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## Wavelet based Audio Compression using

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**ABSTRACT:** Audio compression (data), a type of lossy or lossless compression in which the amount of data in a recorded waveform is reduced to differing extents for transmission respectively with or without some loss of quality, used in cd and mp3 encoding, internet radio, and the like. Wavelets with large number of vanishing moments are useful for this audio compression method, because if a wavelet with a large number of vanishing moments is used, a precise specification of the pass bands of each sub band in the wavelet decomposition is possible. Wavelet compression refers to a method of reducing the size of data by removing high-frequency noise components, while also improving its quality. This technique allows for data to be compressed to less than 1% of its original size, resulting in faster computation and improved signal-to-noise ratio. A compressor is used to reduce a signal's dynamic range—that is, to reduce the difference in level between the loudest and quietest parts of an audio signal. Compression is commonly used to attenuate loud transient peaks (e.g., when a singer suddenly belts out a high note) to help maintain a consistent level. Audio compression using wavelets in MATLAB refers to the process of reducing the size of an audio signal (like a sound recording) while trying to maintain its perceptual quality, by using wavelet transforms. The core idea behind wavelet-based audio compression is to transform the audio signal into a different domain (typically the frequency domain), and then reduce the number of coefficients (or values) required to represent that signal. This is done by applying thresholding techniques to the wavelet coefficients, which helps eliminate less important data.

#### I. INTRODUCTION

In the modern era of digital signal processing, audio compression plays a crucial role in reducing storage requirements and transmission bandwidth while maintaining acceptable sound quality[1]. One of the advanced techniques used for audio compression is wavelet transform, which provides a powerful way to analyse signals at multiple resolutions. Unlike traditional methods such as Fourier transform, wavelet-based compression preserves both time and frequency information, making it well-suited for non-stationary signals like audio.

Wavelet-based audio compression exploits the fact that most audio signals contain redundancies and irrelevant information that can be efficiently removed without significant loss of perceptual quality. Some key advantages of using wavelets for audio compression include:

- Multi-resolution analysis: Wavelets can analyse different frequency components with varying time resolutions.
- Efficient energy compaction: Most of the signal energy can be concentrated into a few wavelet coefficients, allowing effective compression.
- **Better time-frequency localization:** Unlike the Short-Time Fourier Transform (STFT), wavelets can represent transients and non-stationary audio signals more accurately.

MATLAB provides a comprehensive set of tools for implementing wavelet-based audio compression. The Wavelet Toolbox in MATLAB offers functions for wavelet decomposition, thresholding, and reconstruction, making it easier to develop and test compression algorithms.

Wavelet-based audio compression in MATLAB is a powerful technique that balances compression efficiency and audio quality. By leveraging wavelets' ability to analyze signals in both time and frequency domains, significant data reduction can be achieved while preserving the perceptual integrity of the audio signal. This approach is widely used in applications such as music streaming, speech coding, and digital audio storage.





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#### **II. RELATED WORK**

- i. Audio compression is a crucial aspect of digital signal processing that aims to reduce file sizes while maintaining high audio quality. Traditional compression methods, such as Fourier Transform-based techniques (MP3, AAC), often suffer from poor time-frequency localization [2].
- ii. The Wavelet Transform has emerged as a powerful alternative, offering multi-resolution analysis and better handling of non-stationary signals like speech and music [3].
- iii. Audio Compression Using Wavelet Transform: Research papers have demonstrated the effectiveness of wavelet transform in audio compression, achieving compression ratios of up to 12:1 or higher [4].
- iv. Implementation of Audio Compression using Wavelet: MATLAB implementations of audio compression using wavelets have shown promising results, with good reconstruction quality and negligible error [5].
- v. Lossless Audio Compression using Wavelets: Researchers have also explored lossless audio compression using wavelets, achieving compression ratios of around 3:1.
- vi. Daubechies (1988) introduced a family of orthonormal wavelets that are widely used in signal processing. These wavelets provide compact support, making them ideal for compression applications [6].
- vii. Mallat (1989) developed the multi-resolution analysis framework, which serves as the foundation for waveletbased compression. This technique allows signals to be represented at different resolutions, making compression more efficient.
- viii. Misiti et al. (2000) introduced MATLAB's Wavelet Toolbox, which provides built-in functions like wavedec() and wrcoef() for wavelet decomposition and reconstruction.
- ix. Sharma & Singh (2018) analysed different wavelet families (Haar, Daubechies, Coiflet) in MATLAB and found that Daubechies wavelets provide the best balance between compression ratio and audio quality.



#### **III. METHODOLOGY**

Fig:3.1: Flow Chart

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#### 1. Input: Audio signal Segment 'x'

• This is the starting point. A small chunk of the audio signal that needs to be compressed is taken as input.

#### 2. Dynamic Dictionary

• This block represents a collection of previously encountered or learned audio signal segments (represented as 'y'). The dictionary is labeled as "dynamic," implying that its contents can change or adapt over time based on the input audio.

#### 3. Find closet dictionary entry:

- "y" compute d(x,y) For the input audio segment 'x', this block searches the "Dynamic Dictionary" to find the entry 'y' that is most similar to 'x'.
- It then computes a distance or dissimilarity measure, denoted as d(x,y), between the input segment 'x' and the closest dictionary entry 'y'. This distance quantifies how different the two signals are.

#### 4. Is d(x,y) < Threshold of perception?

• This is a decision block. The computed distance d(x,y) is compared against a predefined "Threshold of perception." This threshold represents the maximum amount of distortion that is considered perceptually insignificant to a human listener.

#### 5. If 'Yes' (d(x,y) < Threshold of perception):

- Transmit index of y. If the distance between the input segment 'x' and the closest dictionary entry 'y' is less than the perceptual threshold, it means that 'y' is a sufficiently good approximation of 'x' from a listener's perspective.
- In this case, instead of transmitting the entire segment 'x', only the index (or address) of the dictionary entry 'y' within the "Dynamic Dictionary" is transmitted. This achieves compression because the index is typically much shorter than the audio segment itself.

#### 6. If 'No' $(d(x,y) \ge$ Threshold of perception):

If the distance is greater than or equal to the perceptual threshold, it means that the closest dictionary entry 'y' is not a sufficiently accurate representation of 'x'. In this case, the process moves to the next steps:

#### Optimal wavelet encoding of 1. Signal ,2. Residual r=x-y

**Residual**  $\mathbf{r} = \mathbf{x} - \mathbf{y}$ : The difference between the original signal segment 'x' and the closest dictionary entry 'y' is calculated. This difference is called the "residual".

**Optimal wavelet encoding:** Both the original signal segment 'x' and the calculated residual 'r' are then encoded using an optimal wavelet transform technique. Wavelet encoding is a method that decomposes a signal into different frequency components, allowing for efficient representation by discarding less important components. The term "optimal" suggests that the wavelet encoding parameters are chosen to maximize compression while minimizing perceptual loss.

#### Transmit shorter of the two codes'

The system compares the size (in bits) of the encoded original signal 'x' and the encoded residual 'r' (along with any necessary information about the dictionary entry 'y' that was used to calculate the residual).

The shorter of these two encoded representations is then transmitted. This ensures that the most efficient coding method is used for the current segment.

#### 7. Output:

Compressed audio without noiseThe transmitted information (either the index of 'y' or the wavelet-encoded data) allows the decoder to reconstruct an approximation of the original audio signal segment 'x'. The phrase "without noise" implies that the compression process aims to minimize audible artifacts or distortions.





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#### 8. Feedback to Dynamic Dictionary

There is a feedback loop from the output of the "Transmit shorter of the two codes" block back to the "Dynamic Dictionary." This indicates that the dictionary is updated based on the processed audio segments. This update could involve adding new frequently occurring patterns, modifying existing entries, or removing less useful ones, allowing the dictionary to adapt to the characteristics of the audio signal over time and potentially improve compression efficiency.

#### Software Components:

- 1. MATLAB Environment
- 2. Wavelet Toolbox
- 3. Signal Processing Toolbox
- 4. Audio Toolbox
- 5. Graphical User Interface (GUI) Components
- 6. MATLAB Functions for Performance Metrics

#### **IV. RESULT**

#### **OUTPUT 1:**

Figure 4.1: Displays the frequency spectrum after compression, highlighting significant energy retention.

Figure 4.2: Shows the noisy signal before compression, indicating high signal variations.

Figure 4.3: Depicts the compressed audio signal after noise removal, preserving essential signal characteristics.

- Wavelet-based audio compression achieves high efficiency with minimal quality loss.
- Compression ratio of ~4:1 reduces storage requirements significantly.
- SNR remains high (~18 dB), ensuring good signal retention.
- MATLAB efficiently processes and evaluates compression outcomes.

#### Table 4.1. Metrics Comparison

Metric	Before Compression	After Compression
File Size (KB)	1000 KB	250 KB
Compression Ratio (CR)	-	4:1
SNR (dB)	20 dB	18 dB
MSE	-	0.0025
PSNR (dB)	-	32 dB





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Fig:4.1. Audio Signal Before Compression



Fig:4.2 Audio Signal after Compression

#### V. CONCLUSION

Wavelet-based audio compression in MATLAB proves to be a highly effective technique for reducing audio file sizes while maintaining good sound quality. By leveraging wavelet transforms, this method efficiently removes redundancy and irrelevant frequency components, leading to significant compression without introducing noticeable distortion. The advantages of wavelet-based compression include a high compression ratio, noise reduction, adaptability to different signal types, and efficient multi-resolution analysis. Compared to traditional Fourier-based methods, wavelets provide better performance in preserving transient and non-stationary components of the audio signal. From the results, we observe that wavelet compression achieves an optimal balance between file size reduction and signal fidelity, making it useful for applications such as speech compression, music storage, streaming, and medical signal processing. The choice of wavelet type and thresholding level significantly impacts performance, requiring fine-tuning based on the specific application needs.

In conclusion, wavelet-based audio compression in MATLAB offers a powerful and flexible solution for efficient audio data handling, with potential for real-world applications in communication, entertainment, and healthcare industries. Further research can explore adaptive thresholding techniques and deep learning integration for enhanced compression efficiency.

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