

# High-capacity, transparent and robust blind audio watermarking scheme based DWT and TLBO algorithm

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## Research

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# **High-capacity, transparent and robust blind audio watermarking scheme based DWT and TLBO algorithm**

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## **Abstract**

Digital watermarking is one of the best solutions again the copyright infringement, duplicates, verifies data and illegal distribution of digital media. Recently, the protection of digital audio signals is one of the attracting and interesting topics for scientific and researchers. In this paper we propose a blind audio watermarking mechanism in which it has high capacity, transparency and resistance simultaneously based on digital wavelet transform (DWT) algorithm. The key principle of this work is that in the DWT procedure, using two filters; break down the original audio signal into several sub-bands and transform them on a specific frequency range. It should be noted that the 8 bits of watermarked signal is selected and transform to the original signal. In order to increase the watermarking resistance, framing the high frequency coefficients of the third level of the wavelet and calculate the frames average and place them in the frame memory prime. Moreover, Teacher Learning Based Optimization (TLBO) algorithm used to determination of embedding and extraction coefficients in order to increase the SNR ratio in the embedding process and decrease the bit error rate (BER) in the extraction process. This method increases the embedding payload capacity while the audio SNR and extracted image BER have good qualify. Moreover, experimental results shown that this method has 13kbs hiding rate, ascendency imperceptibility, good payload capacity and intense robustness when resisting against various attacks such as MP3 compression, re-quantization, low-pass filtering, amplitude scaling, re-sampling, echo addition and noise corruption.

**Keyword:** Blind Audio Watermarking; Payload capacity, Transparency, Resistance, TLBO optimization Algorithm, DWT transform

## **Introduction**

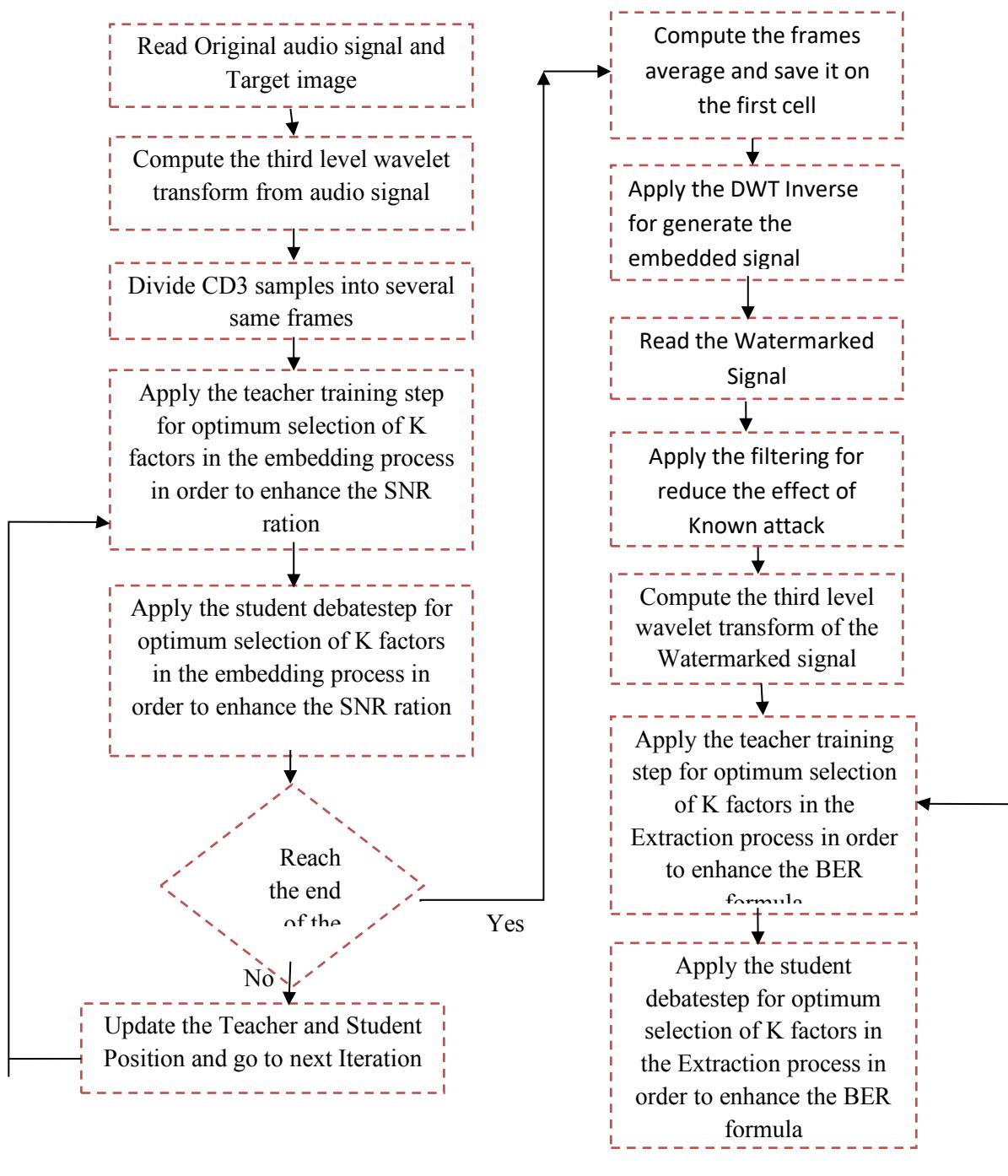
Due to extremist development of the multimedia technologies and Internet, it is appropriate to convey digital information throughout the world swiftly. Nevertheless, the information securement obstacle has become a global matter to be solved. For protection of media content on topics of fingerprint identification, copyright protection, medical security, broadcast monitoring and data authentication, digital watermarking is effective and interest algorithm in which nowadays become a hot and interest topic in the field of information security and communication. Multimedia signals such as image, video, image and data can be candidate as conveyer for digital watermarking algorithm. Since the audio signal includes less superfluous information, it is laborious to expand an audio watermarking algorithm. Moreover, by the extensive usage of the audio signal on the network, scientific focused on the audio signal for watermarking process. In the audio watermarking methods, the blind watermarking is the interest topics for researchers and scientific because in this procedure the primary audio signal is not necessary for extraction of watermarked image from the watermarked audio signal. Since human hearing system is not sensitive to small changes in the high-frequency components of the wavelet function, the encrypted watermark can be embedded into these sub bands. In this paper for increase the watermarking capacity, TLBO algorithm utilized to Embedding 8 bits of watermarks within the appropriate range of third level DWT coefficients. Since many researchers in this topic have exhibited, algorithms in audio watermarking should meet the following requirements: Imperceptibility, Robustness and capacity. Ramin and Hooman propose a real time audio watermarking based FFT transform and 2-bit embedding approach. Experimental results show that their method has good quality in term of payload capacity and SNR ratio [1]. Hu et al proposed an audio watermarking technique, by utilize the DCT method, with a payload capacity about 848 bit per second. There is different audio watermarking science unavailable paper in which represents sufficient efficiency from the perspective of capacity, robustness, and imperceptibility. An audio watermarking technique with a capacity of 848 bit per second, using the DCT domain, has been proposed by Hu et al[2]. In this work they have bring up sufficient robustness versus ordinary signal processing offensives; although, low pass filtering is yet disobey for this procedure [3]. Combination of Schur decomposition and DCT transform was proposed as a new hybrid method in the audio Watermarking area and the desired practical results were obtained [4]. Seyed Mostafa and et al suggest a new watermarking scheme based Lucas regular math sequence and 2-bit embedding procedure. In the represent methodology, audio watermarking scheme was achieved based on the quantity of the energy slight in the Fast Fourier Transform domain [5] HWAI and TUNG propose a high performance blind audio watermarking based FFT transform and AVNM scheme to improve the imperceptibility, robustness, and payload capacity simultaneously and prepare an audio watermarking algorithm with capacities area from 344.53 to 1033.59 bps [6]. The embedding is achieved by exploit FFT coefficients. The experimental results demonstrate that this algorithm depicts sufficient transparency. The robustness versus MP3 agglomeration of their algorithm is apperceived to be extremely sufficient; although, this algorithm was not appraised vis-à-vis further attacks. Mosleh and et all suggested a Multi-objective blind audio watermarking schemes based genetic algorithm where is simultaneously improves the capacity, transparency and persistence factors [7]. Mosleh showing that with optimization of frame length and frequency sub-band the payload capacity with 2002 bit per second can be

received where is the transparency and robustness have good qualify. Arashdeep and Malay present a robust smart audio watermarking with high embedding capacity based Tamper detection and DWT transform. In the proposed study Daubechies wavelets have been used to protect system from signal-processing attacks and they are introduced a successful method with high capacity about 4884 bit per second [8]. Although, the topic that keeps is that it is hot challenging to simultaneously attain high capacity and sufficient robustness versus challenging signal processing attacks like jittering, compression, and lopping within conceptual restrictions. The primary allotment of this survey is a secure and high-capacity audio watermarking mechanism that represents sufficient robustness against challenging signal processing attacks. The strategic selection of this paper is that two step optimization algorithm used to determine the optimum sub-band in the embedding process and the extraction coefficients in the extraction process in which the highest feasible payload capacity whit the good audio SNR and extracted image BER achieved. Evolutionary algorithms are a set of optimization algorithms in which do not require sophisticated mathematical operations such as derivative and integral to obtain the best response [9-11]. One of the interest evolutionary algorithms is TLBO algorithm. Then the watermarked signal is converted into a bit sequence and TLBO algorithm used to placing each 8 bit of the watermarks in the appropriate location of each frame. In the other words, TLBO algorithm determine the embedding wavelet coefficients in the embedding process and the extraction coefficients in the extraction process where is we achieve the SNR ratio up to 22 db and decrease the BER rate under 0.01. In this work TLBO algorithm used to determination of embedding and extraction coefficients in order to increase the SNR in the embedding process and decrease the bit error rate in the extraction process in an 8 bit blind audio watermarking system. The robustness of the suggested mechanism is arriving by utilize the Daubechies wavelets in the selected sub-band. The capacity of 13000 bit per second by sufficient robustness has been received. The embedding calibration subordinate is sufficient to receive signal-to-noise ratio (SNR) quantity superior than 20 dB for whole the audio patterns. The suggested algorithm is viewed to represent extremely sufficient robustness versus jittering, noise, compression, cropping, resampling, re-quantization and low pass filtering at such high payloads. As mentioned above, one of the most important challenges in the audio marketing system is to provide a useful approach to increasing the correlation between the three criteria of transparency, resistance and payload capacity. In this paper, we propose a new audio watermarking scheme based on the DWT transform and TLBO optimization algorithm. The different sections of the article are as follows: In the first step, the introduction section was presented. In the second part, DWT transform and TLBO algorithm will be presented. In the third part, the proposed method is introduced. Simulation results and experimental data are presented in the fourth part of the paper. Finally, the paper ends with a conclusion section.

## METHODS SECTION:

The implementation process of the proposed plan is in accordance with the flowchart drawn in the Fig.1 In the first step, the program read the host audio signal and the target image separately. In the second step, in order to increase the watermarking resistance, the third level wavelet transform is applied to the audio signal and produce CD3 coefficients . In the following, we frame the CD3 coefficients in order to increase

the transparency of algorithm. In the fourth step, to increase the payload capacity, the placement of the hiding location performs using the teacher training optimization (TLBO) algorithm. Up to this point, the watermarking signal has been generated. In the next step, in order to extract the hidden image, inverse wavelet transform apply to the watermarked signal in three steps. After extracting the high frequency sub-band coefficients, we frame them and TLBO algorithm utilized to optimized the proposed relationship at the extraction stage. Finally, we calculate the binary equivalent of the extracted data and extract the hidden image.



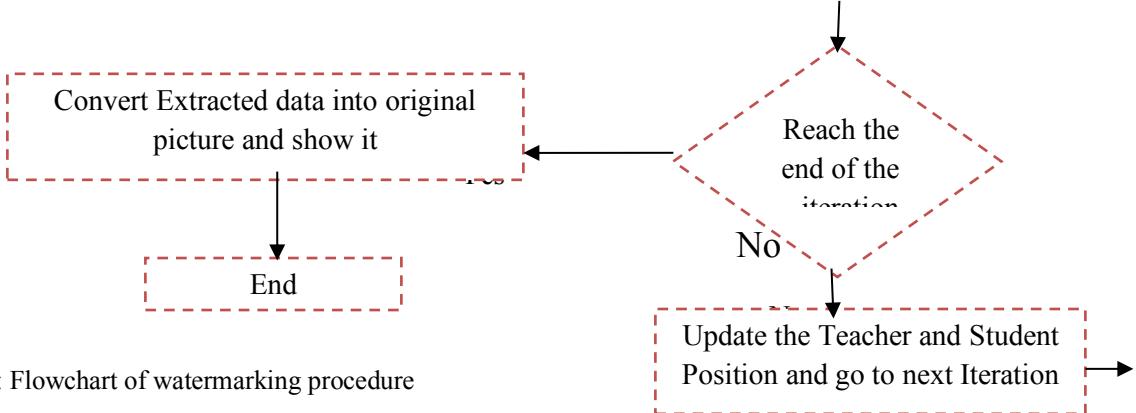


Figure.1: Flowchart of watermarking procedure

## Discrete Wavelet Transformation

In the last decade, Researchers have done many studies about wavelet transform [12]. In the following, many applications such as detection, communications and compression, were implemented based on wavelet transform. The fundamental concept in the DWT for a one dimensional signal is the following. A signal is split into two parts, one section is related to the high frequencies and other part is related to the low frequencies. The edge ingredients of the signal are commonly bounded to the high frequency section. The low frequency section is fracture anew into two sections of low and high frequencies.

This technique is a sequential voluntary issue of times, where is generally distinguished by the application at hand. In addition, through these DWT coefficients, the primary signal can be renovated. This renovation procedure is entitling the inverse of DWT (IDWT). The IDWT and DWT can be mathematically calculated as follows [12]:

$$H(w) = \sum_k h[k]e^{-jk\omega}, \quad G(w) = \sum_k g[k]e^{-jk\omega} \quad (1,2)$$

In which G and H are a high-pass and a low-pass filter, respectively, where convince a specified situation for renovation to be presented later. X[n] is signals that can be disintegrate recursively as :

$$A_j - 1.k = \sum_k h[n - 2k]A_j.n \quad (3)$$

$$D_j - 1.k = \sum_k g[n - 2k]C_j.n \quad (4)$$

here the coefficients,  $A_{j0,k}, d_{j0,k}, d_{j0+1,k}, \dots, d_{j,k}$ , are called the signal DWT, in which  $A_{j0,k}$  is the nethermost resolution part of  $x[n]$  and  $d_{j,k}$  are the details of  $x[n]$  at various bands of frequencies. Furthermore, the signal  $x[n]$  can be reconstructed from its DWT coefficients recursively [20]

$$A_j.n = \sum_k h[n - 2k]A_j - 1.k + g[n - 2k]D_j - 1.n \quad (5)$$

The three levels of discrete wavelet transform decomposition are shown in Fig.2. A1 and D1 are the coefficients of the first X signal decomposition level [n]. At the second level A1 decomposes to A2 and D2 and at the third level decomposes to A3 and D3.

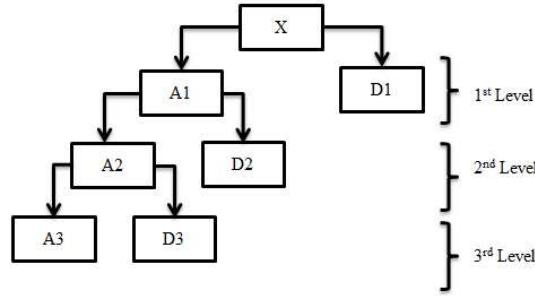


Figure .2: three levels of discrete wavelet transform decomposition

### **TLBO (teacher learn based optimization) Algorithm**

In recent years, meta-heuristic algorithms have been used to optimize the engineering problems. These algorithms are either modeled based on natural phenomena (such as ant colony and birds' algorithms) or sample human social exchanges (such as Imperial competition algorithms and teacher learning algorithm). The most important advantage of these algorithms is that they are simple and do not require complex mathematical problems such as derivative and integral. Teacher learn based optimization algorithm is an interest algorithm for optimizing engineering issues where is modeled based on the teacher training in the classroom. This algorithm has two training steps. The first step is based on the teacher training and second one is based on the student debate after the end of the class. In the first phase, the person who has the best answer in the population is selected as the teacher ( $X_{teacher}$ ) and other members of the population are known as students ( $X_i$ ). In the following, calculate the average position of the students ( $X_{mean}$ ). The reason for calculating the student knowledge average is that the teacher gives the training according to the average level of the class. By considering "r" as a random number as well as  $T_f$  as a constant coefficient, it is possible to model the movement of students in the first step by the following relation [13-16]:

$$X_{new} = X_i + r.(X_{teacher} - T_f \cdot X_{mean}) \quad (6)$$

Here  $X_i$  and  $X_{new}$  are the current and the new situation of the students respectively,  $T_f$  is a training factor that is considered as 2.

In the second stage, the teaching process is the responsibility of the students, so that each student selects another student randomly and shares knowledge with each other's and also updates his / her position; thus trying to use the other student information to raise his / her level of awareness and knowledge. This phase can be modeled as following formulations [13-16]:

$$X_{new} = X_i + r(X_i - X_j) \quad \text{if } f(X_i) < f(X_j) \quad (7)$$

$$X_{new} = X_i + r(X_j - X_i) \quad \text{if } f(X_j) < f(X_i) \quad (8)$$

Where in this stage, the move is made if the new position is better than the previous position. Moreover, the condition for the termination of this algorithm is to reach the end of the iteration. The algorithm will continue until the termination condition is met. The method proposed in this study is a blind audio watermarking technique which is based on applying the Discrete Wavelet Transform (DWT) on the digital audio signal. The algorithm consists of two procedures; watermarking embedding procedure and watermarking extraction procedure

## Watermark Embedding procedure:

In this section, a blind audio watermarking scheme with high-capacity, transparent, and robust features based on the combination of DWT and TLBO algorithm is presented. The embedding procedure performs four major operations: segmentation of the original audio signal, transformation of the audio signal, find the best interval for embedding and watermark embedding. In the segmentation of the original audio signal there is pre-processing of the audio signal, that is calculation of no. of samples, no. of bits presents in the samples. In the second step apply the transformation technique. Here applying the DWT transform for audio signal frequency decomposition. In this work TLBO algorithm utilized for find the best sub-band in which the SNR has good qualify. In the fourth step, we carry out the watermark embedding in the DWT-transformed audio signal. Before the embedding procedure, the image converts to a one-dimensional sequence of bits. We then use the Eighth-order Dabchi wavelet transform to analyze the host audio signal. 3-level sound decomposition is used for break down the audio signal. If the level decomposition is low, the algorithm resistance decreases and the watermarking capacity increase. Since human hearing system is not sensitive to small changes in the high-frequency components of the wavelet function, the encrypted watermark can be embedded into third-level high frequency sub-band (CD3). The reason for choosing these sub bands is their great resistance again common attacks such as noise and mp3. To increase the transparency and robustness of the proposed algorithm, we subdivide the coefficients of this sub-band into the number of frames. If the number of frames is increases dramatically, the algorithm transparency and resistivity becomes high and low respectively. Moreover, the average of each frame calculated as following:

$$m_i = \frac{1}{s} \sum_{j=(i-1)s+1}^{is} |c_j| \quad (9)$$

Where  $C_j$  is sub-band coefficients of wavelet transform,  $s$  is the frame length and  $m_i$  is mean of the  $i$ -th frame. Watermarking wavelet coefficients are calculated using the following equation.:

$$C_1(i, j) = -K_1 \times m(i) + 2 \times K_2 \times m(i) \times \frac{\text{double}(eq)}{2^8} + K_3 \quad (10)$$

In this equation, ‘‘8’’ is the number of bits where stored in each sub-band and corresponds to the details of the third wavelet level, eq is the decimal base of stored bits number and K factors are embedding interval. If k factors are growth the capacity embedding increasedand its very growth reduces the algorithm resistance. In this paper TLBO algorithm is used to select optimum value for K factors to have reached a high signal to noise ratio. Signal to Noise Ratio (SNR) is a statistical difference metric which is used to measure the similitude between the undistorted original audio signal and the distorted watermarked audio signal. Therefore define cost function as following:

$$SNR = 10 \log_{10} \frac{\sum_{n=0}^{N-1} x^2(n)}{\sum_{n=0}^{N-1} [x(n) - \hat{x}(n)]^2} \quad (11)$$

Where  $x(n)$  corresponds to the original signal, and  $\hat{x}(n)$  corresponds to the watermarked signal.

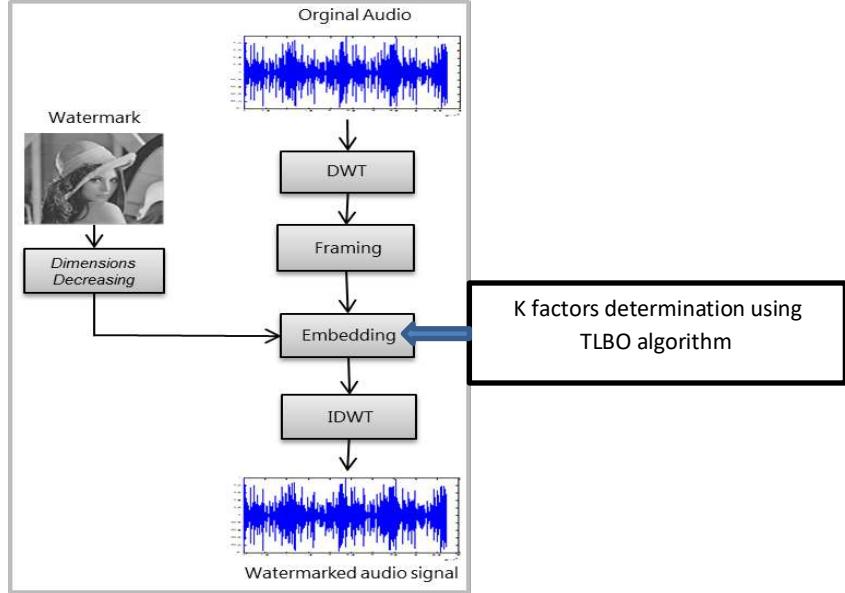


Figure 3: The watermark embedding procedure

## Watermark Extraction Procedure:

The watermark extraction procedure enables the owner of the audio clip to extract the embedded watermark. The first concerns plans which require the original signal to detect the watermark from the watermarked signal known as non-blind techniques and plans which are capable of recovering the watermark data without requiring access to the original audio known as blind techniques. At this point, there are two main operations:

In the previous step, the frame average was embedded in the first bit of each frame. Now calling average values and performs the extraction process using the following equations:

$$Ex = (C(i, j) + K_4 \times C(i, l)) \times \frac{(2^{nb} - 1)}{2 \times K_5 \times C(i, l)} + K_6 \quad (12)$$

$$dectoobin = dec2bin(round(Ex), n_b) \quad (13)$$

$$W_4(l : l + n_b - 1) = unit8(dectoobin(l : n_b) - 48) \quad (14)$$

First, the surface value where the data is stored inside it, eq, will be calculated then this value converted to an nb-bit binary number and finally the data is stored in its own location.

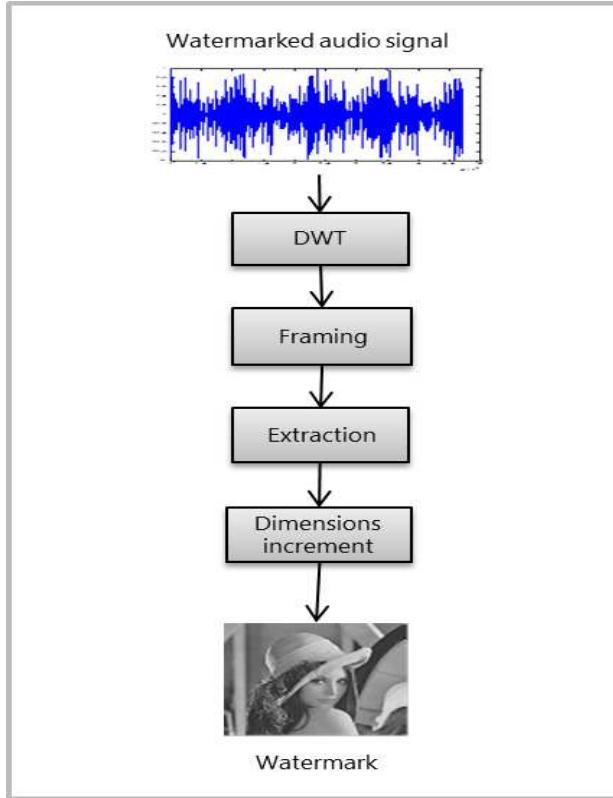


Figure 4: The watermark extraction procedure.

In this work Matlab 7.8 software used to implement the proposed audio watermarking based dwt transform. Loopy Music is an audio signal in which choose to be carrier signal. Lena picture used for watermark image. To implement the watermarking procedure an audio with 44100 hz frequency is samplized and quantized with 16 bits. Then, by applying a three-level wavelet transform, the audio signal is divided into four sub band. These sub band names are CA3, CD3, CD2, CD1 respectively. Here CD3 sub- band is used to hide watermarks. Moreover, in the wavelet transform utilize the 8-rder Dabchi filter because the experimental result shown that this method has good quality in order of resistance. Since the watermarked image product the distortion in the host audio signal alternative way to evaluate the algorithm qualify is assessment them in term of distortion value. Signal to Noise Ratio (SNR)is a statistical difference metric which is used to measure the similitude between the undistorted original audio signal and the distorted watermarked audio signal. The SNR computation is done according to following equation:

$$SNR = 10 \log_{10} \frac{\sum_n A_n^2}{\sum_n (A_n - A'_n)^2} \quad (15)$$

where A corresponds to the original audio signal, and A' corresponds to the watermarked signal. Although SNR is a simple way to measure the noise introduced by the embedded watermark and can give a general idea of imperceptibility, it does not take into account the robustness of the proposed algorithm. The robustness of the proposed algorithm to resist various attacks is evaluated using the bit error rate (BER), which is defined as:

$$BER = 1 - \frac{\text{Number of bits correctly extracted}}{\text{number of bits embedded}} \quad (16)$$

## RESULTS AND DISCUSSION SECTION:

The optimization algorithm was applied to optimize the k parameters. The following table shows the proceeds of proposed algorithm in terms of transparency and storage capacity.

Table 1: Optimization parameters

| K <sub>1</sub> | K <sub>2</sub> | K <sub>3</sub> | K <sub>4</sub> | K <sub>5</sub> | K <sub>6</sub> | BER    | SNR   | Capacity |
|----------------|----------------|----------------|----------------|----------------|----------------|--------|-------|----------|
| 1.3476         | -1.4365        | -3.0222        | 0.7095         | -1.6838        | -0.0395        | 0.0134 | 21.45 | 13k      |

We use a test method called Starmark benchmark to evaluate the robustness of the proposed algorithm. In this section, we compare the proposed algorithm with a number of previously presented algorithms. The algorithms have been selected tried to be from both high-capacity and high-strength algorithms.

Table 2: Results of 3 signals (robust against table 2 attacks)

| Audio File              | Time (m:sec) | SNR (dB) | Payload (bps) |
|-------------------------|--------------|----------|---------------|
| Beginning of the End    | 3:16         | 21.1     | 13000         |
| Citizen,Go Back toSleep | 1:57         | 20.3     | 13001         |
| Loopy Music             | 10           | 22.7     | 13008         |
| Average                 | 1:47         | 21.03    | 13005         |

Table 3: Robustness test results for five selected files and comparison with schemes in this

| Attack          | NC     | BER %    |          |     |     |      |      |
|-----------------|--------|----------|----------|-----|-----|------|------|
|                 |        | proposed | proposed | [9] | [5] | [11] | [12] |
| AddBrumm_100    | 0.9911 | 0.48     | —        | 0   | —   | 0.05 |      |
| AddBrumm_2100   | 0.9950 | 0.19     | —        | 0.1 | —   | —    | —    |
| AddBrumm_3100   | 0.9960 | 0.15     | —        | 0.2 | —   | —    | —    |
| Add FFT noise   | 0.7848 | 9        | 1_2      | 5   | —   | 1_2  | —    |
| FFT_HLPassQuick | 0.9925 | 0.29     | 0_2      | 5   | —   | 1_4  | —    |
| FFT_RealReverse | 0.9873 | 0.6      | 11_24    | 2   | —   | —    | —    |

|                       |        |      |       |   |      |     |    |
|-----------------------|--------|------|-------|---|------|-----|----|
| FFT_Stat1             | 0.3498 | 30   | 14_23 | 8 | _    | _   | _  |
| MP3_128               | 0.7763 | 10.1 | 0_9   | 2 | 37.1 | 0_5 | 14 |
| Requantization16to 12 | 0.8985 | 7    | 3_4   | _ | .    | _   | 1  |
| Resampling 44/22/44   | 0.9980 | 0.08 | 38_47 | 1 | .    | 5   | 2  |
| LSBZero               | 0.9986 | 0.4  | 0     | 0 | _    | 0   | 0  |
| RC_LowPass 2 to 22k   | 0.7810 | 9.1  | 0_4   | 0 | 12.7 | 0   | 3  |
| Amplify               | 0.9837 | 0.6  | 0_1   | 0 | 0    | 0   | 2  |
| Compressor            | 1      | 0    | _     | _ | _    | _   | _  |

Table 4: Robustness results for a variety of audio types under MP3 and RC Low-pass filter attacks

| MP3<br>attack           | MP3 rate  | 320 | 256 | 192 | 160  | 128  |
|-------------------------|---|-----|-----|-----|------|------|
|                         | BER   | 0.9 | 1.7 | 3.9 | 5.1  | 10.1 |
| RC<br>lowpass<br>attack | Cut-off<br>frequency of<br>low-pass filter<br>(kHz) | 20  | 15  | 10  | 5    | 2    |
|                         | BER   | 2   | 2.3 | 3.9 | 11.4 | 26   |

experimental results in the Tabels 1,2 , 3 and 4 shown that this method has 13kbs hiding rate, ascendency imperceptibility, good payload capacity and intense robustness when resisting against various attacks such as MP3 compression, re-quantization, low-pass filtering, amplitude scaling, re-sampling, echo addition and noise corruption

## Conclusion

In this paper we present a novel approach and high capacity blind audio watermarking based DWT transform. In this algorithm, at each high frequency coefficients of the wavelet transform, 8-bits of watermark signal is embedded. One of the strengths of our algorithm is to frames this sub-band coefficient to further increase the resistance, and then calculate the average of each frame. Next TLBO algorithm utilized to perform embedding in optimum location. In addition, our method is one of the blind methods, meaning that the original sound does not need to be retrieved when retrieving watermarks. Experimental results show that our proposed method has an excellent capacity (about 13 kbit / s) while maintaining the audio quality of the watermarked signal, SNR, about 21 db. This method has high resistance to common voice processing attacks such as MP3 compression attacks, filtering, reduced sampling rates, etc.

## **ABBREVIATIONS**

DWT: Digital Wavelets Transform

TLBO: Teacher Learning Based Optimization

BER: Bit Error Ratio

SNR: Signal to Noise Ratio

## **DECLARATIONS**

**“Data sharing not applicable to this article as no datasets were generated or analysed during the current study.”**

**“The authors declare that they have no competing interests”**

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**Author Contribution is: Ali Zeidi joudaki is corresponding author and Marjan Abdeyazdan is Supervisor of Ali and Mehdī Mosleh is Advisor of Ali in this work.**

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# Figures

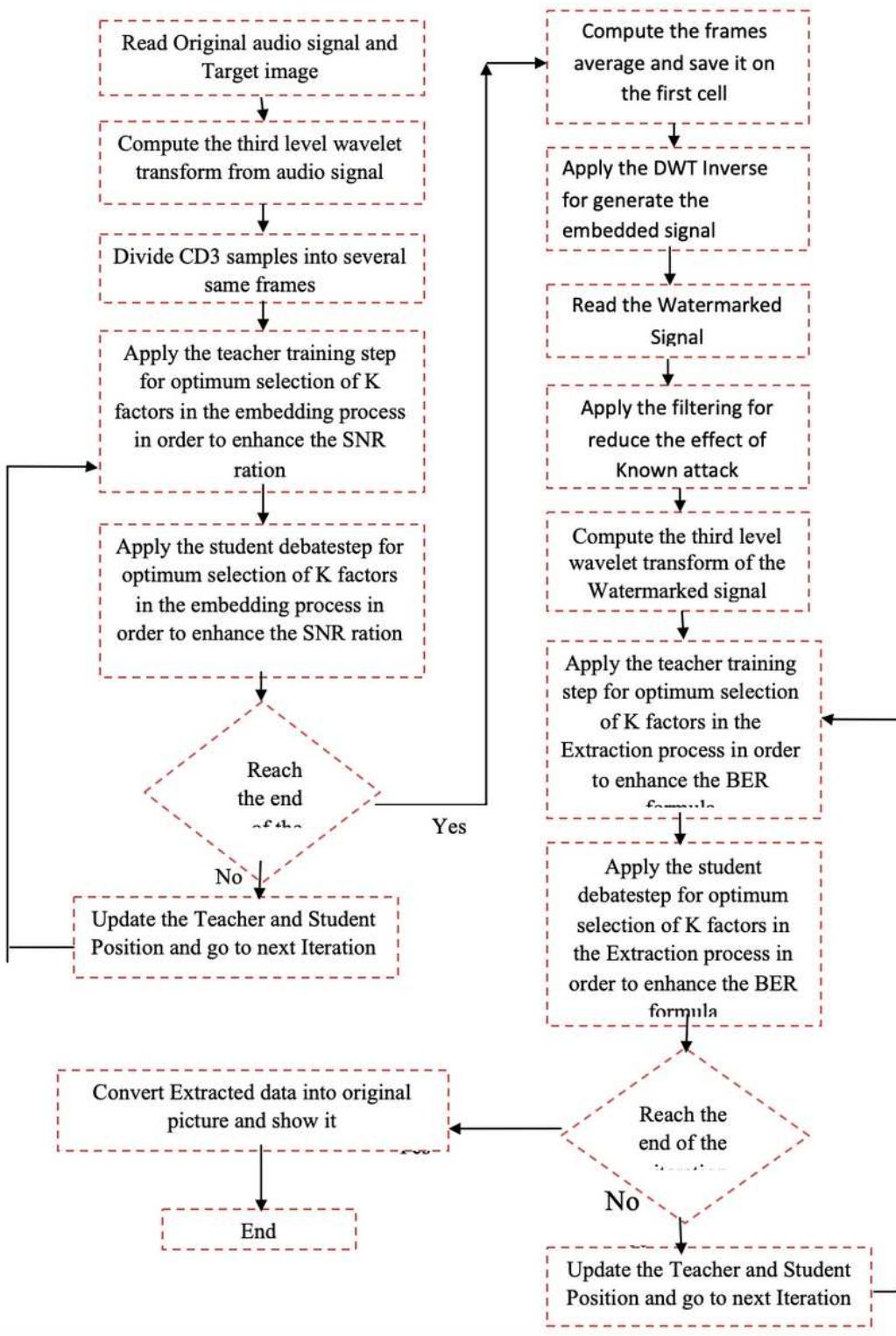
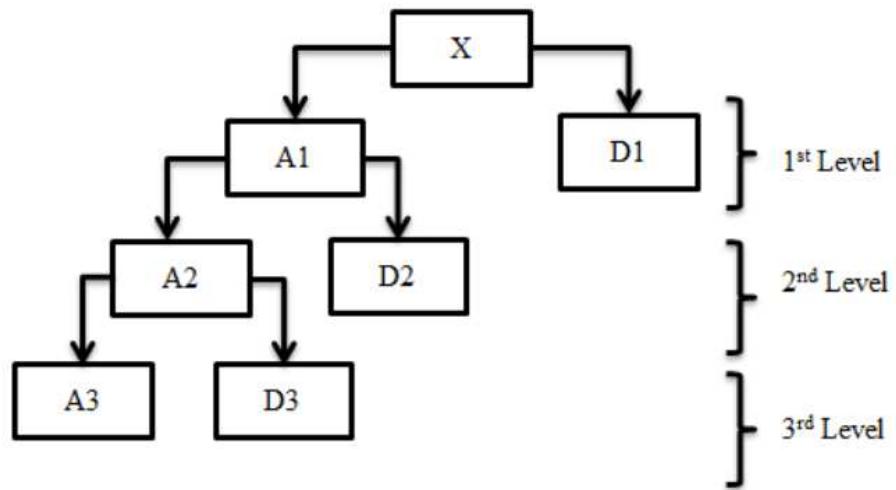


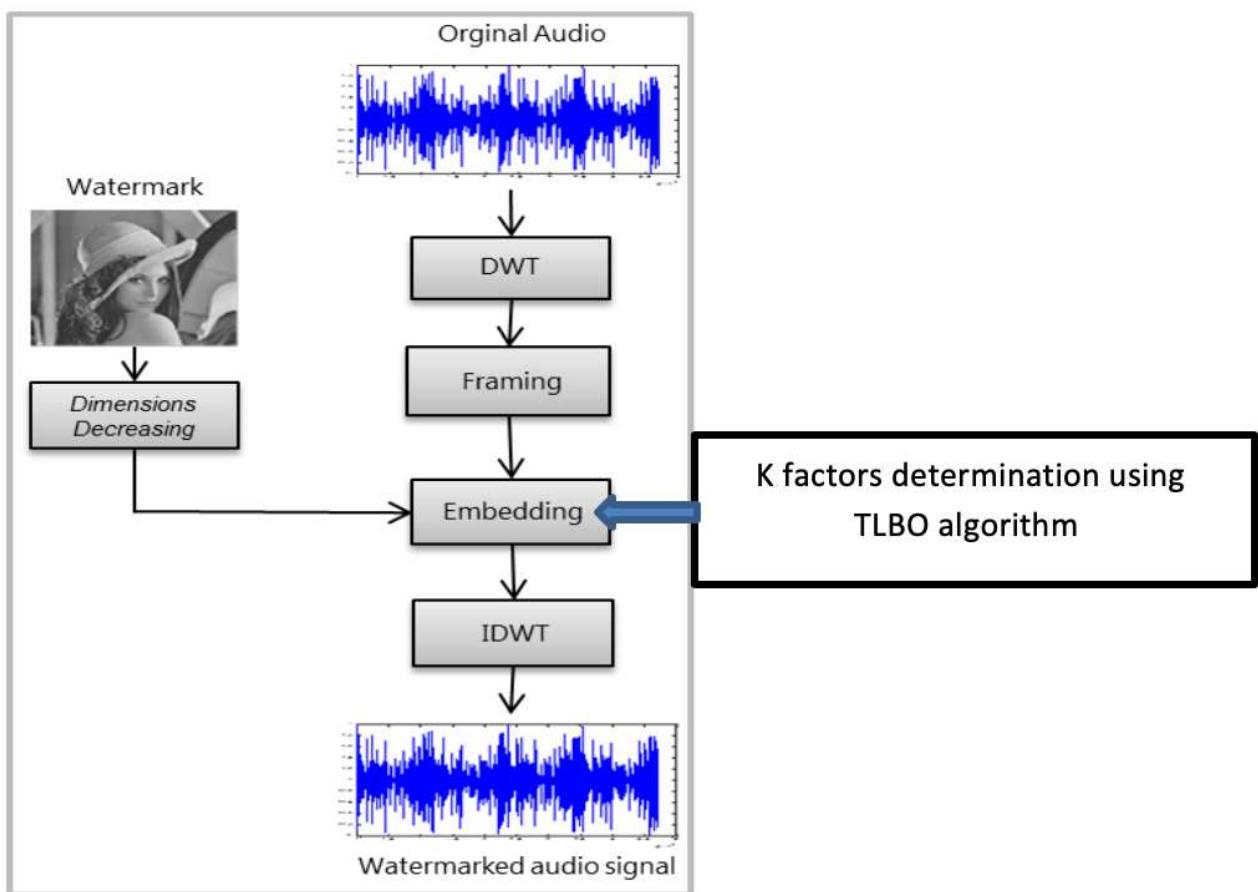
Figure 1

Flowchart of watermarking procedure



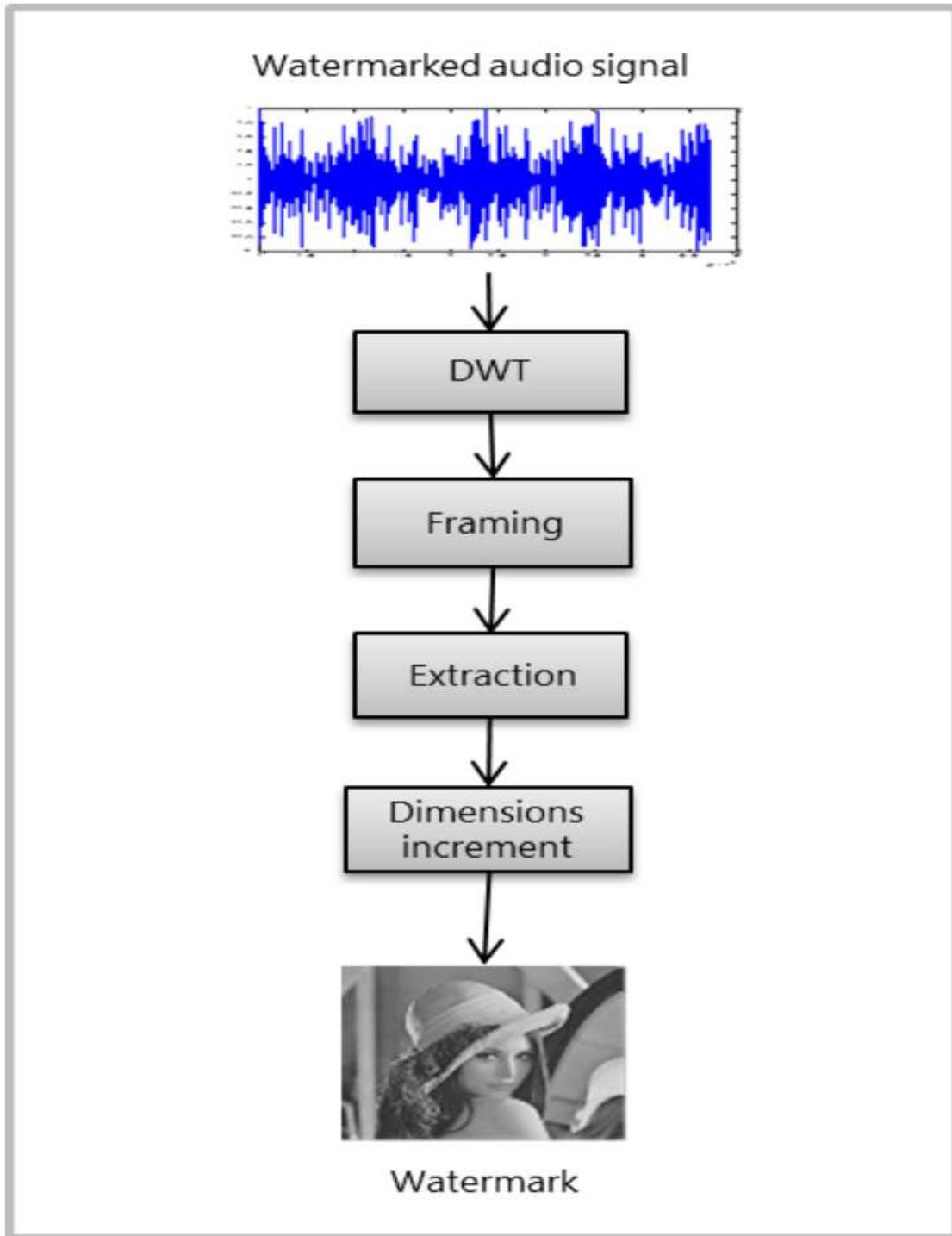
**Figure 2**

Three levels of discrete wavelet transform decomposition



**Figure 3**

The watermark embedding procedure



**Figure 4**

The watermark extraction procedure.