# Reduction of Impulsive Noise from Speech and Audio Signals by using Sd-Rom Algorithm

**G.Manmadha Rao, D.N Raidu Babu, P.S.L Krishna Kanth, B.Vinay, V.Nikhil**

***Abstract*: Removal of noise is the heart for speech and audio signal processing. Impulse noise is one of the most important noise which corrupts different parts in speech and audio signals. To remove this type of noise from speech and audio signals the technique proposed in this work is signal dependent rank order mean (SD-ROM) method in recursive version. This technique is used to replace the impulse noise samples based on the neighbouring samples. It detects the impulse noise samples based on the rank ordered differences with threshold values. This technique doesn't change the features and tonal quality of signal. Rank ordered differences is used for detecting the impulse noise samples in speech and audio signals. Once the sample is detected as corrupted sample, that sample is replaced with rank ordered mean value and this rank ordered mean value depends on the sliding window size and neighbouring samples. This technique shows good results in terms of signal to noise ratio (SNR) and peak signal to noise ratio (PSNR) when compared with other techniques. It mainly used for removal of impulse noises from speech and audio signals.***

***Keywords*: Impulse Noise, Sliding Window, Rank Ordered Differences, Rank Ordered Mean.**

## I. INTRODUCTION

Noise is an unwanted signal which causes interference to the required signal. It is of two types one is external and another is internal noise. Under external noises there are atmospheric, solar, industrial and cosmic noise. In atmospheric noise if the frequency of electromagnetic radiation is same as that of communication system frequency causes interference which damages the communication system. As external noises exist for only short duration of time. Hence, they are not included in the calculation of signal to noise ratio. So internal noises are considered in calculation of signal to noise ratio. Signal to noise ratio at the output of receiver must be as high as possible. Internal noise is the within the communication system and the most dominant is additive white gaussian noise, as internal noises are present for a long duration of time, they can be included in signal to noise ratio calculations.

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The present study proposes the technique in such a way that it should remove the samples which are modified during transmission but it should retain the samples which are not modified. The techniques like median filter and other order static filters modifies uncorrupted samples also. The main objective of the study is to modify only corrupted samples and the uncorrupted samples are left unchanged. The SD-ROM technique used in the present study is in recursive version which modifies only the samples which are corrupted. Each impulse noise is first detected in the sample stream and then is replaced with an estimate based on neighbouring samples. The main principle of this algorithm is to detect and replace the corrupted impulse noise samples with rank ordered mean value using the sliding window mechanism.

## II. LITERATURE SURVEY

An extensive survey made to find works related to removal of impulse noise from speech and audio signals. Charu Chandra et.al [1] developed an efficient method for impulse noise removal that is SD-ROM algorithm in non-recursive version which showed better results than other impulse noise removal techniques. Oudre .L et.al [2] studied automatic detection and removal of impulsive noise in audio signals, image processing. Arce G.R et.al [5] studied median filter theory and its applications which made to develop median filter algorithm and compared it with the present study focused on SD-ROM algorithm in recursive version which gives better results than non-recursive version of SD-ROM algorithm and other techniques like median filters, etc. (in terms of SNR and PSNR)

## III. THE SD-ROM ALGORITHM (RECURSIVE VERSION)

In speech and audio signal processing we use 1-D sliding window of odd size.

Consider a 1-D sliding window vector X of size 5 centered at n as shown in Figure 1.1. This sliding window vector X always carries a set of samples present in a sampled audio signal based on the window size selected. Here for window size of 5 the sliding window vector X always carries a set of 5 samples in sampled audio signal throughout the algorithm. Let W be a vector of size 4 carries 4 samples except the center sample X(n) in vector X.

This center sample X(n) is under inspection.

$$\mathbf{W} = [W_1, W_2, W_3, W_4] = [X(n-2), X(n-1), X(n+1), X(n+2)] \quad (1)$$


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| X(n-2) | X(n-1) | X(n) | X(n+1) | X(n+2) |
|---|---|---|---|---|

**Figure 1.1: Filter window**

1) Now the samples present in **W** are sorted in ascending order,

$$\mathbf{R} = [R_1, R_2, R_3, R_4] \qquad (2)$$

means the samples present in **R** are ordered by rank i.e., $R_1 \leq R_2 \leq R_3 \leq R_4$ that shows the samples are arranged in ascending order.

2) Next the rank ordered differences ($D_i$) are calculated

$$D_i = R_i - X(n) \quad \text{if } X(n) \leq \mu$$
$$D_i = X(n) - R_{4-i} \quad \text{if } X(n) > \mu \qquad (3)$$

Where $\mu = [R_2+R_3]/2$ and is called the rank ordered mean (ROM). For a window of size five, i = 1,2.

3) The algorithm decides whether the X(n) (center sample) is a noisy impulse or not if any of the following conditions hold

$$D_i > T_i \qquad i=1,2 \qquad (4)$$

Where $T_1$ and $T_2$ are two appropriately chosen threshold values. $T_1$ = 4, $T_2$ = 12. (these $T_1$, $T_2$ values work well for most of the inputs). Every detected impulse is replaced by the μ (ROM).

4) Here onwards the recursive process starts, in non-recursive approach the previously filtered samples are not included in sliding window whereas in recursive approach the previously filtered samples are taken into consideration means the samples that are already filtered by the SD-ROM algorithm are considered in sliding window while inspecting the center sample present in that sliding window.

| 1* | 2* | 3 | 4 | 5 | 6 |
|---|---|---|---|---|---|

**Figure 1.2: Recursive approach**

| 1 | 2 | 3 | 4 | 5 | 6 |
|---|---|---|---|---|---|

**Figure 1.3: Non-Recursive approach**

Figure 1.2 shows recursive approach where samples no. 1*,2* are previously filtered samples while sample no. 3 is under inspection. Figure 1.3 shows non-recursive approach where samples no. 1,2 are samples of audio signals while sample no. 3 is under inspection. Recursive approach gives more efficient results than non-recursive approach due to consideration of previous filtered samples.

5) Next the sliding window moves one step forward as shown in figure 1.4 and the entire algorithm continues until last sample of audio signal is filtered using SD-ROM algorithm.

| | | | | | |
|---|---|---|---|---|---|
| 1 | 2 | 3 | 4 | 5 | 6 |
| 1 | 2 | 3 | 4 | 5 | 6 |

**Figure 1.4: Movement of sliding window of size 5**

Note: The sliding window vector always move one step forward (for every one step forward one sample is left behind) for any odd sized window shown in figure 1.4.

## IV. SIMULATION RESULTS

To evaluate the efficiency of SD-ROM algorithm in recursive version, extensive experiments are implemented for detecting and restoring the corrupted audio signals. The results of this algorithm are compared to other methods in terms of SNR and PSNR.

As an initial implementation of this algorithm, an audio sample (sampled at 44.0 kHz) was artificially corrupted at 10% noise rate with random-valued impulse noise amplitudes of range [-20 to 15]. The thresholds $T_1$=4 and $T_2$=12 taken are suitable for most of the input audio signals.

Our trails with increase in size of the sliding window showed less efficiency and more input threshold values. Results for various sliding window sizes in recursive version shown in Table 1.

**Table 1: SNR and PSNR values for different window sizes**

| Sliding window size | SNR | PSNR |
|---|---|---|
| 5 | 32.2083 | 41.3815 |
| 7 | 29.0828 | 38.2625 |
| 9 | 27.3592 | 35.4843 |

As Table 1 shows that the sliding window size of 5 gives more efficient output. However, if increase in sliding window size needs more number of threshold values, for sliding window size of 7 the threshold values taken are $T_1$=6, $T_2$=8, $T_3$=14.

There are other techniques to remove impulse noises from the audio signals. In those techniques the well-known technique to remove impulse noises is median filter technique.

The results of the proposed SD-ROM algorithm in recursive version performance is compared with other techniques as shown in Table 2.

**Table 2: Performance comparison table at 10% noise rate**

| Filter type | Window size | SNR | PSNR |
|---|---|---|---|
| | | | |
| SD-ROM (recursive version) | 5 | 32.2083 | 41.3815 |

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| | | | |
|---|---|---|---|
| | 7 | 29.0828 | 38.2625 |
| | | | |
| SD-ROM | | | |
| (non-recursive version) | 5 | 30.0129 | 39.1714 |
| | | | |
| | 7 | 28.5542 | 37.8631 |
| | | | |
| | | | |
| General Median Filter | 5 | 27.4623 | 38.1994 |
| | | | |
| | 7 | 25.6667 | 36.9978 |
| | | | |

By this Table 2 the SD-ROM recursive version is more efficiently removing the impulse noises from the corrupted samples and replacing with rank ordered mean value.

The SD-ROM in recursive version is more efficient at higher noise rates.

Figure 2.1 shows the original sampled audio signal and this signal is corrupted at 10% noise rate as shown in figure 2.2, this signal is filtered using SD-ROM recursive version and the output is shown in figure 2.3 and the figure 2.4 shows the difference between original audio and SD-ROM signal.

Figure 3.1 shows the original sampled audio signal and this signal is corrupted at 10% noise rate as shown in figure 3.2, this signal is filtered using general MEDIAN FILTER and the output is shown in figure 3.3 and the figure 3.4 shows the difference between original audio and MEDIAN FILTER signal.

**SD-ROM algorithm in recursive version graphs:**

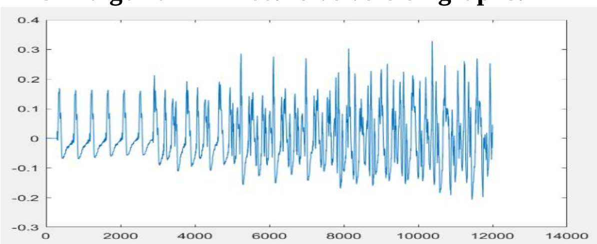

**Figure 2.1: Original audio signal**

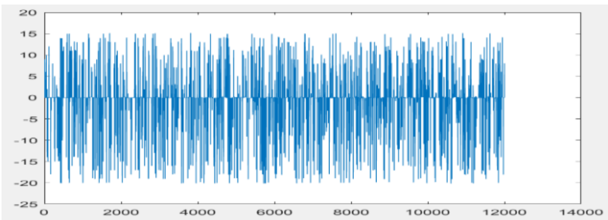

**Figure 2.2: Noisy audio signal (10% noise rate)**

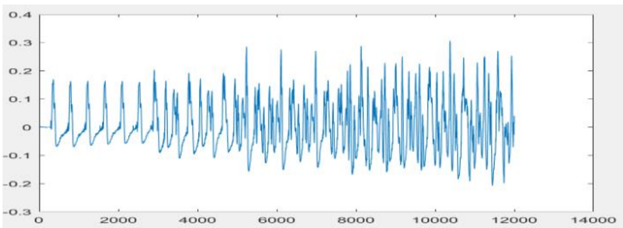

**Figure 2.3: SD-ROM recursive version**

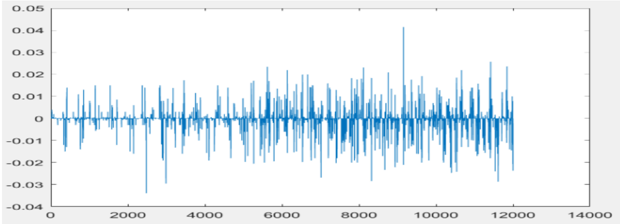

**Figure 2.4: Difference between original audio and SD-ROM signal**

**Median filter Graphs:**

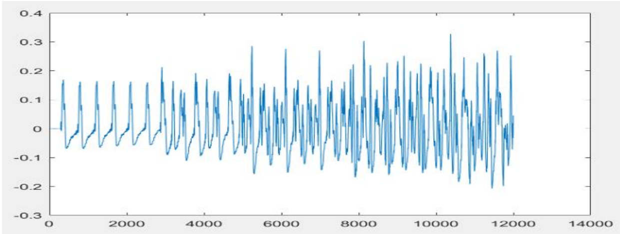

**Figure 3.1: Original audio signal**

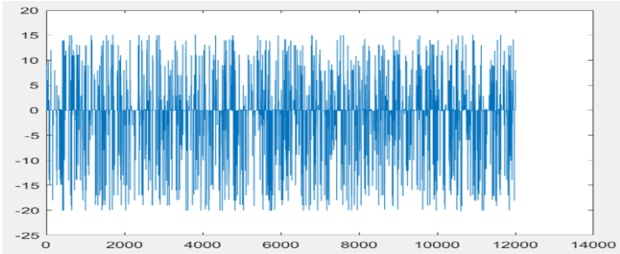

**Figure 3.2: Noisy audio signal (10% noise rate)**

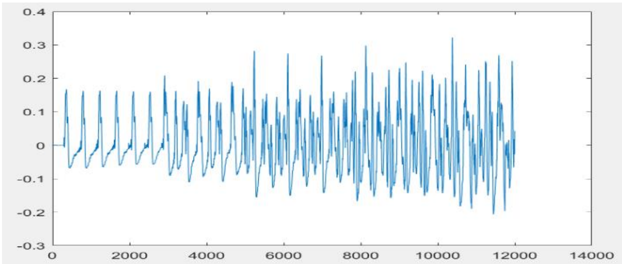

**Figure 3.3: MEDIAN FILTER**

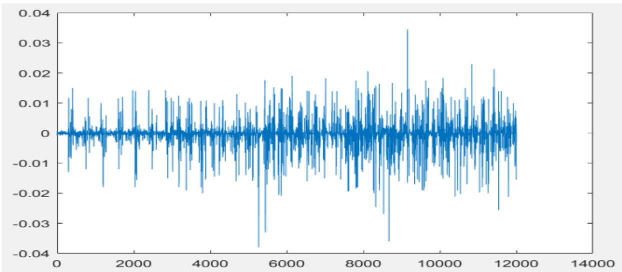

**Figure 3.4: Difference between original audio and MEDIAN FILTER signal**

## V. CONCLUSION

SD-ROM algorithm in recursive version is an efficient method in removing impulsive noise from audio signals.

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Which works efficiently without modifying uncorrupted samples and converting the impulse noised audio signal into de-noised audio signal based on rank ordered differences and threshold values which are used to test a sample whether corrupted or not. This technique gives good results in terms of SNR and PSNR. SD-ROM method has numerous applications like to restore old gramophone discs, in telecommunication systems, etc.

## REFERENCES


1. Charu Chandra, Michael S. Moore and Sanjit IS. Mitra, "An Efficient method for the removal of impulse noise from speech and audio signals" IEEE vol. 4, pp. 206-208, 1998.
2. L. Oudre, "Automatic Detection and Removal of Impulsive Noise in Audio Signals, Image Processing On line", pp. 267-281, 2015.
3. E. Abreu, M. Lightstone, S. Mitra, and K. Arakawa, "A new efficient approach for the removal of impulse noise from highly corrupted images," IEEE Trans. on Image Processing, vol. 5, no. 6, pp. 1012-1025, 1996.
4. A.S. Awad, H. Man, High performance detection filter for impulse noise removal in images, IEE Electr. Lett. 44 (3) (Jan. 2008) 192-194.
5. G.R. Arce, N.C. Gallagher, and T. Nodes, "Median filters: Theory and Applications," Advances in Computer Vision and Image Processing (T. Huang, ed.), Greenwich, CT: JAI Press, 1986.


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