

Bacterial foraging optimization of speech coder for software defined radio application

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Abstract. The paper presents integration of Discrete Wavelet Cosine Transform technique and Bacterial Foraging Algorithm (BFO) for the development and optimization of speech coder. It is depicted how by filtering the limited number of high energy components of transformed coefficients with parallel programming can maintain the speech signal quality in coding over wide range of bit rates. The performance of existing and proposed speech coding technique attributes such as compression ratio, coding delay, computational complexity and quality of reconstructed speech is examined for multiple bit rates and compared with other existing speech coding techniques in Matlab environment. The result shows improvement in performance with respect to all attributes at the cost of increase in complexity.

Keywords: Speech coder attributes, discrete wavelet cosine transform, bacterial foraging optimization, software defined radio

1. Introduction

There has been a major deal of interest in speech coding techniques based on coding attributes such as bit rate, complexity, coding delay and speech quality and application [1]. Speech & audio coding has been distinguished as per the signal bandwidth, lossless or lossy coding techniques by using different coding algorithms. The speech & audio signals are classified as narrowband, wideband, super wideband and full band signals to specify the bandwidth of signal. According to the classification of signal bandwidth, appropriate sampling frequency is selected for its processing. The processing of Narrowband signal (200 Hz to 3400 Hz) is done at 8 KHz sampling frequency. Whereas, wideband (50 Hz –7 KHz) audio signal needs higher range of sampling frequency (16 KHz to 48 KHz) for coding.

The upward movement from narrowband to fullband signal has made significant changes in signal processing and quality expectations [2, 3]. The goal of speech/audio coding is to maintain the quality, intelligibility in minimum possible number of bits and algorithmic delay as per the application need such as digital cellular communication, video conferencing, Voice over Internet Protocol (VoIP) and audio/video streaming. The existing speech coding techniques are basically classified in two categories as lossless and lossy coding techniques. The lossless coding techniques such as Pulse Code Modulation (PCM) and Adaptive Pulse Code Modulation (ADPCM), process the speech signal by removing the redundant information. The lossless coding techniques are in use since long for Public Switched Telephone Network (PSTN). The lossy coding offers compression of speech signal by removing insignificant information content, hence the reconstructed signal differs from the original signal. The extent of loss of information in lossy coding depends on the application need without losing the signal intelligibility and quality of

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Table 1
Existing Speech Coding Techniques – Summary

Algorithm	Application	NB/WB/ FB	Bit Rate(Kbps)	Frame Size(ms)	Codec Delay (ms)
PCM	Telecommunication	NB	6,4	0.125	0.25
ADPCM	Telecommunication	NB	16,32,24,40	0.125	0.25
CS-ACELP	Cellular	NB	11.8	10	25
RPE-LTP	Cellular	NB	12.2, 13	20	20
Transform Coder	Audio and Video	FB/	32–128	20	20
		WB/	24,32	20	20
		SWB	24,32,48	20	20

reconstructed signal. Irrespective of loss of information, the lossy coding techniques has received wide application because of less memory requirement and bit rate to fit the signal for transmission in smaller bandwidth. The has been a huge development in lossy coding techniques which includes Linear Predictive Coding (LPC), Sub-band Coding (SBC), Conjugate Structure Algebraic Codebook Excited Linear Prediction (CS-ACELP), Regular Pulse Excitation Long Term Prediction (RPE-LTP), Transform coding using Fourier Transform (FT), Discrete Cosine Transform (DCT) and Wavelet Transform (WT) [5, 6]. These coding techniques are used in mobile communication, VoIP and Satellite communication etc. There is no regulation set by governing agencies on the use of any speech coding technique in the specific frequency spectrum. The progress in communication field for different applications have made different trade-offs in bit rate, delay, complexity and quality of service as per the requirement is mentioned in Table 1 [7].

After processing the speech signal through any of the above methods followed by further channel formalities, the coded signal is ready for wired or wireless communication. The fundamental medium for wireless communication is natural radio frequency spectrum which is limited and cannot be generated. The resource utilization, distribution and use of radio spectrum in each country is nationally and internationally governed by the corresponding government agencies such as Federal Communications Commission (FCC) and the International Telecommunication Union (ITU) respectively. These governing agencies set the rules and regulations for frequency allocation to different services. The spectrum bands are licensed to certain services such as mobile Communication, satellite Communication, Television and FM broadcast to protect the signal from harmful interference in an allocated radio band [8]. Whereas, survey report reveals that the fixed spectrum allocation is leading towards partial utilization of the radio spectrum only in a particular time

slot otherwise the the spectrum remains unused [9, 10]. A unifying thread to cover narrow band, wide-band, super wideband & full band applications in the wide range of bit rates and comparable speech coder attributes of existing techniques with parallel processing program is essential for the performance enhancement.

This paper presents transform based coding technique which fulfills the requirements of existing speech coders used in different applications to operate at multiple bit rates, frame size, low algorithmic delay and quality of recovered signal. In last decade, Discrete Wavelet Transform has raised as a powerful tool to process and analyze the non-stationary signal such as speech due to its time varying nature. The localization feature of wavelet along with its time-frequency property makes it suitable for analyzing speech signal which is characterized by abrupt changes, drifts and trend. We are proposing the transform based speech codec using Discrete Wavelet Cosine Transform (DWCT) for processing of the speech signal to achieve speech coding at multiple bit rates. The proposed coding technique is designed to facilitate the solution of congestion in mobile spectrum by spreading over the unused spectrum or white spaces through upcoming technology called Software Defined Radio (SDR) [11, 12]. Software Defined Radio defines system architecture for the optimization of wireless resource procedures and spectrum access to enhance efficiency, measurements and effectiveness in wireless networks through software. The SDR devices operate on various air interfaces for effective use of spectrum bands by means of next generation radio compatibility. It can provide instantaneous access to multiple radio in licensed and unlicensed bands in coordination with Cognitive Radio (CR). The proposed coding technique suits for SDR's cognitive radio application at multiple bit rates and appropriate coding attributes. In view of speech codec attributes such as delay and quality, the multiple bit rate codec is optimized by using Bacterial Foraging Optimization

Algorithm (BFOA). The BFOA is biologically inspired optimization algorithm based on the typical behavior of Escherichia Coli (E. Coli) bacteria in its complete life cycle in search of nutrients has been used for selection of appropriate wavelet and its decomposition level for best coding attributes [13] [14] The coding of speech signal and search of optimized wavelet level of decomposition is done through parallel programming.

The rest part of paper is organized in four more sections. Section 2 describes the Discrete Wavelet Cosine Transform for speech signal coding with BFOA implementation for optimization of proposed coding technique in section 3. Section 4 is discussion on methodology and section 5 presents experimental results obtained in Mathworks' Matlab software environment with evaluation. Finally, we conclude in section 6.

2. Discrete wavelet cosine transform

The wavelet transform consist of pair of time domain and time scale domain. In time scale domain, the given signal is decomposed into several other signals at different levels. This is known as forward wavelet transform. The reverse process of recovering the original signal from decomposed signal is called Inverse Wavelet Transform [6].

Let, $x(t)$ be the speech signal which is also called as the signal in time domain. To decompose this signal, the wavelet transform of speech signal $x(t)$ is given by,

$$\Psi x(m, n) = 2^{-\frac{m}{2}} \int_{-\infty}^{\infty} x(t) \Psi * \left(\frac{t - n2^m}{2^m} \right) dt \quad (1)$$

Where, * represent a complex conjugate of $\Psi(t)$ signal. Also, m is the scale and n is time shift parameter. The signal in Equation 1 is dyadic transformation constitutes an orthogonal basis without any information redundancy in decomposed signals [15]. The mother wavelet $\Psi(t)$ is discrete in nature. The discrete time signal is decomposed into its detailed $d(n)$ and approximate $a(n)$ components by using high pass filter $g(n)$ and low pass filter $h(n)$ respectively. The detailed signal $d(n)$ contains high frequency components (transitions, sharp edges) and $a(n)$ contains low frequency components of the original signal. The decomposed component in $d_1(n)$ and $a_1(n)$ present the first level decomposition and determined as fol-

lows.

$$d_1(n) = \sum_k g(k - 2n) \cdot x(k) \quad (2)$$

$$a_1(n) = \sum_k h(k - 2n) \cdot x(k) \quad (3)$$

The length of $a_1(n)$ and $d_1(n)$ is half the length of signal in previous stage. The higher order decomposition can be performed in similar manner [16, 17]. The orthonormality of dilated wavelets supports perfect recovery of the original signal $x(t)$ from $\Psi x(m, n)$ wavelet coefficients. The reconstruction of the signal can be done by Inverse Wavelet Transform as given in Equation 4.

$$x(t) = 2^{-\frac{m}{2}} \sum_m \sum_n \Psi x(m, n) \Psi \left(\frac{t - n2^m}{2^m} \right) \quad (4)$$

Equations 1 and 4 are identical with conjugated time reverse for the decomposition and reconstruction of $x(t)$. The wavelet transform is very effective tool to detect and localize the disturbances. The magnitude of wavelet coefficients associated with disturbances is dominant over the coefficients without disturbances [18]. It helps to compress the signal effectively by retaining the coefficients associated with disturbance and discarding disturbance free coefficients [19]. However signal recovery with quality is possible from the preserved coefficients because most of the discarded coefficients represent noise. The next step towards signal compression is thresholding of wavelet transform coefficients. In this step, the wavelet coefficient below a certain threshold is to be discarded [18]. The threshold value can be decided on the basis of absolute maximum value of the wavelet transform coefficient as given in Equation 5 or Birge-Massart thresholding technique in Equation 6.

$$\eta_s = (1 - \mu) \cdot \max \{ |d_s(n)| \} \quad (5)$$

Where, μ value range is between 0 and 1. If $\mu = 0.8$, then η_s is 20% of maximum absolute value of $d_s(n)$. Therefore, the smaller value $d_s(n)$ components can be discarded.

The Birge-Massart thresholding is given by Equation 6.

$$\text{THR} = W_{\text{bmpen}}(C, L, \sigma, \alpha) \quad (6)$$

Where, σ is the standard deviation of the zero mean Gaussian White Noise and α is tuning parameter for

the penalty term. It's typical value is 2. Let, 't' be the minimizer of,

$$\text{Crit}(t) = -\sum (C(k) + 2A\sigma t(\alpha + \log(n/t))) \quad (7)$$

Where $C(k)$ is the wavelet coefficient & n is the number of wavelet coefficients.

$$\text{THR} = \eta_s = |\text{Crit}(t)| \quad (8)$$

The value as in Equation 5 or 8 is used for thresholding. After thresholding, the wavelet coefficients having value greater than THR value are retained and remaining coefficients are replaced by zero. Hence,

$$d(n) = \begin{cases} d_s(n) & |d_s(n)| \geq \eta_s \\ 0 & |d_s(n)| < \eta_s \end{cases} \quad (9)$$

The wavelet transformed signal after thresholding is further decomposed into cosine packet by applying Cosine Transform as below for the speech sequence of length N .

$$C_i = \alpha_1 \sum_{x=0}^{N-1} d_s(n) \cos \left[\pi \frac{(2x+1)i}{2N} \right] \quad (10)$$

For $i=0,1,2 \dots N-1$

$$\text{Where, } \alpha_1 = \begin{cases} \frac{1}{\sqrt{N}} & \text{for } i = 0 \\ \sqrt{\frac{2}{N}} & \text{for } i \neq 0 \end{cases} \quad (11)$$

And C_i represents DWCT component of speech signal. $F=0$,

$$C_0 = \sqrt{\frac{1}{N}} \sum_{x=0}^{N-1} d_s(n) \quad (12)$$

Where, C_0 is the first transform coefficient. It is the average value of all sample and referred as average coefficient.

3. The bacterial foraging optimization algorithm in proposed codec

Nature ecosystem is one of the rich sources of mechanism to obtain the optimum solution of the problems in any computational system. Similarly Bacterial Foraging Optimization Algorithm is inspired by an activity "chemotaxis" exhibited by foraging behavior of *E. Coli* bacteria [12]. The BFO

system follows three important mechanisms named chemotaxis, reproduction and elimination-dispersal in its life cycle. The detail description of BFO mechanism is formulated as follows.

3.1. Chemotaxis

Chemotaxis is the first step in bacterial foraging life cycle. It is the movement of bacteria in response to any kind of stimulant. The movement can be in the form of swimming or running. In BFO, "Run" indicates a unique walk in the same direction with certain step size as previous one and swimming indicates movement with desired speed in circular direction along set path. Under certain complex situation, the BFO algorithm may take very long time to reach to the optimum result. To avoid such situation, modification can be made in number of search groups and steps. The run length unit parameter $C(i)$ for chemotactic step size during the run and $\theta^i(j, k, l)$ describes the bacterium location status at i^{th} step [20]. After each computational chemotactic step, the moment of i^{th} bacterium is obtained by,

$$\theta(j+i+k+l) = \theta(j, k, l) + C(i) \frac{\Delta(i)}{\sqrt{\Delta^T(i) \cdot \Delta(i)}} \quad (13)$$

Where, $\Delta(i)$ is the direction vector of chemotactic step. The value of $\Delta(i)$ is same as previous chemotactic step for "run" bacterial movement [21]. The additional function $J(i, j, k, l)$ known as step fitness is also evaluated in run activity where, i, j, k, l represent current step value, chemotaxis loop number, group number and level in k^{th} group respectively. The fitness value is evaluated at every step to decide the continuation of search process.

3.2. Reproduction

After ' N_c ' number of chemotaxis steps, the reproduction step is performed. Then fitness value of the bacteria is assessed to retain healthiest bacteria. The health of each bacterium is calculated by summing the step fitness in its life span as in Equation 14.

$$\sum_{J=1}^{N_c} J(i, j, k, l) \quad (14)$$

Where, N_c is the total number of steps in chemotaxis process [23, 24].

3.3. Elimination-dispersal

Each bacterium health is noted with energy level according to its search capability in limited space. High energy level shows better performance of the bacterium. The level of bacterium energy decides the probability of elimination and reproduction [21, 22]. As the bacterium may get stuck in its initial search position or local optima then it is necessary for BFO to change gradually or eliminate the accidents of being trapped in local optima. Then some bacteria are chosen to be killed and the process is moved to another position within the space.

The design of speech coding algorithm by using bacterium behavior in above mentioned steps can lead towards successful optimal solution for the proposed speech coding technique.

4. Methodology

In proposed work, the speech coder represents the bacteria moving towards the best values of coder attributes from one wavelet family to another for several decomposition level. The speech coding using Discrete Wavelet Cosine Transform and optimization of codec performance parameters by Bacterial Foraging Optimization technique is given below.

Step 1: Initially, the speech signal is sampled at any of the standard sampling frequency from 8 KHz to 48 KHz and ordered into frames of fix duration.

Step 2: Initialize BFO parameter N , S , N_c , N_s , N_{re} , $C(i)$ (for $i = 1, 2, \dots, 5$), θ^i .

Where,

N : Search space Dimension for different Daubechies Wavelet.

S : Number of performance parameters '4'.

N_c : Number of decomposition level '5'.

N_s : Swim step for wavelet switching.

N_{re} : Number of reproductive steps '5'.

$C(i)$: The run length unit '1'.

Step 3: Elimination and Dispersal: $l = l + 1$ to switch towards next wavelet.

Step 4: Reproduction loop: $k = k + 1$ to proceed with next level of decomposition.

Step 5: Apply DWT to the frames of speech signal.

Step 6: Apply Birge-Massart thresholding to wavelet coefficients.

Step 7: Apply cosine transform to select and retain the first high energy-low frequency coefficients as per the desired bit rate requirement. The number of coefficients to be retained is given by,

$$N_{Coeff} = \frac{Fz * D_{BR}}{\text{number of bits per coefficient}} \quad (15)$$

Where, Fz is speech frame size and D_{BR} is desired bit rate of speech coder.

Step 8: Chemotaxis loop: $j = j + 1$.

8.1: For $i = 1$, take the chemotactic step for bacteria.

8.2: Compute fitness function $j(i, j, k, l)$.

8.3: Let $J_{last} = J(i, j, k, l)$ to search better value in the next run.

8.4: Move in the specified direction of run given in eqn. 13.

8.5: Calculate $J(i, j + 1, k, l)$.

8.6: If $j(i, j + 1, k, l) > J_{last}$ then, $J_{last} = j(i, j + 1, k, l)$.

The step of size $C(i)$ in same direction to be taken & θ to be used to find $j(i, j + 1, k, l)$.

8.7: If $j < N_c$, return to step 4 and continue for the chemotaxis.

Step 9: For the given value of k, l and for each i , Equation 16 represents the health of bacterium i .

$$J_{health}^i = \sum_{i=1}^{N_c+1} j(i, j, k, l) \quad (16)$$

Step 10: If $k < N_{re}$, switch to step 3 as we have not reached the number of specified reproduction steps.

Step 11: The value obtained in step 9 is the optimized value.

In general, increase in s and N_c can increase the computational complexity of the algorithm [21]. If the search space is too large and optimum value lies in deep then it may turn into never ending process for optimum result. To avoid this situation, the search space is limited to the number of decomposition levels and wavelets.

Table 2
Attributes of Proposed Codec before and after optimization

Codec	NB/WB/FB	Bit Rate (Kbps)	Frame Size (ms)	Compression Ratio	Coding Delay (ms)	Complexity
Proposed Technique Before Optimization	NB	12.2	0.125	10.49 to 1.0	0.042	10
	WB	To	To	20.98 to 2.0	To	
	FB	128	60	62 to 6.0	19.8	
Proposed Technique After Optimization	NB	12.2	0.125	10.49 to 1.0	0.02	16
	WB	To	To	20.98 to 2.0	To	
	FB	128	60	62 to 6.0	6.02	

Table 3
Attributes of Existing Speech Coding Technique in Matlab

Codec	NB/WB/FB	Bit Rate(Kbps)	Frame Size (ms)	Compression Ratio	Coding Delay (ms)	Complexity
PCM	NB	64	0.125	1	0.34	3
ADPCM	NB	16,20,24,36,40,64	0.125	1	0.38	5
ACELP	NB	12.2	20	2.8	5	27
RPE-LTP	NB	13	20	5	32	23
DFT	NB	64	20	1.99	6.2	15
DCT	NB/ WB/ FB	64	20	2.04	1.1	12
DWT	NB/ WB/ FB	64	20	2.89	9.1	10
Psychoacoustic	FB	352	20	9.05	12.2	31
DWT-DCT	NB/ WB/ FB	13–128	1–20	9.84 – 1 19.69 – 2 39.38 – 4	0.5 – 103	10

5. Result & discussion

The proposed coder designed and evaluated for multiple bit rate, Compression Ratio, algorithmic delay and cyclomatic complexity is given in Table 2 for the Codec before and after Optimization. Similarly, implementation of other existing techniques in same environment is given in Table 3. The simulation has been carried out in Matlab 2013a software on PC with Intel core i3, 2.53 GHz, 4GB RAM configuration.

Bit rate: The cognitive radio application of speech coder for mobile communication requires to operate the codec at wide range of bit rate. The proposed technique works successfully for wide range of bit rate from 12.2 Kbps to 128 Kbps. The codec performance for the desired bit rates is sufficient to cover the bit rate requirement of existing coding techniques as given in Table 1.

Compression Ratio (CR): The proposed technique has shown the highest compression ratio 62 at 12.2 Kbps follows declining nature with respect to increase in bit rate as shown in Fig. 1. The num-

ber of coefficients at the codec output is decided as per the bit rate requirement given by Equation 15. The increase in compression ratio in Equation 17 with respect to sampling frequency is observed due to higher energy compaction of Cosine Transform in less number of coefficients.

$$CR = \frac{\text{Original speech signal size}}{\text{Coded signal size}} \quad (17)$$

Coding Delay: Coding delay is the time elapsed from the instant a speech sample appears at the input of the coder to the reconstructed speech sample appears at the decoder output. The frame size selection was done as per the existing codec specification from 0.125 ms to 60 ms. Also, the effect of BFO algorithm has been realized on associated coding delay. The delay value of optimized codec was reduced to almost half of the previous non optimized codec shown in Fig. 2. The proposed technique offers considerably lower delay than the expected value of coding delay (300 ms) for real time communication.

Computational complexity: The computational complexity is determined in the form of Cyclomatic

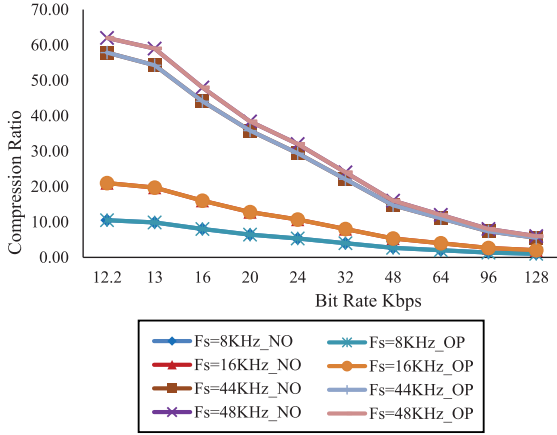


Fig. 1. Compression Ratio at different Sampling Frequencies. (NO - Non Optimized and OP - Optimized).

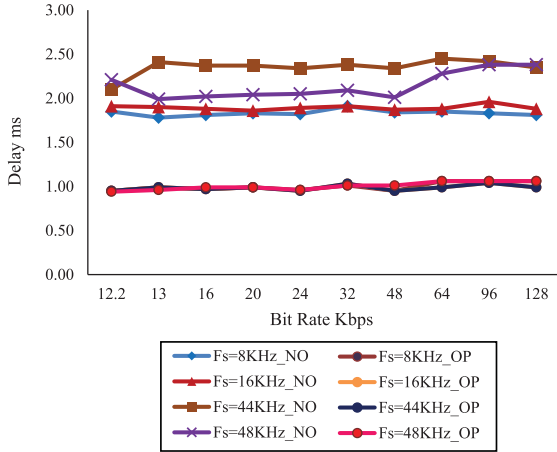


Fig. 2. Proposed codec Coding Delay for frame size 20ms. (NO - Non Optimized and OP - Optimized).

complexity. The cyclomatic complexity is a measure of the structural complexity of the code and expressed in terms of code coverage 'Cm' as given in Equation 18.

$$Cm = 1 + \sum_{i=1}^M (\alpha_d - 1) \quad (18)$$

Where, M is the number of decision points and α_d is the number of outcomes for d^{th} decision points. The complexity value computed for the proposed technique is increased to 16 after the codec optimization which was 10 earlier and shifts the codec complexity in 'Medium' from 'Low' range of complexity.

% Correlation: The correlation coefficient is a way to put a value to the relationship as in Equation 19.

% Correlation

$$= \frac{\sum_{i=1}^N (Q_o - \mu_{Qoi}) (Q_i - \mu_{Qi})}{\sqrt{\sum_{i=1}^N (Q_o - \mu_{Qoi})^2} \sqrt{\sum_{i=1}^N (Q_i - \mu_{Qi})^2}} \times 100 \quad (19)$$

Where, Q_i and Q_o are input and output coefficients. Also, μ_{Qoi} and μ_{Qi} are mean values of input and output coefficients for all 'N' number of samples. Here, the relation between original speech signal and reconstructed speech signal is expressed in terms of % Correlation. In the reverse processing of the coded speech signal, the recovered speech signal is compared with original signal as shown in Fig. 3. This factor is associated with sampling frequency and bit rate. At lower bit rate, large number of vanishing moments of coded signal finds limitation on extent of recovery. Also, the sampling frequency contributes more in % Correlation because of good frequency resolution characteristic of DWT. The signal with lowest correlation of 89% at 12.2 Kbps to 99% at 128 Kbps bit rate for 8 KHz sampling frequency and its improvement with sampling frequency as shown in Fig. 4 confirms the optimization in quality of recovered signal.

Subjective Test: The most popular speech quality measurement method is Absolute Category Rating (ACR). This test is used to evaluate the response given by the subjects for the recovered speech after decoding the coded signal. The coder o/p signal quality has been realized by conducting subjective test as per ITU recommendation P.800 on 3 speech samples in English (1), Hindi (1) and Marathi (1). The judgment of these subjects was defined in terms of rating using "Listening Quality Scale" of Bad (1) to Excellent (5). The evaluation result is expressed in terms of Mean Opinion Score (MOS). The number of subjects invited in the test were 32 (16 Male: 16 Female) in 20 Yrs to 48 Yrs age group to ensure the quality opinion of the average user. The MOS response for proposed technique shown in Fig. 5 received toll quality for the bit rate above 20 Kbps and normal quality otherwise. The improvement in the response for optimized codec can be seen in Fig. 5.

With respect to speech codec attributes, the proposed technique excels the existing speech coding technique standards for wide range of bit rate, compression Ratio, coding delay and recovered speech signal quality in terms of % correlation at the cost

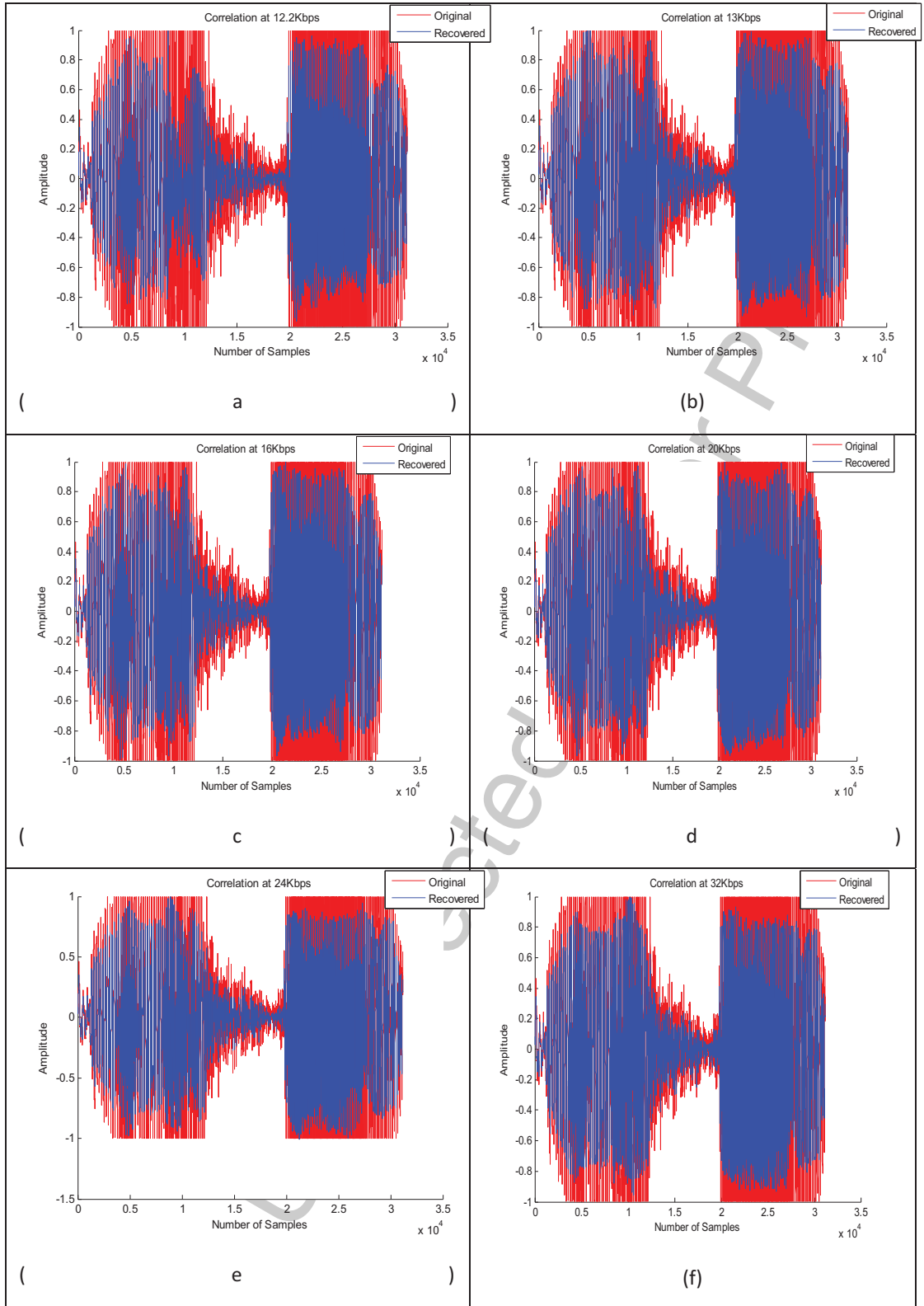


Fig. 3. (Continued).

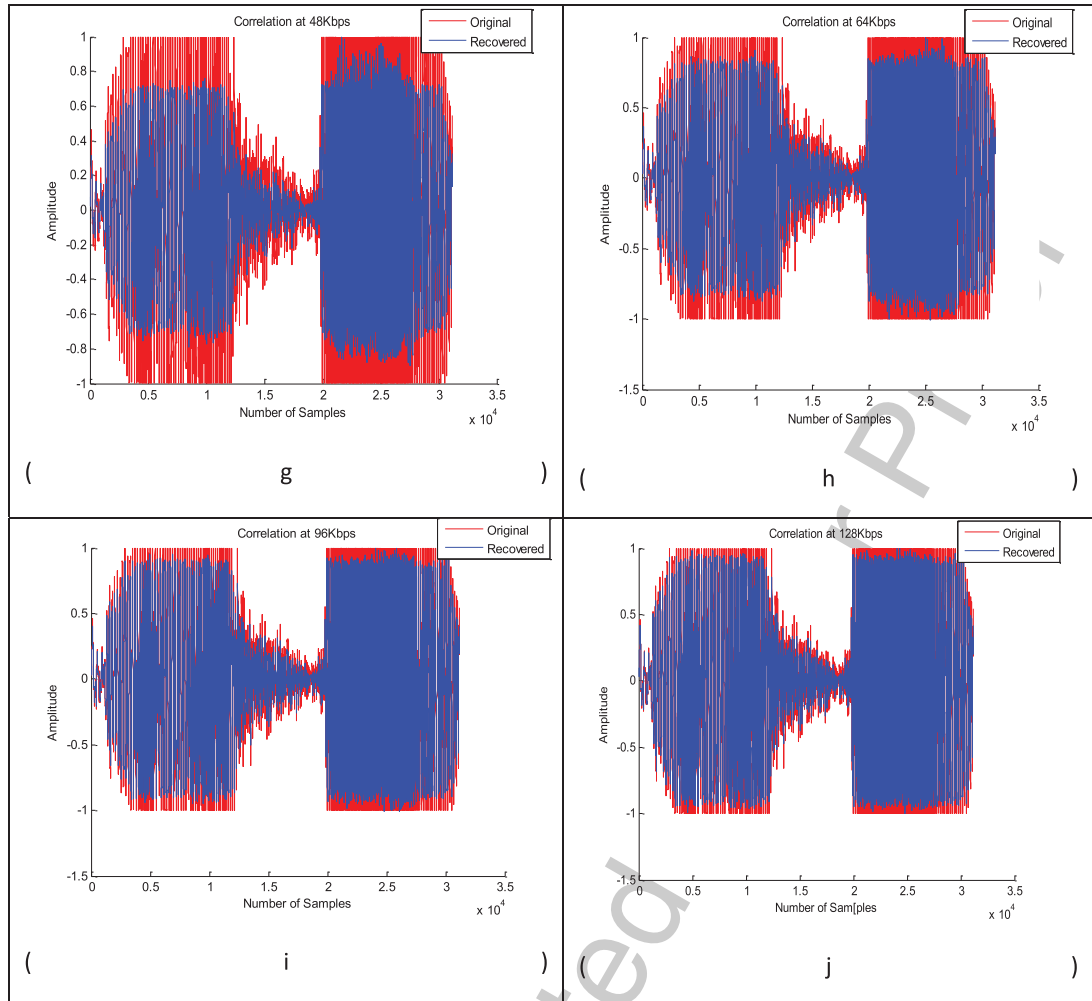


Fig. 3. Correlation between I/P and optimized codec O/P Signal at 8 KHz Sampling frequency : (a) 12.2 KBps (b)13 KBps (c)16 KBps, (d)20 KBps, (e)24 KBps, (f)32 KBps, (g)48 KBps, (h)64 Kbps, (i)96 Kbps, (j)128 Kbps.

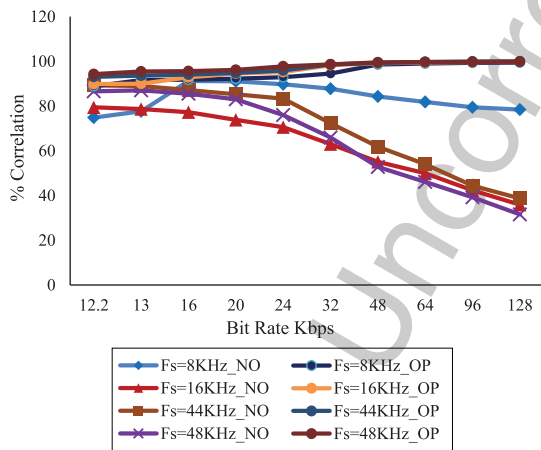


Fig. 4. % Correlation at multiple Bit Rates with respect to Sampling Frequencies. (NO - Non Optimized and OP - Optimized).

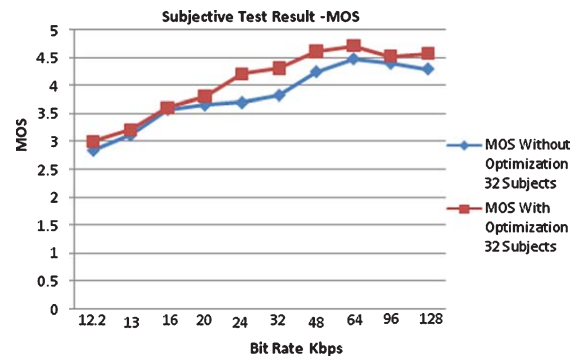


Fig. 5. Optimization effect on MOS Subjective Test.

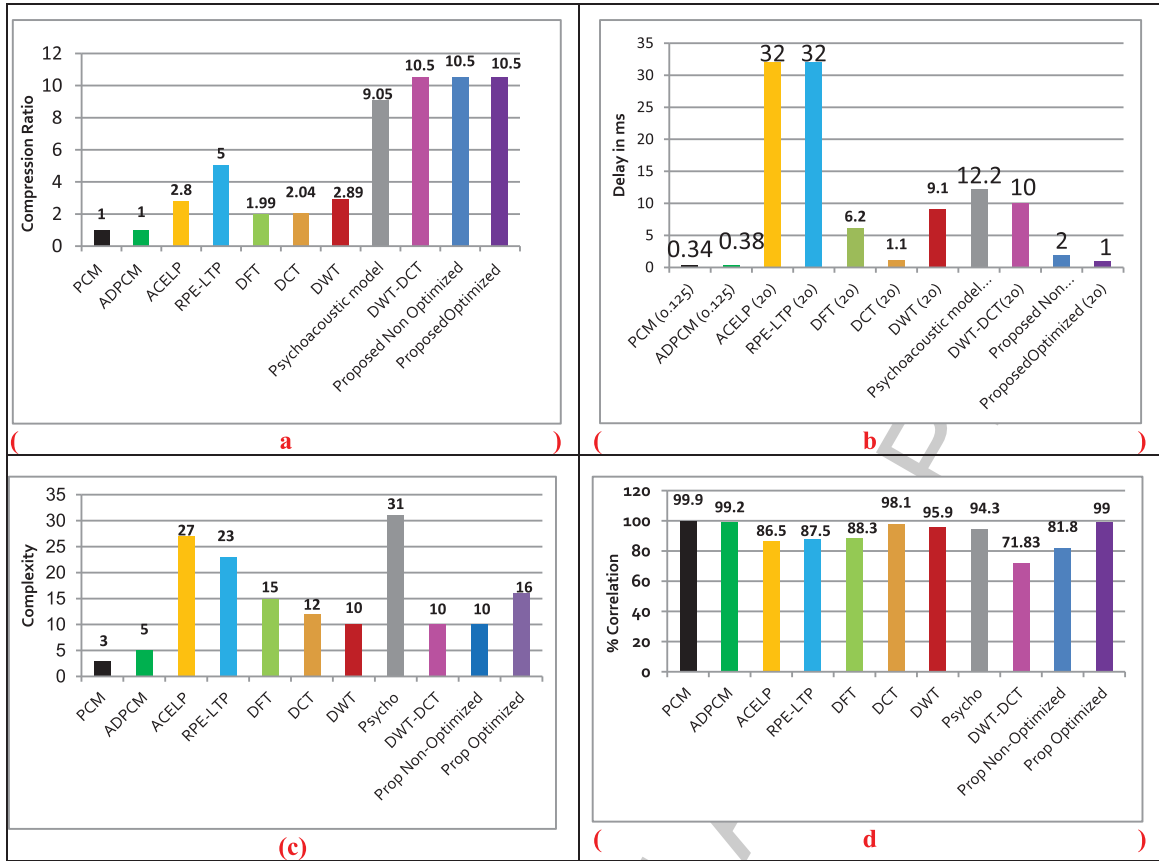


Fig. 6. Comparison of proposed speech coding technique with existing speech coding techniques w.r.t. (a) Compression Ratio, (b) Coding Delay, (c) Complexity and (d) % Correlation.

of complexity. The comparison of the existing and proposed techniques implementation in Matlab environment is shown in Fig. 6. All the existing coders are designed for fix value of bit rate hence can't be used for the application other than it is developed. The Software Defined Radio application of the speech coder needs to work for multiple bit rates which is satisfied by proposed technique. The complexity of proposed technique is increased because of application of optimization algorithm whereas the figure is still below the complexity measure of ACELP, RPE-LTP and Psychoacoustic speech coding techniques.

6. Conclusion

This paper presents the use of multi-resolution capabilities of the DiscreteWavelet Cosine Transform together with BFOA for wide range of bit rates as summarized in Table 2. Simulation result clearly shows the key features of proposed techniqueover

other coding techniques in the field of speech coding. The proposed technique out performs in all attributes at the cost of complexity. Hence, it can be effective speech coder for mobile communication application in Software Defined Radio to make efficient utilization of unused spectrum.

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