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Performance Analysis of Bacterial Foraging Based Audio Enhancement Technique

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ABSTRACT: Audio enhancement is located to be really critical job in the music indicate processing. Many music development techniques have developed in the present literature. The cuckoo based strategy shows greater performance as in contrast to the last techniques applying SNR. The outcomes received proved the potency of the applied method and its ability to curb sound and improve the speech signal. But due to the constraint of low convergence charge to the real world wide minimum actually at large variety of dimensions of cuckoo search it become very hard to have worldwide optimum solution. So in order to overcome the limits of the cuckoo search, the BFOA algorithm can be used rather than the cuckoo search. As BFO assures the international perfect solution therefore it has provided optimistic enhancement effects on the Cuckoo search technique. Different kind of the music signs are thought to judge the effectiveness of the applied technique. The entire studies are obviously show that the applied technique outperforms on the accessible techniques.

KEYWORDS: Audio Enhancement, Cuckoo Search, Bacterial Foraging Optimization Algorithm, Speech signal techniques.

I. INTRODUCTION

Audio Enhancement (AE) seeks to boost the efficiency of audio connection techniques in loud conditions [1]. AE used in, for instance, to a portable radio connection program, a audio to text program, a speech recognition program, a couple of poor recordings, or to boost the efficiency of products for the experiencing reduced [5, 9]. AE is just a traditional issue in signal processing. Reading support customers often have great problem knowledge audio in a loud background. They an average of require a signal-to-noise ratio SNR of approximately 5–10 dB more than regular experiencing fans to attain exactly the same level of audio knowledge. Thus, many simple and multi-microphone noise decrease methods have already been created for modern experiencing aids. Multi-microphone noise decrease techniques are able to exploit spatial as well as spectral data and are thus preferred to single-microphone techniques [2].

CUCKOO SEARCH AND AUDIO ENHANCEMENT:

A. *Cuckoo Search*: Cuckoo Search Algorithm Cuckoo Search (CS) is one of the newest optimization formulas and was created from the motivation that the obligate brood parasitism of some cuckoo species set their eggs in the nests of other variety birds which is of other species. In Cuckoo Research, three idealized rules are thought which says that each cuckoo sits one egg at any given time, and eliminate its egg in randomly plumped for nest. The 2nd principle states that best nests with good quality of eggs may carry over to the next generations and the third one says that how many accessible variety nests is fixed, and the egg set by way of a cuckoo is discovered by the variety chicken with a likelihood in the range 0 to 1.

B. *Audio Enhancement Techniques:* There are several ways available to classify speech enhancement systems based on the applications, the number of input channels and the processing domain. Depending on the number of input channels, the enhancement techniques can be classified as single or multi-channel schemes. In case of single channel enhancement algorithms, a second channel to furnish the reference noise signal is not available. But the systems are less expensive and less complex. Few examples of single channel enhancement algorithms find views in applications like hearing aids and hands-free communication. In multi-channel techniques, multiple channels are

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available that contain reference noise signal, and noise is reduced by making use of this reference signal. Designing complexity is more in these systems. One of the powerful multichannel speech enhancement techniques is adaptive noise cancellation.

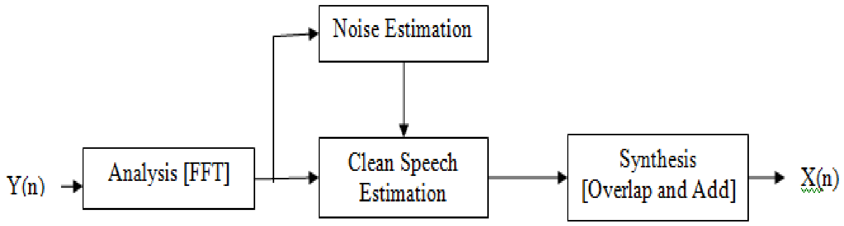

Fig.1. Block Diagram of a Single Channel Speech Enhancement System

C. *Bacterial Foraging Optimization*: Bacterial Foraging Optimization Algorithm (BFOA) is really a new comer to the family of character encouraged optimization algorithms. Program of group foraging technique of a swarm of E.coli microorganisms in multi-optimal function optimization is the important thing strategy of this new algorithm. Germs look for nutrients is a fashion to maximize energy obtained per model time. Specific bacterium also communicates with the others by giving signals. A bacterium takes foraging decisions after contemplating two previous factors. The process, by which a bacterium moves by using little steps while looking for nutrients, is called chemotaxis. The key notion of BFOA is mimicking chemotactic movement of electronic microorganisms in the problem search space. Foraging theory is on the basis of the prediction that animals look for and obtain nutrients in a way that boosts their energy intake Elizabeth per model 65 time T spent foraging. Thus, they try to maximize a function like E/T (or they improve their long-term average charge of energy intake). Maximization of such a function gives nutrient sources to survive and additional time for other crucial activities (e.g., fighting, fleeing, mating, reproducing, asleep, or shelter building).

II. Related Work

Kevin M. Passino et al. (2006) [1] represents steps accompanied by germs to attain worldwide optimum solution: chemotactic step, replica step, elimination and dispersal step. In chemotactic step of BFOA, germs swim in journey of large vitamin floor while they slip when hazardous floor is encountered. Purpose of BFOA is definitely to cut back the price tag on bacteria's action in large vitamin surface. By the conclusion with this, all germs are established in descending obtain of the health value.

Kuldip Paliwal et al. (2010) [2] propose and investigate fusion of modulation spectral subtraction with the MMSE method. The fusion is performed in the short-time spectral domain by combining the magnitude spectra of the above speech enhancement algorithms. Subjective and objective evaluation of the speech enhancement fusion shows consistent speech quality improvements across input SNRs.

Christian D. Sigg et al. (2010) [3] the enhancement of speech degraded by non-stationary interferers is a highly relevant and difficult task of many signal processing applications. We present a monaural speech enhancement method based on sparse coding of noisy speech signals in a composite dictionary, consisting of the concatenation of a speech and interferer dictionary, both being possibly over-complete.

V. Ramakrishnan et al. (2011) [4] presented a two-stage strategy to correct the presentation progress issue in true noisy world. This method comprises of typical spectral subtraction strategy accompanied by a series of perceptually decided post processing algorithms.

Phiplipos C. Loizou et al. (2011) [5] shows theoretical framework that can be used to analyze potential factors that can influence the intelligibility of processed speech. More specifically, this framework focuses on the fine-grain analysis of the distortions introduced by speech enhancement algorithms. It is hypothesized that if these distortions are properly controlled, then large gains in intelligibility can be achieved.

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Ji Ming et al. (2011) [6] introduces method to maximally extract these two features of speech for speech enhancement. We demonstrate that this can reduce the requirement for prior information about the noise, which can be difficult to estimate for fast-varying noise.

Binitha et al. (2012) [7] investigated the significance of varied bio-inspired techniques on basis of the get a grip on variables and application. But this report didn't build the precise dimension selection which numerous nature inspired techniques are evaluated. Thus, each one of these papers give constrained knowledge to the researchers and practitioners in regards to the difficulties and prospects of nature inspired algorithms.

A. Narayanan et al. (2012) [8] presented a SNR opinion method which will be established on computational verbal scene examination (CASA). It is just a binary masking scheme. This method cannot be employed for short-time SNR estimation. This method requires autocorrelation computation and package extraction at each T-F unit. Results of various studies show that the in the offing strategy increases results than different long-term SNR opinion algorithms.

Nima Yousefian et al. (2012) [9] proposed A novel dual-microphone speech enhancement technique. The technique utilizes the coherence between the target and noise signals as a criterion for noise reduction and can be generally applied to arrays with closely spaced microphones, where noise captured by the sensors is highly correlated.

N. Yousefian et al. (2013) [10] planned a coherence-based combined mike method for estimation of SNR. That method can be used for hearing aids and cochlear implant devices. Various experiments have now been done in different conditions. The outcome show that the planned method gives substantial efficiency in anechoic and moderately reverberant conditions.

N. Madhu et. al (2013) [11] experimented with establish a so named binary disguise as the objective of binary disguise estimation. Here, it's shown that practices using binary goggles can improve the intelligibility at low SNR values. For appropriate effects, a low spectral quality, patterned utilizing the Bark-spectrum range is to be used. The efficiency of IBM and IWF has compared. Intelligibility check shows the higher intelligibility values of IWF than IBM.

Nasser Mohammedhia et al. (2013) [12] Reducing the interference noise in a monaural noisy audio signal has been a challenging task for many years. Compared to traditional unsupervised audio enhancement methods, e.g., Wiener filtering, supervised approaches, such as algorithms based on hidden Markov models (HMM), lead to higher-quality enhanced audio signals.

Xugang Lu et al. (2013) [13] We previously have applied deep auto encoder (DAE) for noise reduction and audio enhancement. However, the DAE was trained using only clean audio. In this study, we further introduce an explicit denoising process in learning the DAE. In training the DAE, we still adopt greedy layer-wised pre training plus fine tuning strategy.

J. B. Crespo et al. (2014) [14] shown a method for audio support in a case where there are lots of play right back regions. In such a event, signals from region head to other leading to deterioration of audio intelligibility. An easy distortion is employed to enhance the standard or intelligibility. Results display the features of multizone control on the iterated request of simple region algorithm.

J. Jensen et al. (2014) [15] proposed a method based on common information for estimation of normal intelligibility of loud and refined presentation signal. This technique estimates the common information by evaluating the critical-band amplitude covers of loud or refined presentation signal since mmse can be viewed as an indicator for the intelligibility of loud audio. Simulation results reveal that the proposed method can anticipate the intelligibility of presentation altered by equally stationary and non-stationary noises.

D. P. K. Lun et al. (2014) [16] shown a greater presentation advancement algorithm based on a novel expectation maximization (EM) framework. The traditional TCS method is used to begin the algorithm. The strategy employs the sparsity of audios in the cepstral domain. The strategy performs effectively once the presentation is altered by the non-stationary noises. Experimental results reveal that the proposed method outperforms other methods for verbal audio. However for unvoiced situation different algorithm alongside proposed algorithm is needed.

Seon M. Kim et al. (2014) [17] shown a method for target presentation estimation by thinking about the spatial cues in loud environments. In this process, SNR is estimated utilizing the period big difference obtained from dual-microphone signals. As direction-of-arrival (DOA) of target signal is related to the period big difference between multiple mike signs, so DOA-based SNR is estimated in this method. The performance of this method is examined in terms of SDR, SIR and SAR. Results reveal that the Wiener filter using the proposed DOA based SNR estimation performs much better than other presentation advancement methods.

Yong Xu et al. (2014) [18] this presents a regression-based audio enhancement framework using deep neural networks (DNNs) with a multiple-layer deep architecture. In the DNN learning process, a large training set ensures a powerful modeling capability to estimate the complicated nonlinear mapping from observed noisy audio to desired

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clean signals. Acoustic context was found to improve the continuity of audio to be separated from the background noises successfully without the annoying musical artifact commonly observed in conventional audio enhancement algorithms.

Martin Krawczyk et al. (2014) [19] the enhancement of audio which is corrupted by noise is commonly performed in the short-time discrete Fourier transform domain. In case only a single microphone signal is available, typically only the spectral amplitude is modified. However, it has recently been shown that an improved spectral phase can as well be utilized for audio enhancement, e.g., for phase-sensitive amplitude estimation.

Anil garg et al. (2015) [20] a very successful noise reduction strategy for improvement of presentation signals using enhanced disguise is proposed. Formerly, the loud presentation show is damaged in to numerous time–volume (TF) items and the functions are created by acquiring the Amplitude Magnitude Spectrogram (AMS). The signals are then marked centered on quality percentage in to numerous lessons to help make the unique number of solutions. Therefore, the perfect disguise for each and every single form is done centered on Cuckoo search algorithm.

III. PROPOSED ALGORITHM

A. *Research Methodology:*

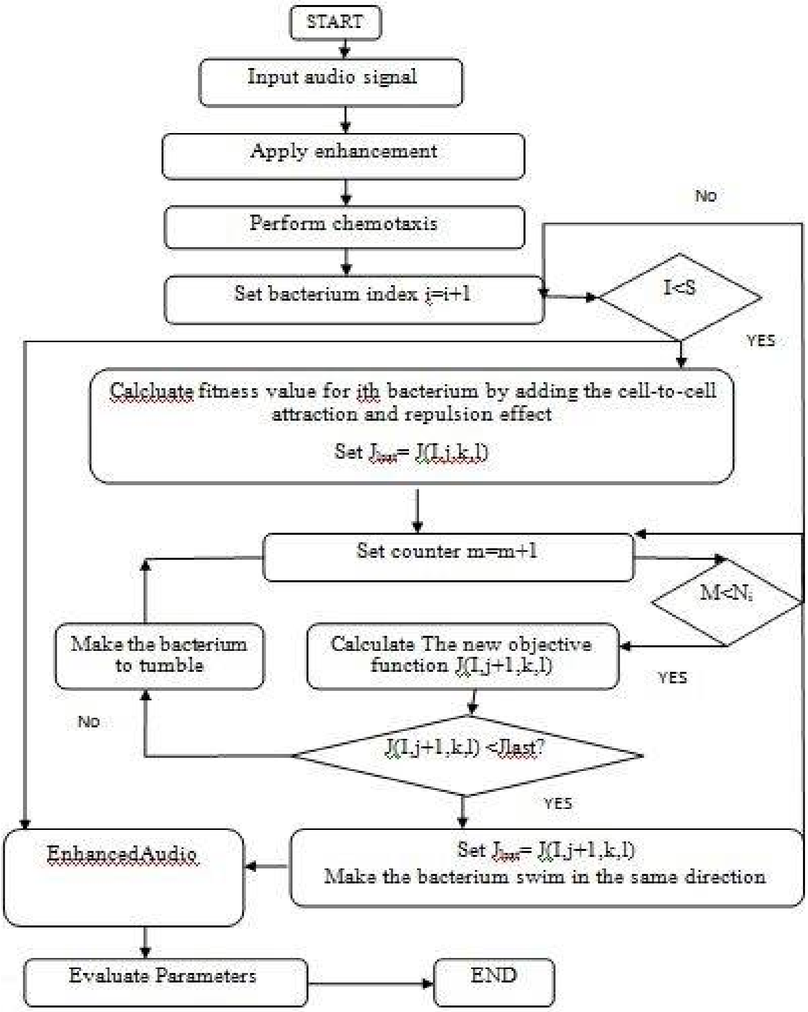

Fig. 2. Methodology of the work

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- Initialize parameters.
- Elimination-dispersal loop.
- Reproduction loop.
- Chemotaxis loop.
- Reproduction.
- Elimination- dispersal.
- End.

B. *Description of the Proposed Algorithm:*

Bacteria Foraging Optimization Algorithm, proposed by Passino, is just a new comer to the household of nature-inspired optimization algorithms. For during the lastfive decades, optimization algorithms like Genetic Algorithms, Evolutionary Programming, Evolutionary Strategies, which draw their inspiration from evolution and natural genetics, have now been dominating the realm of optimization algorithms. Recently natural swarm inspired algorithms like Particle Swarm Optimization, Ant Colony Optimization have discovered their way into this domain and proved their effectiveness. Application of group foraging strategy of a swarm of E.coli bacteria in multi-optimal function optimization is the important thing concept of the newest algorithm. Bacteria look for nutrients in a way to increase energy obtained per unit time. Individual bacterium also communicates with others by sending signals. A bacterium takes foraging decisions after considering two previous factors. The procedure, by which a bacterium moves by taking small steps while trying to find nutrients, is known as chemotaxis and key concept of BFOA is mimicking chemotactic movement of virtual bacteria in the issue search space.

IV. PSEUDO CODE

Step 1: Initialize parameters p, S, N_C, N_S ,N_{re}, N_{ed}, P_{ed}, C(i)(i=1,2…S), θ^i Algorithm.
Step 2: Elimination-dispersal loop: l=l+1
Step 3: Reproduction loop: k=k+1
Step 4:Chemotaxis loop: j=j+1

[a] For i =1,2…S take a chemotactic step for bacterium i as follows.
[b] Compute fitness function, J (i, j, k, l).
Let, J (i, j, k, l) J (i, j, k, l) +J_{CC} (θ^i(j,k,l),P(j,k,l)) (i.e. add on the cell-to cell attractant–repellant profile to simulate the swarming behavior) where, J_{cc} is defined in (2).
[c] Let J_{last}=J (i, j, k, l) to save this value since we may find a better cost via a run.
[d] Tumble: generate a random vector p Δ (i) ∈$\Re^p$ with each element Δm(i),m =1,2 ,..., p, a random number on [- 1, 1].
[e] Move:

$$\text{Let } (\theta^i(j+1,k,l) = (\theta^i(j,k,l) + \frac{C(i)\Delta(i)}{\sqrt{\Delta^T(i)\Delta(i)}} \qquad \text{eq.(1)}$$

This results in a step of size C(i) in the direction of the tumble for bacterium i.
[f] Compute $J(i, j+1, k, l)$ and let

$$J(i,j+1,k,l) = J(i,j,k,l) + J_{cc}(\theta^i(j+1,k,l), P(j+1,k,l) \qquad \text{eq.(2)}$$

[g] Swim i) Let m=0 (counter for swim length).
ii) While m< N_s (if have not climbed down too long).
Let m=m+1.
If $J(i,j+1,k,l)$ =<J_{last}(if doing better), let $J_{last} = J(i,j+1,k,l)$and let

$$\text{Let } (\theta^i(j+1,k,l) = (\theta^i(j,k,l) + \frac{C(i)\Delta(i)}{\sqrt{\Delta^T(i)\Delta(i)}} \qquad \text{eq.(3)}$$

And use this $(\theta^i(j+1,k,l)$ to compute the new $J(i,j+1,k,l)$as we did in [f]
Else, let m= N_S . This is the end of the while statement.
[h] Go to next bacterium (i+1) if i ≠ S (i.e., go to [b] to process the next bacterium)

Step 5: If j <N_C, go to step 4. In this case continue chemotaxis since the life of the bacteria is not over.
Step 6: Reproduction:
[a] For the given k and l, and for each i = 2,1 ,..., S , let

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$$J^{i}_{health}=\sum_{j=1}^{N_c+1} J(i,j,k,l)$$

be the health of the bacterium i (a measure of how many nutrients it got over its lifetime and how successful it was at avoiding noxious substances). Sort bacteria and chemotactic parameters C(i) in order of ascending cost J_{health} (higher cost means lower health).

[b] The S_r bacteria with the highest J_{health}values die and the remaining S_r bacteria with the best values split (this process is performed by the copies that are made are placed at the same location as their parent).

Step 7: If k <N_{re}, go to step 3. In this case, we have not reached the number of specified reproduction steps, so we start the next generation of the chemotactic loop.

Step 8: Elimination-dispersal: For i = 1,2, ..., S with probability P_{ed} , eliminate and disperse each bacterium (this Keeps the number of bacteria in the population constant). To do this, if a bacterium is eliminated, simply Disperse another one to a random location on the optimization domain. If l < N_{ed} , then go to step 2; otherwise end.

V. SIMULATION RESULTS

This paper has designed and implemented the proposed technique in MATLAB tool u2013a. This section offers the cross validation between existing and proposed techniques. Some well-known audio performance parameters are mean error and mean square error i.e. for audios have already been selected to prove that the performance of the proposed algorithm is very much better than the existing methods.

1) *Mean Error:* It shows the quantized analysis of the mean error. Table 1 has shown the mean error have to be reduced which means applied algorithm is showing the better results compared to the available methods as mean error is minimum in all of the cases.

TABLE I. Mean Error

Input speech signal	**Existing Value**	**Experimental Values**
1	6.2956	2.1209
2	5.3679	1.3343
3	3.4222	0.8734
4	9.1299	0.9328
5	10.3741	1.1262
6	1.9577	1.0795
7	10.2059	4.0284
8	9.1122	2.8265
9	11.8861	5.2599
10	12.2079	2.3793

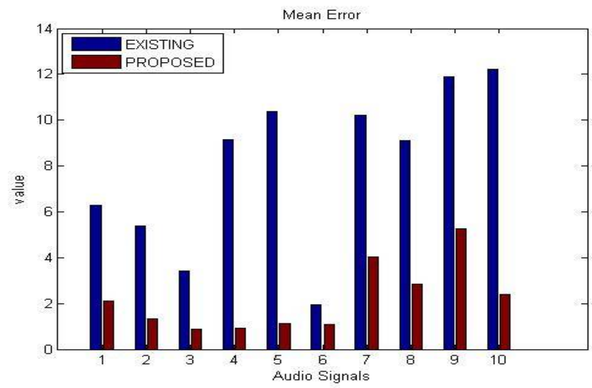

Fig. 3. Mean Error

The above fig.3 indicates the quantized analysis of the mean error of various audios using BFOA by existing technique (Blue color) and BFOA by applied approach (Red Color). . This decrease represents improvement in the objective quality of the audio signal.

2) *Mean Square Error (MSE):* It represents the quantized analysis of the mean square error. Table 2 shows the mean square error have to be reduced which means applied algorithm is showing the better results compared to the available methods as mean square error is less in all of the cases.

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TABLE II. Mean Square Error

Input speech signal	Existing Values	Experimental Values
1	64.8305	27.1573
2	48.3800	10.7372
3	22.6485	4.6006
4	148.7194	6.4796
5	190.9397	9.4492
6	12.7631	9.5144
7	186.4438	137.7999
8	148.8238	59.4716
9	248.6994	223.4763
10	228.0471	42.1171

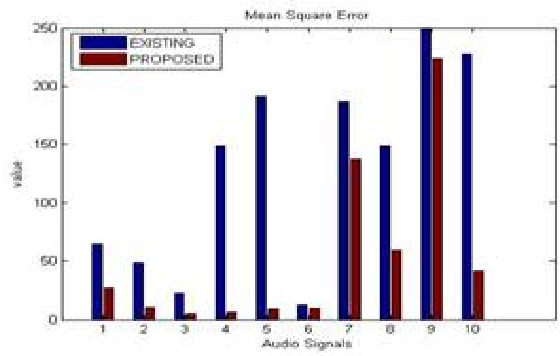

Fig. 4. Mean Square Error

The above fig.4 indicates the quantized analysis of the mean square error of various audios using BFOA by Existing Technique (Blue color) and BFOA by applied approach (Red Color). It has clearly shown from the plot that there is decrease in MSE value of audios with the usage of applied method over other methods in all audios.

3) *Peak Signal to Noise Ratio (PSNR):* As PSNR have to be maximized; so the main goal is to increase the PSNR as much as possible. Table III has clearly shown that the PSNR is maximum in the case of the applied algorithm therefore applied algorithm provides better results compared to the available methods.

TABLE III. Peak Signal to Noise Ratio

Input speech signal	Existing Values	Experimental Values
1	30.0130	33.7919
2	31.2481	37.8219
3	34.5804	41.5027
4	26.4071	40.0153
5	25.3218	38.3769
6	37.0712	38.3470
7	25.4253	26.7383
8	26.4041	30.3877
9	24.1741	24.6385
10	24.5506	31.8862

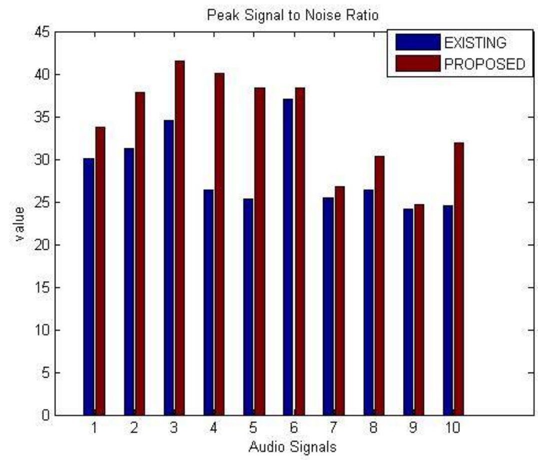

Fig. 5. Peak Signal to Noise Ratio

The above fig5 has shown the quantized analysis of the peak signal to noise ratio of various audios using BFOA by Existing Technique (Blue Color) and BFOA by applied Approach (Red Color). It has clearly shown from the table that there is increase in PSNR value of audios with the usage of applied method over other methods. This increase represents improvement in the objective quality of the audio signal.

4) *Root Mean Square Error (RMSE):* As root mean square error have to be reduced which means applied algorithm is showing the better results compared to the available methods as root mean square error is less in all cases. Table IV has clearly shown that the RMSE is minimum in the case of the applied algorithm therefore applied algorithm provides better results compared to the available methods.

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TABLE IV. Root Mean Square Error

Input speech signal	Existing Values	Experimental Values
1	8.0517	5.2113
2	6.9556	3.2768
3	4.7590	2.1449
4	12.1951	2.5455
5	13.8181	3.0740
6	3.5726	3.0845
7	13.6544	11.7388
8	12.1993	7.7118
9	15.7702	14.9491
10	15.1012	13.2616

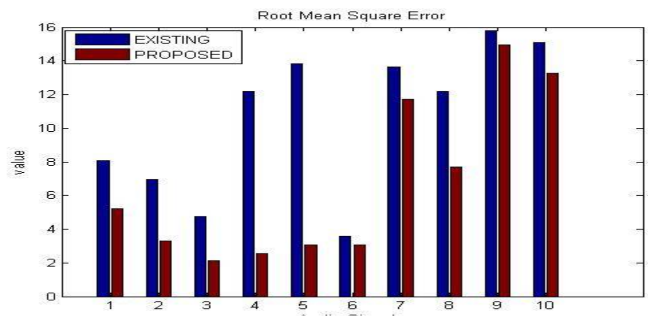

Fig. 6. Root Mean Square Error

Fig 6 shows the quantized analysis of root mean square error of various audios using BFOA by Existing Technique (Blue color) and BFOA by applied Approach (Red Color). It has clearly shown from the plot that there's decrease in RMSE values of audios with the usage of applied method around other methods in all the audios. This decrease represents improvement in the objective quality of the audio.

5) Bit Error Rate (BER): As bit error rate have to be reduced therefore the applied algorithm is showing the better results compared to the available methods as bit error rate is minimum in all the cases. Table V has clearly shown that the BER is minimum in the case of the proposed algorithm therefore proposed algorithm provides better results compared to the available methods.

TABLE V. Bit Error Rate

Input speech Signal	Existing Values	Experimental Values
1	0.0333	0.0296
2	0.0320	0.0264
3	0.0289	0.0241
4	0.0379	0.0250
5	0.0395	0.0261
6	0.0270	0.0267
7	0.0393	0.0374
8	0.0379	0.0329
9	0.0414	0.6406
10	0.0407	0.0314

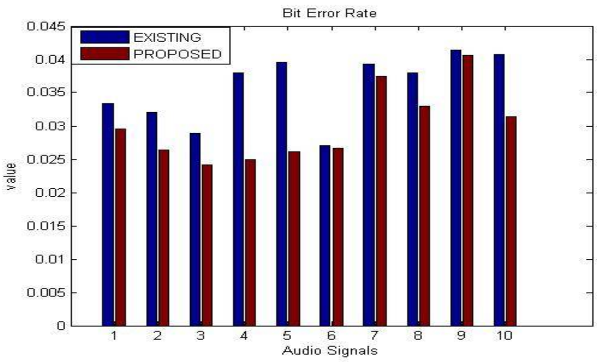

Fig. 7. Bit Error Rate

Fig 7 shows the quantized evaluation of the bit error rate of various audios using BFOA by Existing Technique (Blue color) and BFOA by applied Approach (Red Color). It has clearly shown from the table that there is decrease in BER value of audios with the usage of applied technique around other methods in all the audios. This decrease represents improvement in the objective quality of the audio signal.

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6) Structural Similarity Index Metric (SSIM): As SSIM have to be maximized; so the main aim is to increase the SSIM. Table VI has clearly shown that the SSIM is maximum in case of the applied algorithm thus applied algorithm provides better results compared to the available methods.

TABLE VI. Structural Similarity Index Metric

Input speech signal	Existing Values	Experimental Values
1	0.9667	0.9704
2	0.9680	0.9736
3	0.9711	0.9759
4	0.9621	0.9750
5	0.9605	0.9739
6	0.9730	0.9739
7	0.9607	0.9626
8	0.9621	0.9671
9	0.9586	0.9594
10	0.9593	0.9686

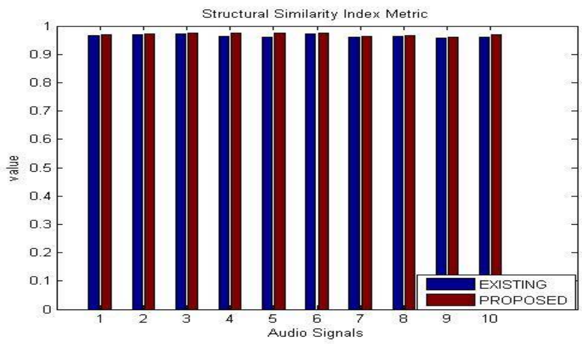

Fig. 8. Structural Similarity Index Metric

Fig 8 indicates the quantized analysis of the structural similarity index metric of different audios using BFOA by existing technique (Blue Color) and BFOA by applied approach (Red Color).It has clearly shown from the table that there's increase in SSIM values of audios with the usage of applied method over other methods. This increase represents the improvement in the objective quality of the audio signal.

VI. Conclusion and Future Work

Audio Enhancement is just a traditional issue in signal processing. Reading support customers often have great problem knowledge audio in a loud background. This paper has proposed the BFOA algorithm is used instead of the cuckoo search. The proposed technique is designed and implemented in the MATLAB 2010a.Different parameters have been considered for experimental purpose. As BFO guarantees the global optimum solution therefore it has provided optimistic enhancement results over the Cuckoo search technique. Different kind of the audio signals are considered to evaluate the effectiveness of the proposed technique. The overall experiments are clearly indicated that the proposed technique outperforms over the available techniques.

This work has focused on BFO based enhancement only, in near future BFO will be hybridized with some other evolutionary optimization techniques. Also technique may also be used for some other signals.

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