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Development of Algorithm for Audio Watermarking Embedding and Decoding using Patchwork Method under Attacks

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Abstract*:*To protect digital multimedia content from unauthorized reproduction, digital audio watermarking played crucial role. Audio watermarking for the patchwork method has a relatively good level of perception quality.The challengesbetween security, robustness, and imperceptibility is contemporary area of researchand remains relevant issues. This paper introduces discrete cosine transforms (DCT)-based audio watermarking process using the patchwork method for conventional and advanced signal processing attacks. In the first stage of the watermarking audio signal is divided into an equal number of segments and its sub-segments, and then its coefficients are computed. After eliminating high-frequency related coefficients, remaining coefficients are used to form frame pairs of equal length. Watermarks are embedded in a frame using specific criteria and secured data key.The adjustments are made in such a way that the identification ofWatermarked pairs of DCT frames is done in the decoding process by applying the selection criteria used during the embedding process. From watermarked frames, watermark data is extracted by using a secure data key. The proposed audio watermarking algorithm is implemented and tested under conventional and advance signal processing attacks for robustness, imperceptibility, security, and data payload.

***Keywords* :*Patchwork, Audio watermarking, Robustness, Discrete cosine transform*.**

I. INTRODUCTION

Day by day, Internet technologies are used more and more for the suffering of multimedia data on the internet; this is happening because of developments happens in electronic devices and internet technology. Hence, the multimedia data storage, transmission, manipulation, and distribution of electronic devices without any deterioration of their performance is possible. This leads to heavy demands for copyrighted data to avoid unethical access. Digital watermarking is a crucial technology in open network environments to protect copyright and authenticate authenticity [1].

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Technically, digital watermarking is deliberately designed to keep secret watermarking information in the actual media data without damaging its frequent usage. The copyright can be declared by the owners when necessary to extract watermarks data [2]– [4]. Video, image, and audio watermarking are the classification of digital watermarking techniques [2],[3],[5],[6],[7] based on the relevant areas. The main focused of this paper is the audio watermarking concept. It is a one-dimensional signal. Similarly, the perception that the human sounds perceived is more sensible than alternative perceptions of the sensory, such as vision [2][3][7]. Additional information can be concealed in audible signals without reducing the quality of the media data than other multimedia data. Imperceptibility, robustness, and reliability are becoming three essential features of an efficient and effective audio watermarking system. Imperceptibility signifies the almost inaudible information of the embedded watermark. Robustness means the ability to collect watermark information in the appearance or absence of signal processing attacks from the watermarked signal. Imperceptibility and robustness criteria are incompatible, but they must be coped with each other. Security means that in the watermarking process, a secret key should be used so that unauthorized people cannot remove watermarks without knowing the secret key.In addition to these features, there are other advantages of less computational complexity and sufficient data payload to improve the watermarking process. For time-overwhelming practical application (e.g., for the transmission of audio information via the net), an efficient watermarking scheme is particularly important, and a custom watermarking scheme fits different uses. Besides this, In the decrypting process, the watermarks embedded can be separated without using the audio signal of the host [1]. It is also desirable. Such type of audio watermarking is called blind audio watermarking.Several watermarking approaches for audio signals have been proposed in recent years. Such methods of audio watermarking include Least Significant Bit, supporting vector regression [7], [8], [9], Spread spectrum, patchworks [1], [7], [10], echo hiding [2],[11]. Existing all watermarking techniques, patchwork-based audio watermarking shows high robustness and have the great potential to survive against conventional as well as advanced attacks such as pitch and time scaling, cropping, re-quantization, amplification, re-sampling, filtering, noise addition, jitter and loss compression (e.g., Advanced Audio Coding (ACC) and MP3 attacks.). It can also attain a high level of imperceptibility and security.

Bender et al. [12] forecast the patchwork approach as a forthcoming tool for watermarking with high robustness to sophisticated signal processing attacks. The watermarking system was first presented in 1996 by Laurence Boney et al. for audio. This technique was then extended to an audio signal by Arnold [13].

With the improved patchwork algorithm, In-Kwon Yeo et al. [14] suggested his work based on domain transformations but fails to achieve a high bit rate. In a psychologically adaptable patchwork methodology, Hyunho Kang et al. [15] recommended a blind algorithm for watermarking using a full index integration to achieve an excellent rate. A robust multiplicative audio watermarking patchwork approach has been proposed by N.K. Kalantari et al. [16] but fails against advance signal processing attack.The audio watermarking algorithm Chi-Man Pun et al. [17] proposed and implemented using a Neural Network adaptive patchwork approach in two phases but fails against advance attack.I. Natgunanathan et al. [18] suggested the use of patchwork-embedding and decoding systems in the latest, most creative stage of digital audio watermarking with improved performance and analyzed mathematically by comparing practical audio signal measurements with theoretically determined values [19]. An enhanced audio watermarking algorithm has been suggested by Peng Hu et al. [20] for the Constant Q Transform (CQT) domain. Audio Quality perceptual assessment (PEAQ) and BER are used as quality parameters in the range of 0 – 1.2% to demonstrate the validity of the algorithm. In response to the similarity observes in the stereo signal, I. Natgunanathan et al. [21] proposed a new and most advanced stage of a digital audio watermarking algorithm for a stereophonic audio signal in the field of frequency. But does not provide a robust response to advanced attacks, such as de-synchronization, pitch, or time changes. So, it is observed that most of the research work proposed under some specific attacks. But no one considers all signal processing attacks because it needs to satisfy characteristics of audio watermarking such as robustness, imperceptibility, and data payload.

This paper recommends an audio signal patchwork-based watermarking framework. In the proposed strategy, the host audio signal segments are subdivided into two sub-sections, and then DCT of each sub-section coefficient is then computed. After that, the high-frequency variable DCT coefficients are removed, and remaining are divided into several framework pairs. Specific criteria used to choose DCT coefficient frame pairs which are suitable for watermark embedding. The DCT framework pairs are divided into several frames. The watermark data is inserted into the chosen pairs by modifying the corresponding coefficients under the influence of a pseudo-noise (PN) sequence utilize as a synchronization code and secured key.The integration algorithm has been developed to detect desired watermark frame pairs from the watermarked audio signal by using the same selection criterion of the encrypting phase. Using the PN sequence and secured key, one can extract watermarks,later on, finding the system pairs containing watermarks. The suggested patchwork scheme is somewhat superior to current watermarking strategies since the watermarked frame pairs at the watermark detection stage do not need more data and are more resistant to conventional and advanced attacks. The results of the simulation show its performance.

The respite of the paper has been organized accordingly. Section II introduces the latest patchwork based embedding and decoding system. Section III presents the effects of simulation to demonstrate the development's success. Section IV represents the conclusion.

II. EMBEDDING AND DECODING SCHEME

A. Embedding scheme

Next, to reach a variety of segments, the host audio signal is sampled and split up into an equal quantity of segments with the fixed-length R. where R is real and even number. Let consider $y(n)$ be the audio signal of 60 seconds. Let $y(n)'$be the host audio signal segment. Each audio signal subsegment further divided into subsegments of equal length L, which is real and even number. Then DCT coefficients of each subsegment are calculated by considering the following equation (1).

$$f(\zeta) = \frac{1}{\sqrt{L}} \sum_{n=0}^{L-1} y(n)' cos\left\{\frac{\pi(2n+1)\zeta}{2L}\right\} \qquad (1)$$

$Where \quad k = 0,1, \ldots, L-1$

DCT is a true transformation that transforms real data points into its real frequency range. It used to avoid the redundancy problem, or DCT function has the power to compress the spatial sequence energy to the fewest possible frequency coefficients. As we know that high-frequency coefficients are susceptible to signal processing attack, and hence high-frequency coefficients will not be considered at the time of the watermark embedding process. Only low and middle-frequency coefficients of each subsegment are considered for watermark embedding. Let $f_s(\zeta)$beselected frequency range coefficient from $f(\zeta)$. After removing, low and middle-frequency coefficients of each subsegment are used to form a frame consisting ofan equal number of coefficients. Let $f_s(\zeta)$is further divided the equal number of frames of equalP_{LS}length$f_{s,\rho}(\zeta), \rho = 1,2, \ldots \ldots, P_{LS}$

$$f_{s,\rho}(\zeta) = [f_{s,1}(\zeta), f_{s,2}(\zeta), \ldots \ldots, f_{s,P_{LS}}(\zeta)] \qquad (2)$$

Each frame is further segmented into two subframes of equal length $f_{s,\rho}(\zeta)$as given below:

$$f_{s,\rho,1}(\zeta) = [f_{s,\rho}(0), f_{s,\rho}(1), \ldots \ldots, f_{s,\rho}\left({}^{P_{LS}}/_{2} - 1\right)] \qquad (3)$$

$$f_{s,\rho,2}(\zeta) = [f_{s,\rho}\left({}^{P_{LS}}/_{2}\right), f_{s,\rho}\left({}^{P_{LS}}/_{2} + 1\right), \ldots \ldots \ldots, f_{s,\rho}(P_{LS} - 1)] \qquad (4)$$

Thus, each frame consists

$$f_{s,\rho,}(\zeta) = [f_{s,\rho,1}(\zeta), f_{s,\rho,2}(\zeta)] \qquad (5)$$

When a set of frames has been created, a PN sequence $q(n) = \{q(1), q(2), q(3), \dots., q(M)\}$ ofthe frame size length is generatedfor security and synchronization purposes.

It is again scrambled with secret key $w(0)$ of one-dimensional sequence to provide higher security, which is one of the essential parameters for watermarking. Let ω(n) be the scrambled sequence generated by

$$\omega(n) = q\big(mod(n, M)\big) \oplus w(0) \qquad (6)$$

Where $\oplus$denotes the Modulo-2 binary operation and $mod(y, z)$ provides the remainder of the division y by z. Once the scrambled data generated, theyget combined with DCT frames. Then the succeeding equation is written for the corresponding frames equation

$$f_{s,\rho,1}(\zeta) = \{f_{s,\rho,1}(\zeta) + \omega(\zeta)\}$$
$$f_{s,\rho,2}(\zeta) = \{f_{s,\rho,1}(\zeta) + \omega(\zeta)\} \qquad (7)$$

In the framework pairs, several frame pairs may not be suitable for watermark insertion. The insertion of the watermark in those frame pairs could reduce to an unacceptable level of the perceptual characteristicsof theaudio watermarked signal. We define as follows, the mean for the absolute value of frames for selecting the appropriate frame pairs$|f_{s,\rho,1}(\zeta)|$ and $|f_{s,\rho,2}(\zeta)|$ as per equation (7).

$$M_{\rho,1} = \in\left(|f_{s,\rho,1}(\zeta)|\right)$$
$$M_{\rho,2} = \in\left(|f_{s,\rho,2}(\zeta)|\right)$$
$$M_{\rho} = \in\left(|f_{s,\rho}(\zeta)|\right) \qquad (8)$$
$$N_{\rho} = \frac{M_{\rho,1} + M_{\rho,2}}{2}$$

Where the expected value is representing by $\in$, and the mod value represents averaging value.

The i^{th} DCT segment is known as an un-silent segment when the means of the DCT segments absolute value$m_i > \delta$. We use all non-silent DCT segments to embed watermarks to boost embedding efficiency and ensure excellent perceptual reliability with constraints

$$M_{\rho} > \delta \quad and \quad N_{\rho} \geq \mu \qquad (9)$$

Where $\rho = 1,2, \dots \dots \dots, P_{LS}$ and δ and μ is a randomly chosen threshold value to control imperceptibility and robustness.

Once the above-specified equation (9) is fulfilled, then watermark bit 0 and 1 are inserted into the selected frame only by using specific rules. After adding watermark bits, frames coefficients $M_{\rho,1}$ and $M_{\rho,2}$are modified in terms of the mean value.

$$M'_{i,1} \quad and \quad M'_{i,2} \qquad (10)$$

For every i, subsegment coefficients modified employing the succeeding equation:

$$f'_{s,\rho,1}(\zeta) = f_{s,\rho,1}(\zeta) \times \frac{M'_{\rho,1}}{M_{\rho,1}} \qquad and$$

$$f'_{s,\rho,2}(\zeta) = f_{s,\rho,2}(\zeta) \times \frac{M'_{\rho,2}}{M_{\rho,2}} \qquad (11)$$

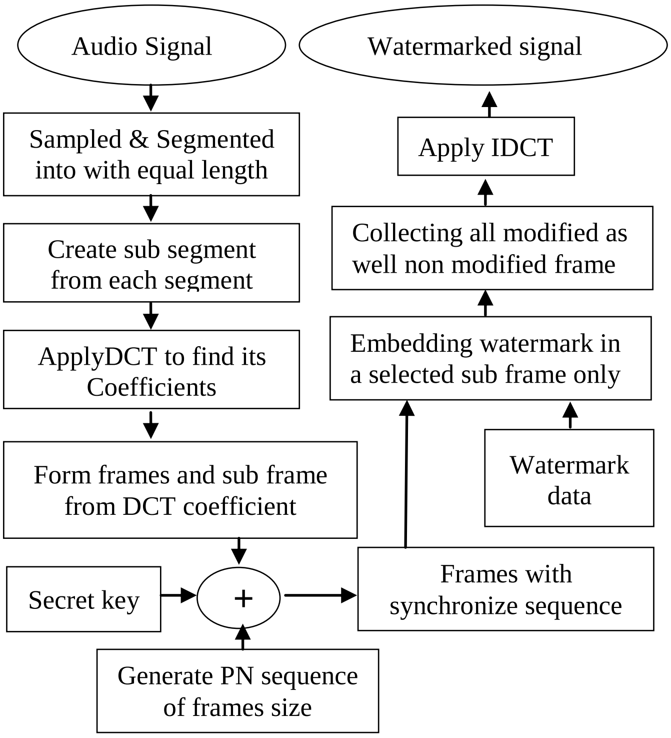

Fig.1. The process of the watermark Encryption

Then all counterparts marked with watermark can be obtained respectively:

$$f'_{s,\rho}(\zeta) = [f'_{s,\rho,1}(\zeta), f'_{s,\rho,2}(\zeta)], \quad \rho = 1,2, \dots \dots, P_{LS} \qquad (12)$$

and

$$f'_{s}(\zeta) = [f'_{s,1}(\zeta), f'_{s,2}(\zeta), \dots \dots.., f'_{s,P_{LS}}(\zeta)] \qquad (13)$$

In the same way, we can tuck the identical watermark bit in all picked out DCT coefficient frame pairs from the original host audio signal segment. After that, we are using the Inverse Discrete Cosine Transform (IDCT) to reconstruct the audio watermarked segment.

B. Decrypting scheme

Once the audio signal received, the next step is the equal partition of the audio watermarked signal, and then we use the sub-segmentation technique similarly used in the embedding procedure. To obtain its DCT coefficients, we utilized DCT operation to the received audio signal segments. Let $f(\zeta)'$ be the DCT coefficients of the segment.

Let $f_s(\zeta)'$is further divided the equal number of frames of equalP_{LS}length$f_{s,\rho}(\zeta)$, $\rho = 1,2, \dots \dots, P_{LS}$

$$f_{s,\rho}(\zeta)' = [f_{s,1}(\zeta)', f_{s,2}(\zeta)', \ldots\ldots, f_{s,P_{LS}}(\zeta)'] \quad (14)$$

Each frame is further segmented into two subframes of equal length$f_{s,\rho}(k)'$as given below:

$$f_{s,\rho,1}(\zeta)' = [f_{s,\rho}(0)', f_{s,\rho}(1)', \ldots, f_{s,\rho}\left({}^{P_{LS}}/_{2} - 1\right)'] \quad (15)$$

$$f_{s,\rho,2}(\zeta)' = [f_{s,\rho}\left({}^{P_{LS}}/_{2}\right)', f_{s,\rho}\left({}^{P_{LS}}/_{2} + 1\right)', \ldots, f_{s,\rho}(P_{LS} - 1)'] \quad (16)$$

Thus, each frame consists

$$f_{s,\rho,}(\zeta)' = [f_{s,\rho,1}(\zeta)', f_{s,\rho,2}(\zeta)'] \quad (17)$$

When a set of frames has been created, a PN sequence of frame size length is generated for synchronization and security purposes. It again scrambled with a secret key.It is necessary to detect out that the relevant DCT coefficient frame pair covers a watermark before trying to extract a watermark from a desired frame pair. So, it is indispensable to compute the means for the absolute value of frames pairs $|f_{s,\rho,1}(\zeta)'|\ and\ |f_{s,\rho,2}(\zeta)'|$.

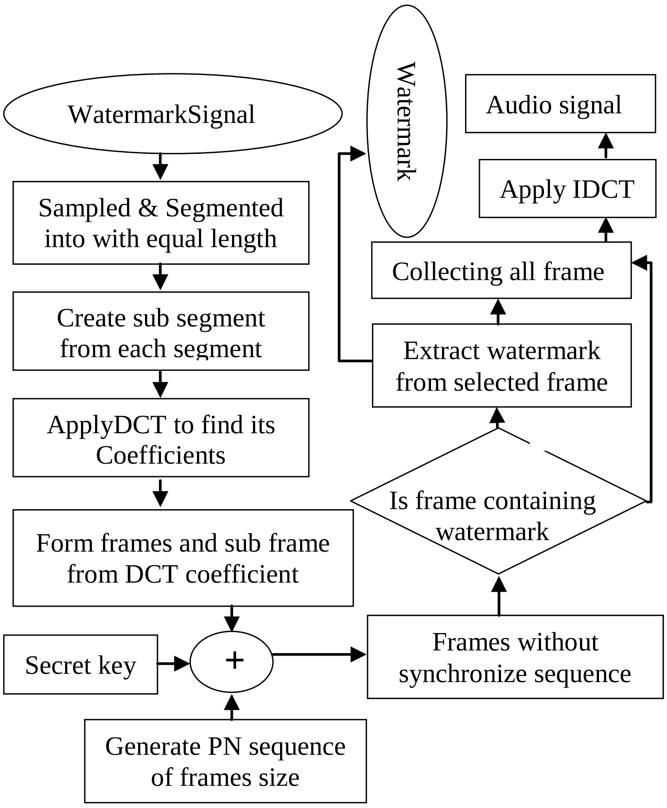

Fig.2. The process of the watermark decryption

$$M_{\rho,1}' = \in\left(|f_{s,\rho,1}(\zeta)|'\right)$$

$$M_{\rho,2}' = \in\left(|f_{s,\rho,2}(\zeta)|'\right) \quad (18)$$

$$M_{\rho}' = \in\left(|f_{s,\rho}(\zeta)|'\right)$$

$$N_{\rho}' = \frac{M_{\rho,1}' + M_{\rho,2}'}{2}$$

If $M_{\rho}' > \delta\ \ and\ \ N_{\rho}' \geq \mu$ is satisfied, then it indicates that the respective frames pair consist of watermark otherwise not. Based on this, it is possible to find all such frame pairs with Watermark data. The watermark can be removed by using specific rules. In this way, watermark extracted otherwise watermark not extracted.

III. SIMULATION RESULT AND DISCUSSION

Simulation examples demonstrating the output of the proposed scheme are given in this paragraph. We have 20 audio-clips, as mention in Table I are randomly selected from different groups as host signals in the simulations. Table II shows the simulation results in terms of performance analysis.

Table- I: Different host audio signal for evaluation

Host Audio Signals	Style of Music
AU1 - AU4	Classical
AU5 - AU8	Pop
AU9 - AU12	Instrumental
AU13 - AU16	Jazz
AU17 -AU20	Bollywood

The audio signals are more than 60 seconds in duration. Samples are taken at 44.1 kHz, are quantified, and then segmented at 16 bits.The DCT coefficient is chosen within a frequency range of less than 5.5 kHz. The selected frame for encryption consists of 500 samples. A proposed watermarking strategy should be reliable to traditional as well as advanced attacks while retaining the high quality of sensory operation. The physical property of the proposed watermark depends on the parameters δ, μ of the watermarking.Depending on the chosen parameter value, the embedding power of the proposed method can be estimated as 5 to 10 bps.The imperceptibility verified by using the following parameter:

1. **Signal to Noise ratio (SNR)**: As per standard mention by the International Federation of the Phonographic (IFPI) standards, it required greater than 20dB. Here we predicted imperceptibility by using SNR.

$$SNR = \frac{Original\ audio\ signal}{Watermark\ audio\ signal}$$

The average SNR ismore than 25dB for all twenty audio clips is observed.

This illustration assesses the watermark robustness implementation against traditional as well as advanced attacksdepends on the parameters δ, μ.

Table- II: Performance parameter BER for Suggested scheme

Attacks	Host audio signal	BER
Close-loop	AU1 - AU4	0.10
	AU5 - AU8	0.11
	AU9 - AU12	0.13
	AU13 - AU16	0.15
	AU17 -AU20	0.12
Re-sampling	AU1 - AU4	0.20
	AU5 - AU8	0.21
	AU9 - AU12	0.30
	AU13 - AU16	0.20
	AU17 -AU20	0.19
Low pass filtering	AU1 - AU4	0.11
	AU5 - AU8	0.13
	AU9 - AU12	0.10
	AU13 - AU16	0.14
	AU17 -AU20	0.12
High pass filtering	AU1 - AU4	0.18
	AU5 - AU8	0.13
	AU9 - AU12	0.17
	AU13 - AU16	0.18
	AU17 -AU20	0.11
Amplitude	AU1 - AU4	0.10
	AU5 - AU8	0.12
	AU9 - AU12	0.10
	AU13 - AU16	0.10
	AU17 -AU20	0.12
Pitch scaling	AU1 - AU4	0.18
	AU5 - AU8	0.21
	AU9 - AU12	0.20
	AU13 - AU16	0.20
	AU17 -AU20	0.19
Time scaling	AU1 - AU4	0.21
	AU5 - AU8	0.23
	AU9 - AU12	0.29
	AU13 - AU16	0.23
	AU17 -AU20	0.20
Jitter	AU1 - AU4	0.23
	AU5 - AU8	0.24
	AU9 - AU12	0.20
	AU13 - AU16	0.22
	AU17 -AU20	0.28
Noise	AU1 - AU4	0.29
	AU5 - AU8	0.31
	AU9 - AU12	0.23
	AU13 - AU16	0.25
	AU17 -AU20	0.15
Echo	AU1 - AU4	0.30
	AU5 - AU8	0.27
	AU9 - AU12	0.29
	AU13 - AU16	0.28
	AU17 -AU20	0.27
Cropping	AU1 - AU4	0.32
	AU5 - AU8	0.31
	AU9 - AU12	0.27
	AU13 - AU16	0.28
	AU17 -AU20	0.30
Re quantization	AU1 - AU4	0.21
	AU5 - AU8	0.22
	AU9 - AU12	0.20
	AU13 - AU16	0.19
	AU17 -AU20	0.25
MP3	AU1 - AU4	0.10
	AU5 - AU8	0.10
	AU9 - AU12	0.11
	AU13 - AU16	0.10
	AU17 -AU20	0.10
AAC	AU1 - AU4	0.11
	AU5 - AU8	0.10
	AU9 - AU12	0.11
	AU13 - AU16	0.11
	AU17 -AU20	0.10

The robustness is measured with the following definition:

2. Bit error rate (BER):

$$BER = \frac{Watermark\ bit\ incorrectly\ detected}{Watermark\ bit\ inserted}$$

For robustness analysis, the following conventional and advance attacks are used:

Closed-loop or no attack: The data of the watermarks is stripped from the watermarked audio signals without attack

Re-sampling attack: Watermarked signal Sampled down to 16 kHz and return sampling frequency 44.1 kHz.

Low-pass filtering (LPF): For signals with watermarking, a low-pass filter with a cutting frequency of 12 kHz is added.

High-pass filtering (HPF): Here, the watermarked signals are applied with high-pass filters with 50-100 Hz cutoff frequency.

Amplitude attack: Here, 1.2 times increased the amplitude of the watermark signals.

Pitch scaling attack: Watermarked audio signals get shifted 110% in pitches.

Time scaling attack: Watermarked audio signals get shifted 110% in times.

Jitter attack: Remove a sample randomly from the watermarked signals for every 5,000 samples.

Noise attack: Random noise with SNR of 20 dB added to the watermarked signal.

Echo attack: In this attack, the echo of the original watermark signal is incorporated in the watermark information.

Cropping attack: First, 100 Samples of watermarked audio are deleted signals.

Re-quantization attack: For watermarked signals,each sample is quantized between 16 bits to 8 bits.

MP3 attack: TheMPEG 1 Layer III audio compression process is carried out on watermarked signals.

AAC attack: The advanced MPEG 4 audio compression process is carried out on the watermarked signals.

The experimental results shown in Table-II, indicates that the watermark is not much affected by traditional attacks but get slightly affected due to advanced signal processing attack.The watermark continues to be imperceptible for a high-energy audio signal even though the low-frequency coefficients are changed. Because the components with relatively low frequency are perceptually important, there is not much data loss in these areas, and watermarked stability can be improved. Thus, for the specific value of $\delta = 0.2\ and\ \mu = 0.1$, we get an optimized value between imperceptibility, robustness, and Payload. If we exceed this value, any one of the parameters gets affected slightly. i.e., Robustness and Payload increases, and imperceptibility decreases and bit error rate increases.

IV. CONCLUSION

This paper proposed a suitable and reliable audio signals watermarking system based on the patchwork method, which incorporates watermark data on the basis of changing the DCT coefficient, for audio signal segments. The suggested scheme contains watermarks in acceptable DCT frame pairs only to make them extremely imperceptible.

The embedding formula intended in the encryption process by selecting specific frame pairsof DCT, while in the decrypting step same insertion is the straightforward way used to search out the watermarked pairs of frames.The design of the recommended scheme combined with the use of chosen frequency areas and various watermark encryption so it will guarantee a high level of robustness. The new strategy will additionally secure, thanks tothe requirement of employing a secured key within thedecryptingmethod. Results obtained from simulation for conventional and advance attacks show the superior performance of our system as indicated in the Table-II. Further, this research analysis can be extended by using the stationary wavelet transform to get better results.

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