation between the signal was determined and the performance is analyzed based on the correlations between the speech signal. A partial multichannel equalization using P-MINT. In arrange to advance increment strength against channel assessment

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errors, binary expansions are proposed, i.e. consolidation of a regularization parameter within the converse filter plan and a condensed particular esteem deterioration method. The test comes about for discourse dereverberation appear that regularized P-MINT strategy gives better outcomes state-of-the-art strategies such as channel shortening and loose multichannel least-squares strategy. The speech signal reverberation based on the speech reception threshold in various reverberation times. To obtain a quality measurement, ITU-T P.835 recommendations must need which comprised the valuation of the qualities: reverberant, colored, slanted, and overall superiority. Many calculations progressed, speech intelligibility for brief resonation times related to the reverberant state.

In [8], explained the application of versatile postfiltering for the upgrade of reverberant discourse is utilized in Code Energized Straight Expectation (CELP) discourse coding to lower the effect of quantization commotion within the excitation flag and the ghastly envelope. To enhance speech with accurate the underlying additive noise model is enough, it is achieved by an adaptive filter, in which the amplitudes of the unwanted signal peak are attenuated and the amplitude of the original speech is highlighted. Any channel procedure containing a modest computational complication but with the help of this filtering complexity is reduced by alternating the high peak noise signal. Both calculations are intended for consuming a direct computational complication. Tests consume appeared that this method is able to diminish initial resonation and weaken the 'distance-effect' emerging.

In [9], explained two state-space show in acoustic resounds diminishment, For the primary one, resound channels are respected as a time-varying state vector for the moment one quiet multichannel discourse signals are respected as a state trajectory which is valuable in a loud environment to overhaul the resonation channel coefficient. Variational Bayes system and 2 Kalman smoother depends constraint optimization phases are achieved on other hands. At last, the information is assessed by utilizing recorded information in a genuine teleconferencing room. A blind multi-microphone discourse dereverberation depends on the weighted forecast blunder strategy, where the reverberant perceptions are displayed utilizing multi-channel direct expectation within the short-time Fourier change space. Rather than consuming usually utilized Gaussian conveyance for specified discourse signals, these approach employments a Laplacian dispersion which is identified to be more precise in displaying discourse signals. Maximum-likelihood assessment is utilized for evaluating the demonstrate parameters, driving to a straight programming optimization difficult. The test comes about with reverberant discourse show that the execution of the planned calculation is significantly way improved when related to current dereverberation calculations.

In [10], proposed a two-stage daze dereverberation plot for reverberant discourse upgrade in which the flag is separated into distinctive subgroups. Within the first arrange, the reverse filters are evaluated utilizing the blind MINT, whereas within the moment arrange this strategy massacres late reverberant vitality by deducting control range of late motivation workings from control range of reverse filtered discourse flag. Exploratory comes about with reverberant discourse demonstrate that execution of calculation is considerably way improved when related to current dereverberation calculations. The major application of this method is for a sphere-shaped microphone array. The calculation utilizes inadequate recuperation, a compressed detecting strategy, to assess the location of the objective flag and its initial reflections. The execution of planned strategy is assessed utilizing computer re-enactment and our comes about show viability of the planned dereverberation calculation.

In [11], proposed a dereverberation calculation to successfully expel late resonation from watched sound signals. Meanwhile, such late resonation is preeminent negative to the execution of modified discourse affirmation, the calculation is


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predictable to create execution of ASR execution when a far-off receiver is utilized to denote an objective discourse flag. The footages were achieved for 4 dissimilar microphone-loudspeakers the partition between them is 0.5, 1, 1.5 and 2 m.

In [12], introduced a modern multi-channel discourse dereverberation calculation depends on measurable demonstrate of late resonation. flag. To evacuate the clamor spatial averaging was utilized. The re-enactment comes about appeared a diminish in resonation and twisting when utilizing more mouthpieces. Due to the utilization of spatial sifting, the point by point characteristics of the sound flag is mostly restored. (DimitriNion Et.al 2005) displayed a frequency-domain method based on a parallel Calculate (PARAFAC) examination that accomplished a multichannel daze source partition (BSS) of discourse blends. PARAFAC calculations were shared with a dimensionality decrease step to altogether condensed computational complexity.

## 3. INDEPENDENT COMPONENT ANALYSIS (ICA)

Independent component analysis (ICA) may be anapproach for observing essential variables or apparatuses from multi-dimensional arithmetical info. A free component examination is utilized to perform the division by calculating the division lattice values. This works based on considering that the source signals are autonomous so it was chosen ICA examination. There are a few strategies to execute ICA investigation, the esteem of sources can be extricated from the blend signals. After, changing over flag into the frequency domain for preprocessing the ICA weighting of the flag is performed. Typically, moreover known as the half portion in ICA, where the relationship between the two signals is removed and the signals are autonomous. This is by calculating the eigenvectors of the covariance lattice of the blended flag and the network with eigenvectors.

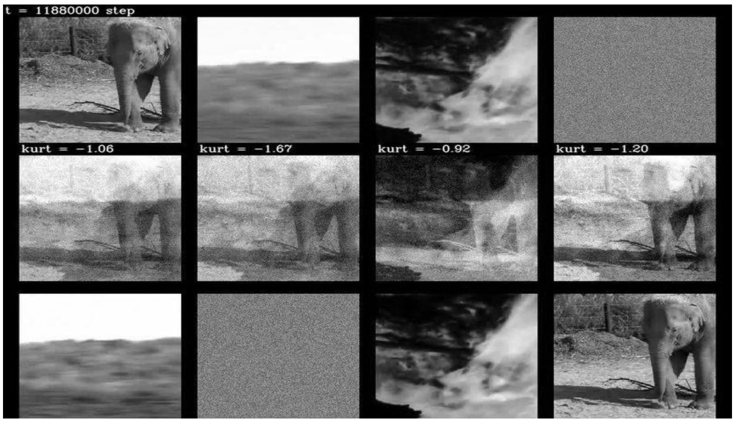


**Figure 1: Example of ICA on randomly mixed images**

In figure 1, four input images are merged together thereby results froma mixture of different source signals with low resolution.

### 3.1 CS Parameters Optimization ICA

#### 3.1.1 Cuckoo Search Algorithm

Cuckoo Search is another metaheuristic algorithm for tackling an enhancement issue. Cuckoo is a Blood Parasites; they never build their own specific home and lay their eggs in the home of another host winged creature or species. Each species of cuckoo builds up its very own system to grow the bring forth likelihood of its own eggs. It works on three rules:

(i) Each cuckoo lay one egg at some random minute and dumps its egg in a haphazardly picked home.

(ii)The best egg with the high caliber of eggs will continue to the people to come.

(iii) A Number of assessable hosts homes are settled, an egg laid by a cuckoo are found by the host winged animal with likelihood Pa £ (0, 1)

The point of using cuckoo search calculation is to have better arrangements that are in the settled host. In view of the over three a standard, the CS calculation invigorates the flying creature's best area search way, and they communicated as pursues,

$$X_i^{t+1} = X_i^t + \alpha \oplus \text{Levy} \quad \text{……….} \ (1)$$

Where $X_i^t$ represents the position of the $i^{th}$ nest at repetition t, $\oplus$ represent entry-wise multiplication, α is a stepwise parameter, L is the


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Levy flight utilized for a random walk, $Xi^{t+1}$ is generated by utilizing levy fight.

$$x_i^{t+1} = x_i^{(t)} + \alpha \oplus \text{Levy}(\lambda) \quad \text{…. (2)}$$

In the event that α>0 is the progression estimate which depends upon the kind of issues, for most cases, α=1 is chosen. Toll flight is essentially an arbitrary walk. Duty can be communicated as,

$$\text{Levy}=0.01\times\frac{\mu}{|\nu|^{1/\beta}}\times (g_{best} - x_i^t) \qquad \text{…(3)}$$

Where μ,ν - normal distribution, $g_{best}$ - best nest.

$$\mu\sim M (0,\delta_\mu^2),\nu\sim M (0,\delta_\nu^2)$$

$$\delta_\mu = \left\{\frac{\Gamma(1+\beta)\sin(\pi\beta/2)}{\Gamma[(1+\beta)/2]\beta 2(\beta-1)/2}\right\}^{\beta/2} \qquad \text{….(4)}$$

Where β=1.5

CS Algorithm simply has two parameters (M and $P_a$). M is settled and $P_a$controls the balance amid random and local search.

### 3.1.2 CS Based on ICA Parameters Optimization

In this method, we are using, ICA technique, adaptive equalization, and Cuckoo algorithm. These three strategies are used to achieve the speech signal dereverberation with best performance compared with other techniques. The parameters are evaluated based on power and SNR, resource utilization, finally total proposed work is implemented in FPGA system. Field Programmable Gate Arrays (FPGAs) are semiconductor devices that are created near a matrix of configurable logic blocks (CLBs) joined through programmable interconnects. FPGAs can be reprogrammed to a chosen appliance or functionality necessities later manufacturing.

This characteristic differentiates FPGAs from Application-Specific Integrated Circuits (ASICs), which are custom factory-made for specific design tasks. Though one-time programmable (OTP) FPGAs are presented, the leading categories are SRAM based which can be reprogrammed.

The regular process of CS-ICA is illuminedexposed in figure 2. The CS algorithm is utilized to optimize the ICA parameters C and γ as take after,

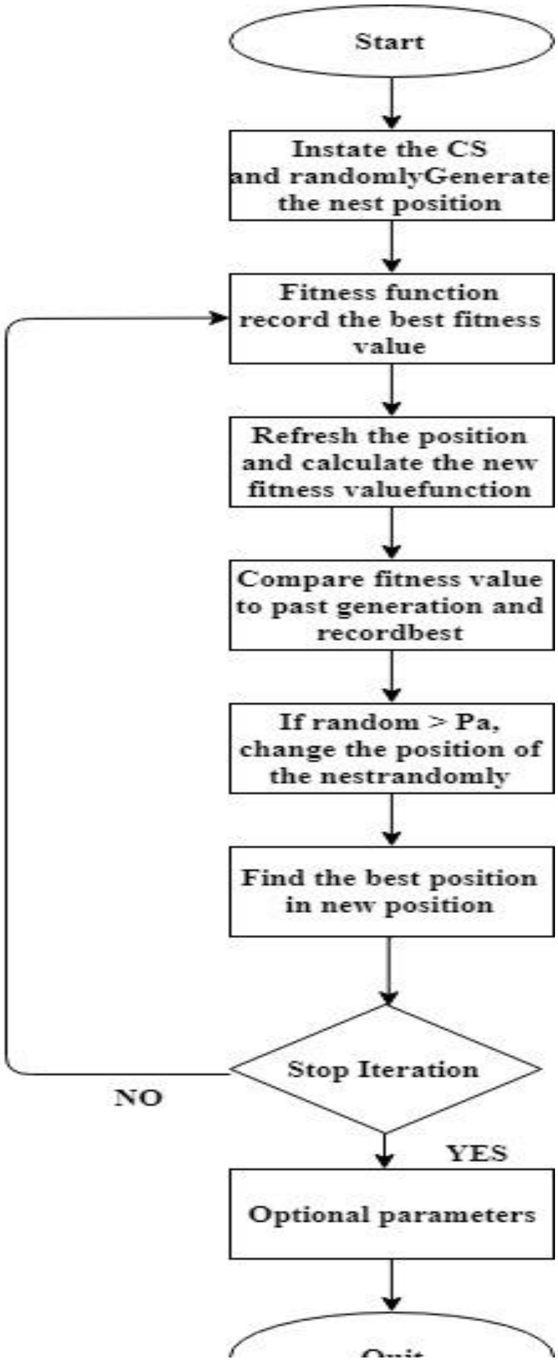


**Figure 2: Flow diagram of CS algorithm for ICA Parameter**

1. Initiate the CS algorithm and set the number of the nest (M), probability parameters ($P_a$), the maximum iterations ($t_{max}$), and the ranges of C andγ.
2. Randomly produce the nest position by using $q_1^0 = [x_1^0, x_2^0 \ldots x_n^0]^T$. Each nest relates to a set of parameters (C,γ). The fitness evaluation function is defined as follows,

$$I=\sum_{i=1}^{n}(Y^\wedge(i) - Y(i)^2/n$$



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Where, Y(i) is the actual value and Y^(i) is the prediction value, and n is the number of preparing samples.

3. Compute the fitness value of each nest to discover the present best solution,
4. Record the lowest fitness value and its corresponding position.
5. Keep the best solutions from the past generation. Record the position of other nests and calculate the fitness values.
6. Obtain a better solution for the previous generation which is enhanced than that of the previous generation.
7. Keeps an account of the position of the best nest.
8. Initialize a random number as the probability of egg position.
9. Contrast it with Pa.
10. If random > Pa, alter the location of the nest randomly to obtain a new set of locations.
11. Obtain the best nest location in step (5).
12. Stop searching when the highest iteration limit is reached.
13. output the best position to achieve the optimal parameter value; otherwise, return to step (3).

## 4. PROPOSED METHOD

The mixed signals are implied as $Y_i$ recorded through the microphones. The process of signal separation and dereverberation is explained in following steps.

- **Time to Frequency Conversion**

Signals in time instants are converted to frequency domain signals. The conversion is an offer by using the FFT. The conversion of signals from time to a frequency domain makes the process in a frequency-domain BSS approach. After, this signal is whitened for preprocessing of ICA.

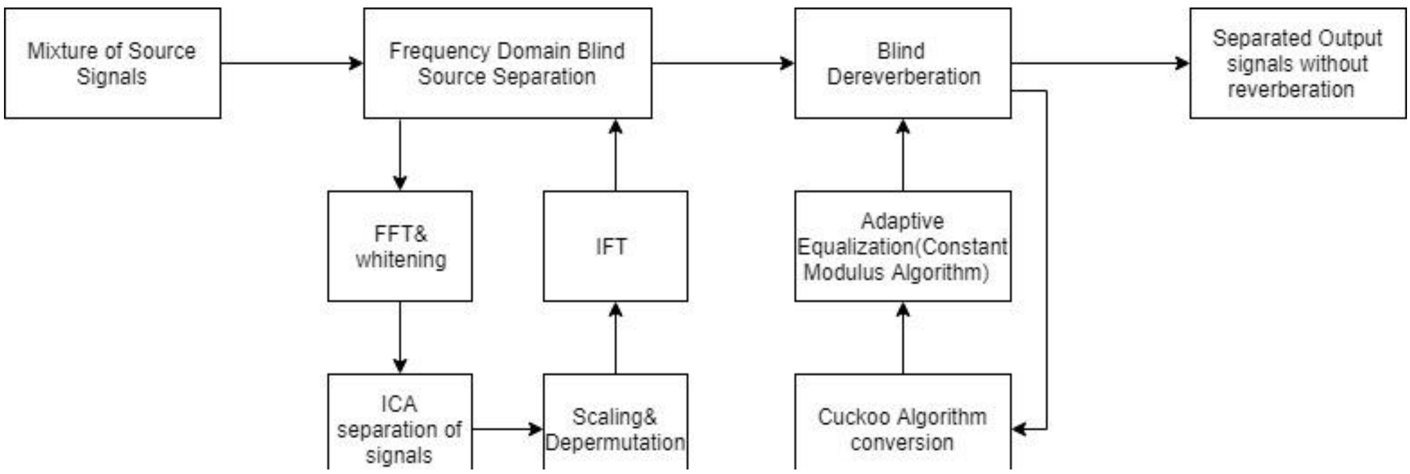


**Figure 3: proposed ICA with Cuckoo algorithm**

- **Source Separation**

Signals are separated by using the ICA analysis. Source signals from the different sources mix together and have to perform separation of signals without knowledge of source signal. This can be known as the Blind separation. After separation of the signals again it is converted to the time signals by using IFT.

- **Alignment of Signals**

The separated outputs should be aligned by scaling and permutation. Otherwise, signals will be in the form of a shuffled manner.

- **Dereverberation**

To remove the reverberant effect of the separated signals by adaptive equalization technique the CMA Algorithm is employed. After source separation of signals, this forms to remove the effect of echoes and reverberation. This is called blind dereverberation. The general adaptive filtering



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method in which the digital filter carries filtering on the original signal, produce afiltered signal.

- **Cuckoo Optimization**

Cuckoo is an optimization technique is used to improve the effectiveness of ICA. The de-reverberated signals are initially fed into the Cuckoo algorithm convertor in order to lay down the coefficients. The random n number of components have arrived at the convertor with arbitrary signal estimation values. The error estimated signal is now adapted with respect to the channel characteristics by providing a training sequence. A known value of training sequence is transmitted to the receiver through the channel to estimate the channel parameters. By using known transmitter value and the predicted receiver value the channel characteristics are evaluated. With the help of predicted channel parameter, the unknown source corresponding to the original speech signal is estimated thoroughly.

- **Adaptive Equalization**

Thusly, adaptive filter automatically goes on a proposition based on the quality of the original signal and the required signal. This strategy of an adaptive filter can be altered to the global place by these signals. The CMA has predominantly utilized calculation in adaptive filtering.

It is an inclination plunge calculation; it controls the adaptive filter taps directing them by a sum similar to the quick surmised to the slope from the mistake surface. Its iterative procedure incorporates registering the yield of a Finite Impulse Response (FIR) channel framed by a lot of channel coefficients.

- **Source Signal Separation**

The shuffled speech signals are estimated via channel estimator or adaptive equalizer. At the output node, multi-reflected components or reverberated speech signals are received and it might be eradicating the unwanted linear mixture of signals. The de-reverberated signal is finally obtained at the output with less amount of noise.

- **FPGA Implementation**

The Cuckoo search optimization implemented in FPGA using Virtex-7 2000T FPGA. This is to improve the low power consumption of about 50 percentage than the previous Virtex-6 algorithms. In addition, Virtex-7 2000T FPGA has twice its memory transfer speed compared to past era Virtex FPGAs with 1866 Mbit/s memory meddle execution and over two million rationale cells. It is well-known that the FPGA usage of the Quick ICA calculation is carried out utilizing XILINX virtex5-XC5VLX50t FPGA chip. The LX50t chip has predominant speed and a bigger zone over the other virtex5 family. The plan is executed utilizing VHDL dialect.

Since the framework is outlined to account for higher arrange information (four sensors), chain of command is embraced all through the plan to supply distant better; a much better; a higher; a stronger; an improved">a much better control over the generally equipment structure and to screen the flood and sub-current of each square. Besides, execution of DSP frameworks utilizing floating-point math requires a tremendous equipment range and may lead to a wasteful plan particularly for FPGA usage, on the other hand, fixed-point representation comes about in a productive equipment plan.

In this proposal, two’s complement fixed-point math is utilized. The word length was chosen based on a few recreation endeavors. Most of the come about were flawed when a little word length was utilized since the little word length was not adequate to speak to the values. After a few simulations’ endeavors, the choice of the word length was chosen not to be the same for different usage pieces. For illustration, the QR deterioration piece, the I/O and the middle signals word lengths were set to (26:13) which shows 26 bits with 13 bits speaking to the numbers portion and 13 bits speaking to the fragmentary bits. This way, the numbers portion can speak to numbers within the run of 2^13 = 8192. For the Centering and Covariance squares, the word length was set to 16 bits since the calculation of the Centering and the Covariance were

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not complex and 16 bits were sufficient to speak to for the middle factors like signals and capacity components inside the execution pieces.

## 5. RESULT AND DISCUSSION

The error estimation is calculated using output signal $y(n)$ as following equation, an Error signal is,

$$e(n) = (|y(n)|^2 - 1)^2 \quad ....(5)$$

Where, $y(n)$ is the output of the filter and $e(n)$ is the error output.

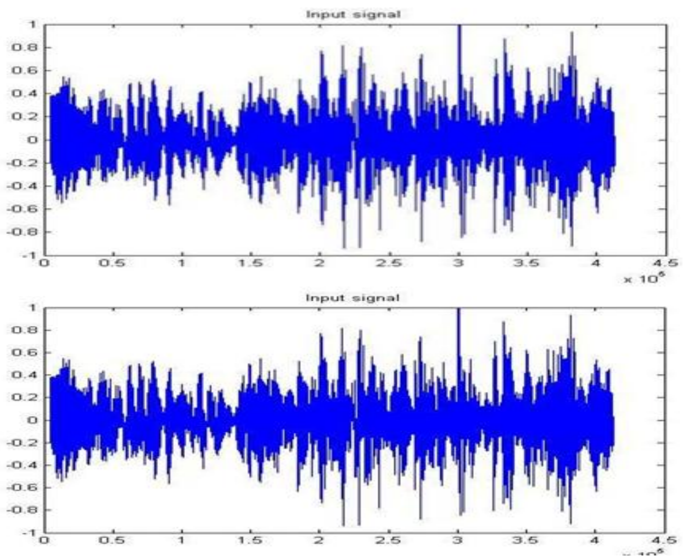

**Figure 4: Input Speech Signals**

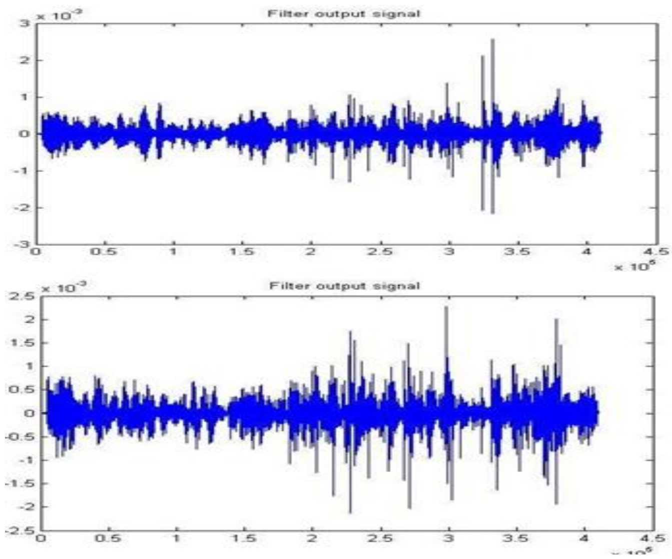

**Figure 6: Filtered Output Signals**

### 5.1 Signal to Noise Ratio (SNR)

SNR stands for signal to noise ratio, it is describedas the ratio between signal power to the noise power, which is used to estimate the performance evaluation using the following expression, with input SNR and output SNR, as:

$$SNR_o = 10\ log_{10} \frac{\sum_j [B_j^2]}{\sum_j [B_j - O_j]^2} \quad ... \quad (6)$$

Where, $O_j$ is the output signal and $B_j$ is the resynthesized signal. The original SNR is given as,

$$\Delta SNR = SNR_o - SNR_i \quad .... (7)$$

**Table 1: SNR Comparison**

| **Window size** | $\boldsymbol{SNR_i}$ | $\boldsymbol{SNR_o}$ | **LSNR** | **LSNR** |
|---|---|---|---|---|
| $2^8$ | 1.21 | 7.81 | 6.60 | 6.39 |
| $2^9$ | 0.98 | 8.12 | 7.14 | 6.70 |
| $2^{10}$ | 1.2 | 7.9 | 6.7 | 6.39 |
| $2^{11}$ | 1.2 | 6.91 | 5.71 | 5.53 |
| FFT | $SNR_i$ | $SNR_o$ | LSNR | LSNR |
| $2^9$ | 1.04 | 7.92 | 6.88 | 6.34 |
| $2^{10}$ | 1.02 | 7.94 | 6.92 | 6.69 |
| $2^{11}$ | 1.1 | 8.31 | 7.21 | 6.70 |
| **Reverberation Time** | $\boldsymbol{SNR_i}$ | $\boldsymbol{SNR_o}$ | **LSNR** | **LSNR** |
| 40 | 1.21 | 14.21 | 13.00 | 12.16 |
| 60 | 1.22 | 12.41 | 11.19 | 9.95 |
| 80 | 1.18 | 10.51 | 9.33 | 8.49 |
| 100 | 1.14 | 8.41 | 7.27 | 6.70 |



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| | | | | |
|---|---|---|---|---|
| 120 | 1.07 | 7.81 | 6.74 | 5.64 |
| 140 | 0.98 | 6.01 | 5.03 | 4.88 |
| **Noise(dB)** | $SNR_i$ | $SNR_o$ | **LSNR** | **LSNR** |
| -10 | 1.04 | 7.2 | 6.16 | 5.9 |
| -20 | 1.12 | 7.72 | 6.6 | 6.5 |
| -30 | 1.13 | 7.81 | 6.68 | 6.55 |
| -40 | 1.12 | 7.91 | 6.79 | 6.59 |

**5.2 Area, power and time delay:**

During the implementation of FPGA, it is important to consider some parameters such as resource utilization, power consumption of the device and frequency utilization.

The various performance parametric assigned to proposed Cuckoo ICA (Independent Component Analysis) of FPGA implementation is tabulated as shown below,

**Table 2: Comparison Results of Different ICA Implementations**

| Technologyof ICA | FPGA | FPGA | FPGA | Cuckoo + ICA |
|---|---|---|---|---|
| Corresponding Algorithms | Infomax ICA | ICA | Fast ICA | CICA |
| Speed in MHz | 0.111 | 10.2 | 12.29 | 56 |
| Power Consumption | 98.56mW | NA | 24.992mW | 162µW |
| Resource Utilization | 35 | 42 | 28 | 19 |

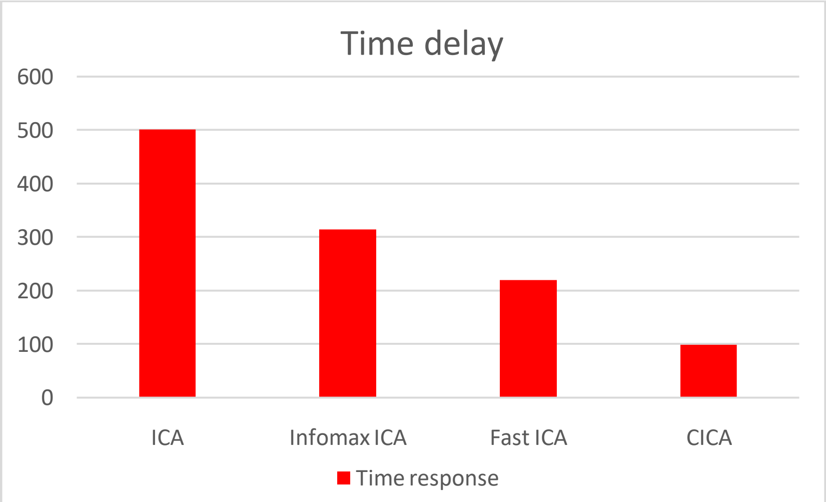


**Figure 7: Comparison of Time Response**


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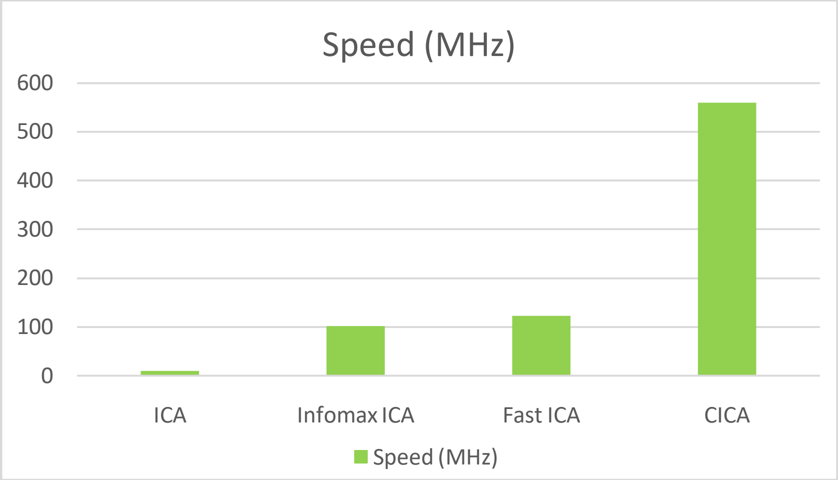


**Figure8: Comparison of Speed in MHz**

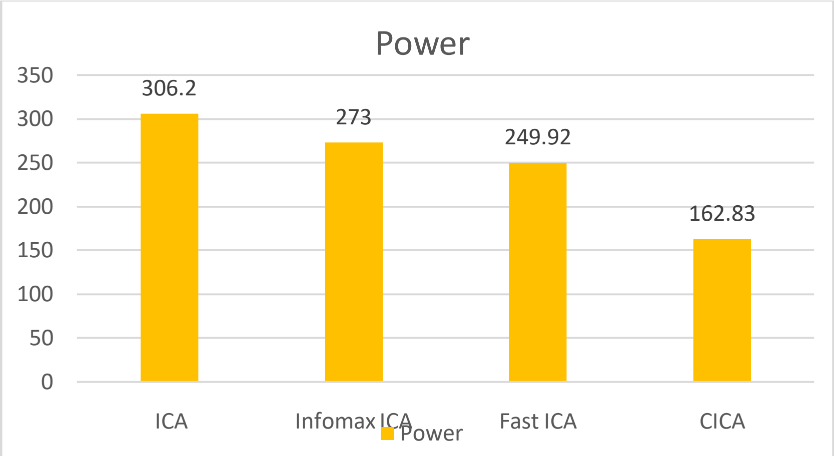


**Figure 9: Power Consumption**

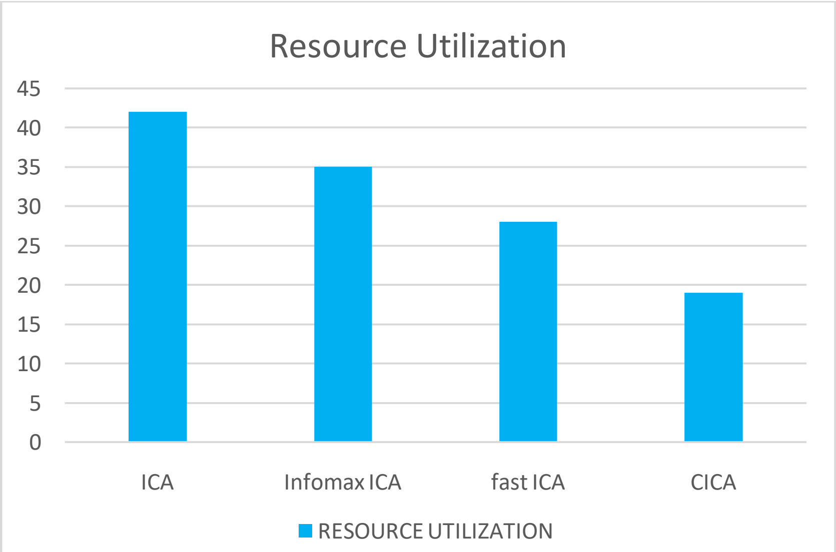


**Figure10: Resource Utilization in MHz**

When compared with existing algorithms the proposed Cuckoo algorithm delivers an efficient performance by means of reducing the time delay, power consumption, and area utilization as shown in above diagram.


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## 6. CONCLUSION

In this work, an efficient implementation of FPGA architecture has been presented with a Cuckoo Algorithm and the result was evaluated. In addition, it gains good area and power consumption for further reduction of reverberation on a nosy channel. In previous methods, training sequences were transmitted to know the characteristics of channel. To reduce the redundant bit and hardware cost the new FPGA was proposed to improve the reliability. Hence the reliability and SNR values are increased. Various methods are compared with proposed Cuckoo search algorithm to demonstrate the efficiency of frequency, time delay and power consumption with reduced area utilization.The proposed method avoids the drawback of separated sounds with improper localization, directivity and spatial quality of separate sources. The wavelet filter and multi-step linear prediction coding (mLPC) for extraction of coefficients in the late reverberation. The reverberated signals are eliminated using backward differentiation. Cuckoo is an optimization technique is used to improve the effectiveness of ICA. To reduce the redundant bit and hardware cost the new FPGA was proposed to improve the reliability. Hence the reliability and SNR values are increased. Various methods are compared with proposed Cuckoo search algorithm to demonstrate the efficiency of frequency, time delay and power consumption with reduced area utilization.

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