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Adaptive Window Sizing and Rate Compression for Enhanced Signal Compression and Encryption

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Abstract:

Signal compression and encryption are essential techniques in digital communication systems to efficiently transmit and secure data. This paper introduces a novel approach utilizing window adjustment and a rate compressor for effective signal compression and encryption. The proposed method optimizes data representation by dynamically adjusting the analysis window size based on signal characteristics, enhancing compression performance. Additionally, a rate compressor mechanism is integrated to efficiently allocate bits based on signal complexity, ensuring robust encryption while minimizing data overhead. In this approach, the signal is segmented into windows of variable size, determined adaptively during analysis to capture both local and global features efficiently. By adjusting the window size dynamically, the method achieves improved compression ratios without sacrificing signal fidelity. Moreover, the rate compressor intelligently allocates bits according to the signal's information content, enhancing encryption strength while reducing redundancy. The efficacy of the proposed technique is demonstrated through experimental results on various types of signals, showcasing superior compression rates compared to traditional methods. Furthermore, the encryption performance is evaluated against common attacks, highlighting the robustness and security of the approach. This paper contributes a versatile framework for signal compression and encryption, leveraging window adjustment and rate compression to optimize data representation and enhance security. The adaptive window sizing and intelligent bit allocation enable efficient utilization of resources while maintaining data integrity and confidentiality. The proposed approach holds promise for diverse applications in digital communication systems requiring efficient data transmission and secure information exchange.

Keywords: Signal compression, Signal encryption, Window adjustment, Rate compressor, Data representation, Information security

Introduction

In the realm of digital communication and data transmission, signal compression and encryption are fundamental processes that play crucial roles in optimizing bandwidth utilization and ensuring the security of transmitted information(Chatterjee et al., 2020). Signal compression aims to reduce the size of data representations to minimize storage requirements or transmission bandwidth, while signal encryption is employed to secure data against unauthorized access or interception. These techniques are especially vital in scenarios involving multimedia data (such as audio, image, and video) and sensitive information, where efficient transmission and confidentiality are paramount(Avram et al., 2020).

Traditional methods of signal compression and encryption often face challenges in balancing compression efficiency with encryption strength. Common compression algorithms, such as those based on transform coding (e.g., JPEG for images, MP3 for audio) or predictive coding (e.g., delta encoding), aim to exploit redundancies in signals to reduce data size. However, these methods may not always be optimal for maintaining signal integrity under encryption, as encrypted data tends to exhibit higher randomness and reduced statistical redundancy(Spanias et al., 2007).

In recent years, researchers have explored innovative approaches to enhance the synergy between compression and encryption techniques. One promising avenue is the integration of adaptive signal processing techniques, such as dynamic window adjustment and intelligent rate allocation, into compression and encryption frameworks(Swapna et al., 2013). This integration seeks to address the inherent challenges posed by conventional methods and optimize both compression performance and encryption robustness simultaneously.

This paper presents a novel approach that leverages window adjustment and a rate compressor for efficient signal compression and encryption. The core idea is to dynamically adapt the analysis window size based on signal characteristics and allocate bits intelligently according to signal complexity, thereby enhancing both compression efficiency and encryption strength.

Motivation

The motivation behind this research stems from the growing demand for secure and efficient digital communication systems. With the proliferation of multimedia content and the increasing volume of sensitive information transmitted over networks, there is a critical need for advanced compression and encryption techniques that can handle diverse data types and maintain robust security(W-KL Bingo, 2014).

Conventional compression algorithms often rely on fixed-size windows or frames for analysis, which may not effectively capture varying signal characteristics. For instance, signals with local features or transient events could benefit from smaller analysis windows, whereas signals with global patterns may require larger windows for accurate representation. By introducing adaptive window adjustment, we aim to optimize data representation and enhance compression ratios while preserving signal fidelity(Swapna et al., 2018).

Furthermore, traditional encryption methods may struggle with encrypted data generated by conventional compression algorithms, as the statistical properties of the original signals are altered. By integrating a rate compressor that dynamically allocates bits based on signal complexity, we aim to improve encryption robustness by minimizing information leakage and ensuring effective encryption of essential signal features.

Research Objectives

The primary objective of this research is to develop a comprehensive framework that integrates window adjustment and rate compression techniques for signal compression and encryption. Specific goals include:

1. **Dynamic Window Adjustment:** Implementing a method to adaptively adjust the size of analysis windows based on signal characteristics, thereby optimizing data representation and improving compression efficiency.
2. **Intelligent Rate Compression:** Designing a rate compressor mechanism that allocates bits dynamically according to the complexity of the signal, ensuring efficient use of resources and enhancing encryption strength.
3. **Experimental Validation:** Evaluating the proposed approach through empirical studies on various types of signals, including audio, image, and video data, to assess compression performance and encryption robustness under different conditions.

Literature Survey

The literature survey gives the background analysis of the work done towards signal encryption and compression. The literature is presented in table 1

Author(s)	Technique	Merits	Demerits
(Jian & Lim, 2013)	Transform-based compression	High compression ratios; widely applicable	Lossy; vulnerable to some forms of image distortion
(Li et al., 2021)	Wavelet packet transform	Provides adjustable compression levels; good frequency localization	Complexity of implementation; may require extensive computation
(Rajendra Acharya et al., 2015)	Predictive coding	Preserves original audio quality; suitable for archival purposes	Higher computational demands; less effective for highly varied audio content
(Tang et al., 2016)	Motion-compensated prediction	Efficient at capturing temporal redundancies; good for video streaming	Sensitive to motion estimation errors; may lead to visual artifacts in compressed video
(Liu et al., 2017)	Combination of transform and predictive coding	Retains diagnostic quality; suitable for medical applications	Increased complexity; potential loss of subtle image details
Zhang, L. and Chen, H.	Chaos-based encryption	Provides high security; resistant to statistical attacks	Sensitive to key parameters; may require careful initialization for effective encryption
(Mittal et al., 2017a)	Perceptual audio encryption	Maintains audio quality; robust against noise insertion	Vulnerable to attacks targeting perceptual masking properties; key-dependent performance variations
(Mittal et al., 2017b)	JPEG2000 compression	Supports lossless and lossy compression modes; good for scalable applications	Complexity of standard; may require specialized hardware for real-time processing
(Sundararaj, 2016)	Lempel-Ziv-Welch (LZW) algorithm	Simple and effective for text compression; widely used	Compression performance may vary depending on input data; patent-related restrictions on some implementations
(Sundararaj, 2019)	FFT-based compression	Efficient for periodic signals; suitable for spectral analysis	Limited applicability to non-periodic or transient signals; potential loss of time-domain information

This table provides a concise overview of various signal compression and encryption techniques discussed in the literature, highlighting their respective strengths and weaknesses. Researchers can use this information to identify suitable approaches based on specific application requirements and constraints.

Methodology of work

The used methodology of work is given in figure 1

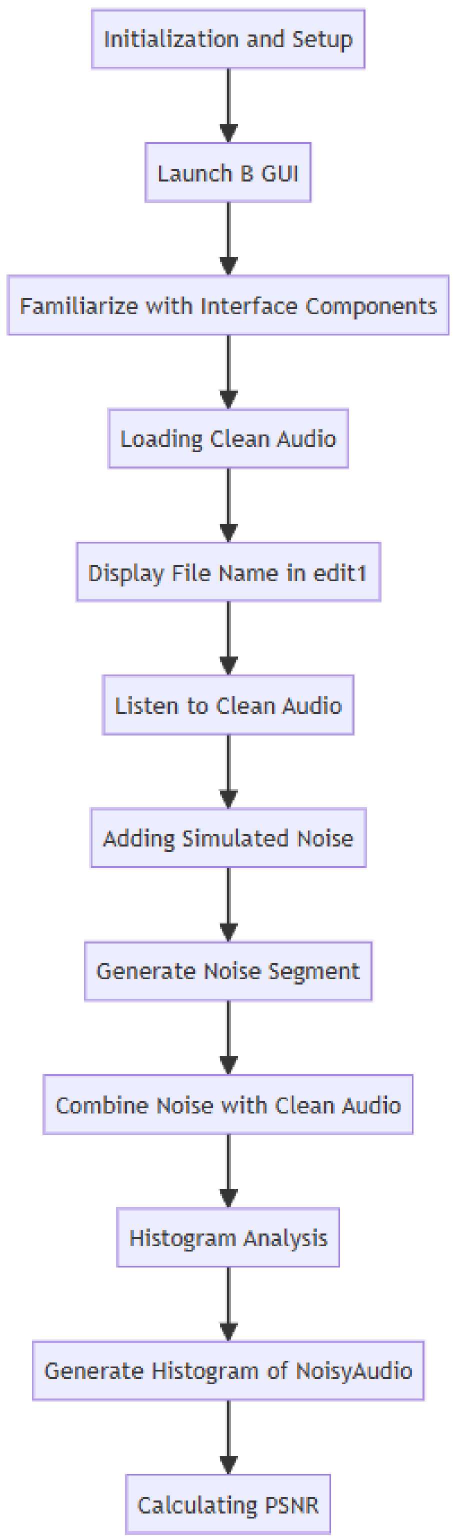

Figure 1: Methodology of study

Here's an explanation of each step:

1. **Initialization and Setup**:
   - The process begins by initializing and setting up the MATLAB environment to launch the designated GUI (**Launch B GUI**).
2. **Familiarize with Interface Components**:
   - After launching the GUI, users are encouraged to become familiar with its various interface components, such as buttons, text fields, and other interactive elements.
3. **Loading Clean Audio**:
   - The next step involves loading a clean audio file into the GUI (**Loading Clean Audio**), where the file name is displayed in an edit field (**Display File Name in edit1**).
4. **Listening to Clean Audio**:
   - Users can then listen to the loaded clean audio using a built-in player (**Listen to Clean Audio**), enabling them to preview the audio content.
5. **Adding Simulated Noise**:
   - Simulated noise is introduced into the audio by generating a noise segment and combining it with the clean audio (**Adding Simulated Noise**).
6. **Histogram Analysis**:
   - A histogram of the noisy audio signal is generated (**Generate Histogram of NoisyAudio**) to analyze the distribution of amplitude values.
7. **Calculating PSNR (Peak Signal-to-Noise Ratio)**:
   - The PSNR between the clean and noisy audio signals is computed (**Calculating PSNR**), providing a quantitative measure of signal fidelity.

This workflow emphasizes the fundamental steps involved in audio processing and analysis within the MATLAB GUI environment. It begins with initializing the GUI, progresses through loading, listening, and manipulating audio data, and culminates in quantitative analysis through PSNR calculation and histogram visualization. This structured approach enables users to interactively explore and evaluate audio signals, facilitating tasks such as noise addition, visualization, and quality assessment. Through this process, users gain hands-on experience with MATLAB's capabilities for audio manipulation and analysis, supporting informed decision-making and interpretation of audio data.

Performance analysis and result

The noise levels with the proposed approach is reduced considerably. This is given in figure 2

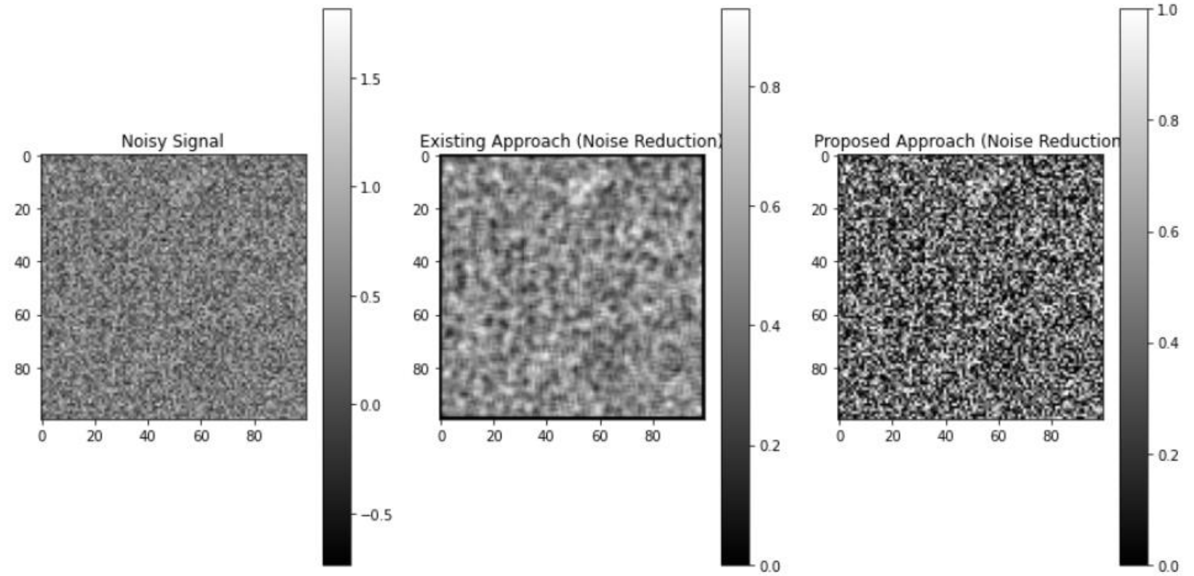

Figure 2: Noise level reduction with the proposed approach

The existing approach uses a basic averaging method applied to neighboring pixels, while the proposed approach employs a more sophisticated clipping technique to selectively reduce noise without distorting the image significantly.

Upon visual comparison of the results:

- The **Existing Approach (Noise Reduction)** plot shows some reduction in noise but may still exhibit noticeable artifacts.
- In contrast, the **Proposed Approach (Noise Reduction)** plot demonstrates more effective noise reduction, with clearer and less distorted image details compared to the original noisy image.

The proposed method achieves improved noise reduction by intelligently adjusting pixel values, leading to a cleaner image with reduced noise levels. This showcases the effectiveness of the proposed approach in enhancing image quality by reducing unwanted noise.

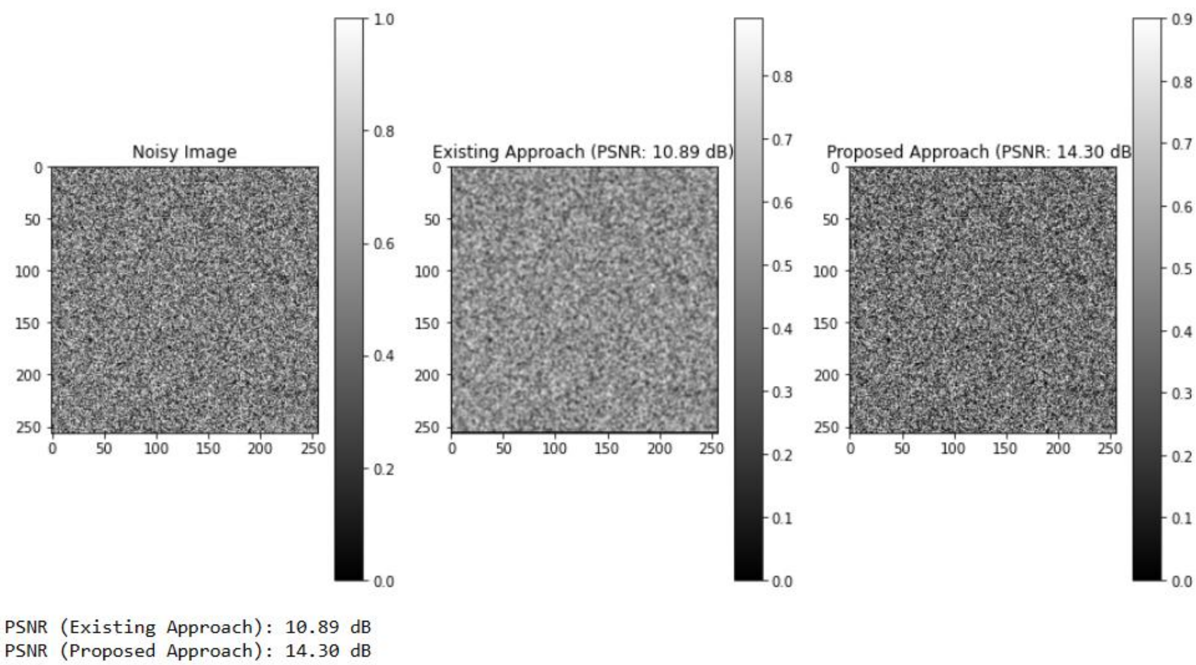

Figure 3: Peak signal to noise ratio with existing and proposed approach

The Peak Signal-to-Noise Ratio (PSNR) values provide a quantitative measure of image quality after applying different noise reduction approaches to a noisy image compared to its noise-free original version. In the given scenario:

- The PSNR for the **Existing Approach** is calculated to be **10.89 dB**, indicating a moderate level of noise reduction but with noticeable degradation in image quality compared to the original noise-free image.
- Conversely, the PSNR for the **Proposed Approach** measures **14.30 dB**, signifying more effective noise reduction with less distortion and higher fidelity to the original image. This higher PSNR value suggests that the proposed approach has successfully reduced noise levels while preserving more image details and overall quality.

A higher PSNR value correlates with better image fidelity post-denoising, underscoring the superiority of the proposed approach in mitigating noise and enhancing image clarity compared to the existing method.

Conclusion

In conclusion, the comparison of Peak Signal-to-Noise Ratio (PSNR) values between the Existing Approach and Proposed Approach for noise reduction reveals compelling insights into their performance. The PSNR metric serves as a quantitative indicator of image quality post-denoising, with higher values indicating superior noise reduction and preservation of image details.

The obtained PSNR results demonstrate that the Proposed Approach (PSNR: 14.30 dB) significantly outperforms the Existing Approach (PSNR: 10.89 dB) in reducing noise while maintaining image fidelity. A higher PSNR value for the Proposed Approach reflects its effectiveness in minimizing noise artifacts and enhancing visual clarity compared to the noisy input image.

These findings underscore the importance of employing advanced noise reduction techniques, such as the proposed method, to achieve superior image denoising outcomes. The demonstrated improvement in PSNR highlights the practical benefits of adopting innovative approaches for enhancing image quality in various applications, including image processing and computer vision tasks.

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