REDUCTION OF IMPULSIVE NOISE FROM SPEECH AND AUDIO SIGNALS BY USING SD-ROM ALGORITHM

A Project report submitted in partial fulfilment of the requirements for the award of the degree of

BACHELOR OF TECHNOLOGY

IN

ELECTRONICS AND COMMUNICATION ENGINEERING

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ANITS

DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

ANIL NEERUKONDA INSTITUTE OF TECHNOLOGY AND SCIENCES

(UGC AUTONOMOUS)

(Permanently Affiliated to AU, Approved by AICTE and Accredited by NBA & NAAC with 'A' Grade)

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CERTIFICATE

This is to certify that the project report entitled "**Reduction of Impulsive Noise from Speech and Audio Signals by using SD-ROM Algorithm**" submitted by **D.N Raidu Babu (317126512073), P.S.L Krishna Kanth (317126512098), B. Vinay (317126512066), V. Sai Nikhil (317126512119)** in partial fulfilment of the requirements for the award of the degree of **Bachelor of Technology in Electronics & Communication Engineering** of Andhra University, Visakhapatnam is a record of Bonafide work carried out under my guidance and supervision.

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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.

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1)	dB	decibels
2)	no. of	number of
3)	PSNR	Peak signal to noise ratio
4)	SD-ROM	Signal Dependent Rank Ordered Mean
5)	SNR	Signal to noise ratio

CHAPTER 1

INTRODUCTION

1.1 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).

1.2 Overview:

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. 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.

1.3 Organization of the Project:

The project report is organised in seven chapters. It starts from the introduction. The current Chapter 1 introduces the project and gives a brief description of SD-ROM algorithm. Chapter 2 provides the brief idea about steps involved in speech and audio signal processing, its applications. Chapter 3 gives the detailed working of the SD-ROM algorithm in non-recursive and recursive version. Chapter 4 gives the detailed working of Median Filtering technique. Chapter 5 talks about the introduction to MATLAB. Chapter 6 gives the simulation results, SNR enhancement, SNR, PSNR and performance comparison with other techniques. Chapter 7 gives conclusion and future scope for advancements in speech and audio signal processing.

CHAPTER 2

SPEECH AND AUDIO SIGNAL PROCESSING

2.1 Introduction:

Audible sound arises from pressure variations in the air falling on the ear drum. The human auditory system is responsive to sounds in the frequency range of 20 Hz to 20 kHz as long as the intensity lies above the frequency dependent “threshold of hearing”. The audible intensity range is approximately 120 dB which represents the range between the rustle of leaves and boom of an aircraft take-off. Figure 1 displays the human auditory field in the frequency intensity plane. The sound captured by a microphone is a time waveform of the air pressure variation at the location of the microphone in the sound field. A digital audio signal is obtained by the suitable sampling and quantization of the electrical output of the microphone. Although any sampling frequency above 40 kHz would be adequate to capture the full range of audible frequencies, a widely used sampling rate is 44,100 Hz, which arose from the historical need to synchronize audio with video data. “CD quality” refers to 44.1 kHz sampled audio digitized to 16-bit word length. Sound signals can be very broadly categorized into environmental sounds, artificial sounds, speech and music. A large class of interesting sounds is time varying in nature with information coded in the form of temporal sequences of atomic sound events. For example, speech can be viewed as a sequence of phones, and music as the evolving pattern of notes. An atomic sound event, or a single gestalt, can be a complex acoustical signal described by a specific set of temporal and spectral properties. Examples of atomic sound events include short sounds such as a door slam, and longer uniform texture sounds such as the constant patter of rain. The temporal properties of an audio event refer to the duration of the sound and any amplitude modulations including the rise and fall of the waveform amplitude envelope. The spectral properties of the sound relate to its frequency components and their relative strengths. Audio waveforms can be periodic or aperiodic. Except for the simple sinusoid, periodic audio waveforms are complex tones comprising of a fundamental frequency and a series of overtones or multiples of the fundamental frequency. The relative amplitudes and phases of the frequency components influence the sound “colour” or timbre. Aperiodic waveforms, on the other hand, can be made up of non-harmonically related sine tones or frequency shaped noise. In general, a sound can exhibit both tone-like and noise-like spectral properties and these influence its perceived quality. Speech is characterized by alternations of tonal and noisy regions with tone durations corresponding to vowel segments occurring at a more regular syllabic rate. Music, on the other hand, being a melodic sequence of notes is highly tonal for the most part with both fundamental frequency and duration varying over a wide range. Sound signals are basically physical stimuli that are processed by the auditory system to evoke psychological sensations in the brain. It is appropriate that the salient acoustical properties of a sound be the ones that are important to the human perception and recognition of the sound. Hearing perception has been studied since 1870, the time of Helmholtz. Sounds are described in terms of the perceptual attributes of pitch, loudness, subjective duration and timbre. The human auditory system is known to carry out the frequency analysis of sounds to feed the higher-level cognitive functions. Each of the subjective sensations is correlated with more than one spectral property (e.g., tonal content) or temporal property (e.g., attack of a note struck on an instrument) of the sound. Since both spectral and temporal properties are relevant to the perception and cognition of sound, it is only appropriate to consider the representation of audio signals in terms of a joint description in time and frequency. While audio signals are non-stationary by nature, audio signal analysis usually assumes that the signal properties change relatively slowly with time. Signal parameters, or features, are estimated from the analysis of short windowed segments of the signal, and the analysis is repeated at uniformly spaced intervals of time. The parameters so estimated generally represent the signal characteristics corresponding to the time center of the windowed segment. This method of estimating the parameters of a time-varying signal is known as "short-time analysis" and the parameters so obtained are referred to as the "short-time" parameters. Signal parameters may relate to an underlying signal model. Speech signals, for example, are approximated by the well-known source-filter model of speech production. The source-filter model is also applicable to the sound production mechanism of certain musical instruments where the source refers to a vibrating object, such as a string, and the filter to the resonating body of the instrument. Music due to its wide definition, however, is more generally modelled based on observed signal characteristics as the sum of elementary components such as continuous sinusoidal tracks, transients and noise

- **Analog signals:**

  An analog audio signal is a continuous signal represented by an electrical voltage or current that is "analogous" to the sound waves in the air. Analog signal processing then involves physically altering the continuous signal by changing the voltage or current or charge via electrical circuits. Historically, before the advent of widespread digital technology, analog was the only method by which to manipulate a signal. Since that time, as computers and software have become more capable and affordable, digital signal processing has become the method of choice. However, in music applications, analog technology is often still desirable as it often produces nonlinear responses that are difficult to replicate with digital filters.

- **Digital signals:**

  A digital representation expresses the audio waveform as a sequence of symbols, usually binary numbers. This permits signal processing using digital circuits such as digital signal processors, microprocessors and general-purpose computers. Most modern audio systems use a digital approach as the techniques of digital signal processing are much more powerful and efficient than analog domain signal processing.

- **Application areas:**

  Processing methods and application areas include storage, data compression, music information retrieval, speech processing, localization, acoustic detection, transmission, noise cancellation, acoustic fingerprinting, sound recognition, synthesis, and enhancement (e.g. equalization, filtering, level compression, echo and reverb removal or addition, etc.).

1) **Audio Broadcasting:**

   Audio signal processing is used when broadcasting audio signals in order to enhance their fidelity or optimize for bandwidth or latency. In this domain, the most important audio processing takes place just before the transmitter. The audio processor here must prevent or minimize overmodulation, compensate for non-linear transmitters (a potential issue with medium wave and shortwave broadcasting), and adjust overall loudness to desired level.

2) **Active noise control:**

   Active noise control is a technique designed to reduce unwanted sound. By creating a signal that is identical to the unwanted noise but with the opposite polarity, the two signals cancel out due to destructive interference.

3) **Audio synthesis:**

   Audio synthesis is the electronic generation of audio signals. A musical instrument that accomplishes this is called a synthesizer. Synthesizers can either imitate sounds or generate new ones. Audio synthesis is also used to generate human speech using speech synthesis.

4) **Audio effects:**

   Audio effects are systems designed to alter how an audio signal sounds. Unprocessed audio is metaphorically referred to as dry, while processed audio is referred to as wet.

   - **delay -** or echo - To simulate the effect of reverberation in a large hall or cavern, one or several delayed signals are added to the original signal. To be perceived as echo, the delay has to be of order 35 milliseconds or above. Short of actually playing a sound in the desired environment, the effect of echo can be implemented using either digital or analog methods. Analog echo effects are implemented using tape delays or bucket-brigade devices. When large numbers of delayed signals are mixed a reverberation effect is produced; The resulting sound has the effect of being presented in a large room.
   - **flanger** - to create an unusual sound, a delayed signal is added to the original signal with a continuously variable delay (usually smaller than 10 ms). This effect is now done electronically using DSP, but originally the effect was created by playing the same recording on two synchronized tape players, and then mixing the signals together. As long as the machines were synchronized, the mix would sound more-or-less normal, but if the operator placed their finger on the flange of one of the players (hence "flanger"), that machine would slow down and its signal would fall out-of-phase with its partner, producing a phasing comb filter effect. Once the operator took his finger off, the player would

speed up until it was back in phase with the master, and as this happened, the phasing effect would appear to slide up the frequency spectrum. This phasing up-and-down the register can be performed rhythmically.

- **phaser** - another way of creating an unusual sound; the signal is split, a portion is filtered with a variable all-pass filter to produce a phase-shift, and then the unfiltered and filtered signals are mixed to produce a comb filter. The phaser effect was originally a simpler implementation of the flanger effect since delays were difficult to implement with analog equipment.
- **chorus** - a delayed version of the signal is added to the original signal. The delay has to be short in order not to be perceived as echo, but above 5 ms to be audible. If the delay is too short, it will destructively interfere with the un-delayed signal and create a flanging effect. Often, the delayed signals will be slightly pitch shifted to more realistically convey the effect of multiple voices.
- **equalization** - frequency response is adjusted using audio filter(s) to produce desired spectral characteristics. Frequency ranges can be emphasized or attenuated using low-pass, high-pass, band-pass or band-stop filters. Moderate use of equalization can be used to fine-tune the tonal quality of a recording; extreme use of equalization, such as heavily cutting a certain frequency can create more unusual effects. Band-pass filtering of voice can simulate the effect of a telephone because telephones use band-pass filters.
- **overdrive effects** - can be used to produce distorted sounds, and increase loudness. The most basic overdrive effect involves clipping the signal when its absolute value exceeds a certain threshold.
- t**imescale-pitch modification** - this effect shifts a signal up or down in pitch. For example, a signal may be shifted an octave up or down. Blending the original signal with shifted duplicate(s) can create harmonization. Another application of pitch shifting is pitch correction where a musical signal is adjusted to improve intonation. The complement of pitch shift is timescale modification, that is, the process of changing the speed of an audio signal without affecting its pitch.
- **resonators** - emphasize harmonic frequency content on specified frequencies. These may be created from parametric equation or from delay-based comb-filters.
- **robotic voice effects** - are used to make an actor's voice sound like a synthesized human voice.
- **ring-modulation** - is an effect made famous by Doctor Who's Daleks and commonly used throughout sci-fi.
- **dynamic range compression** - the control of the dynamic range of a sound to avoid unintentional or undesirable fluctuation in level. Dynamic range compression is not to be confused with audio data compression, where the amount of data is reduced without affecting the amplitude of the sound it represents.

- **3D audio effects** - placement of sounds outside the spatial range available through stereo or surround imaging.
- **wave field synthesis** - a spatial audio rendering technique for the creation of virtual acoustic environments.
- **De-esser** - control of sibilance in speech and singing.

2.2 Modulation and Demodulation:

- **Modulation:**

  In electronics and telecommunications, **modulation** is the process of varying one or more properties of a periodic waveform, called the carrier signal, with a separate signal called the modulation signal that typically contains information to be transmitted. For example, the modulation signal might be an audio signal representing sound from a microphone, a video signal representing moving images from a video camera, or a digital signal representing a sequence of binary digits, a bitstream from a computer. The carrier is higher in frequency than the modulation signal. The purpose of modulation is to impress the information on the carrier wave, which is used to carry the information to another location. In radio communication the modulated carrier is transmitted through space as a radio wave to a radio receiver. Another purpose is to transmit multiple channels of information through a single communication medium, using frequency division multiplexing (FDM). For example in cable television which uses FDM, many carrier signals carrying different television channels are transported through a single cable to customers. Since each carrier occupies a different frequency, the channels do not interfere with each other. At the destination end, the carrier signal is demodulated to extract the information bearing modulation signal. In **analog modulation** an analog modulation signal is impressed on the carrier. Examples are amplitude modulation (AM) in which the amplitude (strength) of the carrier wave is varied by the modulation signal, and frequency modulation (FM) in which the frequency of the carrier wave is varied by the modulation signal. These were the earliest types of modulation, and are used to transmit an audio signal representing sound, in AM and FM radio broadcasting. More recent systems use **digital modulation**, which impresses a digital signal consisting of a sequence of binary digits (bits), a bitstream, on the carrier. In frequency shift keying (FSK) modulation, used in computer buses and telemetry, the carrier signal is periodically shifted between two frequencies that represent the two binary digits. In digital baseband modulation (line coding) used to transmit data in serial computer bus cables and wired LAN computer networks such as Ethernet, the voltage on the line is switched between two amplitudes (voltage levels) representing the two binary digits, 0 and 1, and the carrier (clock) frequency is combined with the data. A more complicated digital modulation method that employs multiple carriers, orthogonal frequency division multiplexing (OFDM), is used in Wi-Fi networks, digital radio stations and digital cable television transmission. In music production, the term modulation has a different meaning: it is the process of gradually changing sound properties in order to reproduce a sense of movement and depth in audio recordings. It involves the use of a source signal (known as a modulator) to control another signal (a carrier) through a variety of sound effects and methods of synthesis. With singers and other vocalists, modulation means to modify characteristics of their voices during a performance, such as loudness or pitch.

- **Demodulation:**

  Demodulation was first used in radio receivers. In the wireless telegraphy radio systems used during the first 3 decades of radio (1884-1914) the transmitter did not communicate audio (sound) but transmitted information in the form of pulses of radio waves that represented text messages in Morse code. Therefore, the receiver merely had to detect the presence or absence of the radio signal, and produce a click sound. The device that did this was called a detector. The first detectors were coherers, simple devices that acted as a switch. The term detector stuck, was used for other types of demodulators and continues to be used to the present day for a demodulator in a radio receiver. The first type of modulation used to transmit sound over radio waves was amplitude modulation (AM), invented by Reginald Fessenden around 1900. An AM radio signal can be demodulated by rectifying it to remove one side of the carrier, and then filtering to remove the radio-frequency component, leaving only the modulating audio component. This is equivalent to peak detection with a suitably long time constant. The amplitude of the recovered audio frequency varies with the modulating audio signal, so it can drive an earphone or an audio amplifier. Fessendon invented the first AM demodulator in 1904 called the electrolytic detector, consisting of a short needle dipping into a cup of dilute acid. The same year John Ambrose Fleming invented the Fleming valve or thermionic diode which could also rectify an AM signal. There are several ways of demodulation depending on how parameters of the base-band signal such as amplitude, frequency or phase are transmitted in the carrier signal. For example, for a signal modulated with a linear modulation like AM (amplitude modulation), we can use a synchronous detector. On the other hand, for a signal modulated with an angular modulation, we must use an FM (frequency modulation) demodulator or a PM (phase modulation) demodulator. Different kinds of circuits perform these functions. Many techniques such as carrier recovery, clock recovery, bit slip, frame synchronization, rake receiver, pulse compression, Received Signal Strength Indication, error detection and correction, etc., are only performed by demodulators, although any specific demodulator may perform only some or none of these techniques. Many things can act as a demodulator, if they pass the radio waves on nonlinearly.

2.3 PCM Modulation and Demodulation:

We know that modulation can be defined as the process of changing the carrier signal's parameters by the instant values of the message signal. The transmission of message signal can be done mainly for communication & the high-frequency signal like a carrier signal doesn't include data, however, it is used for lengthy-distance communication.

The classification of modulation techniques can be done based on the type of modulation used. For instance, the digital modulation uses PCM or Pulse Code Modulation technique. In PCM, the message signal can be signified through a series of coded pulses. So, this message signal can be attained through signifying the signal in the form of discrete in both times as well as amplitude. This article discusses an overview of pulse code modulation and demodulation.

Pulse code modulation is a method that is used to convert an analog signal into a digital signal so that a modified analog signal can be transmitted through the digital communication network. PCM is in binary form, so there will be only two possible states high and low (0 and 1). We can also get back our analog signal by demodulation. The Pulse Code Modulation process is done in three steps Sampling, Quantization, and Coding. There are two specific types of pulse code modulations such as differential pulse code modulation (DPCM) and adaptive differential pulse code modulation (ADPCM).

- **Pulse Code Modulation Block Diagram:**

  The basic elements of PCM mainly include the transmitter section and receiver section. The pulse code modulation steps are discussed below

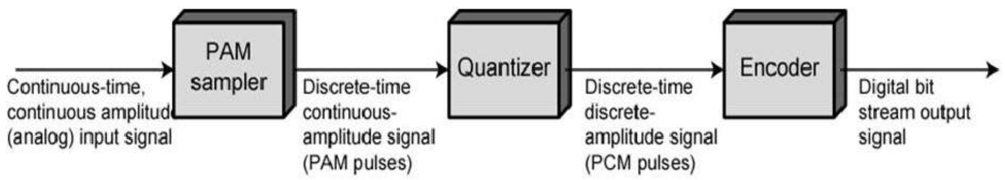

Fig 2.1: Block Diagram of Pulse Code Modulation

Here is a block diagram of the steps which are included in PCM. In sampling, we are using a PAM sampler that is Pulse Amplitude Modulation Sampler which converts continuous amplitude signal into Discrete-time- continuous signal (PAM pulses). The basic block diagram of PCM is given below for better understanding.

To get a pulse code modulated waveform from an analog waveform at the transmitter end (source) of a communications circuit, the amplitude of the analog signal samples at regular time intervals. The sampling rate or the number of samples per second is several times the maximum frequency. The message signal converted into the binary form will be usually in the number of levels which is always to a power of 2. This process is called quantization.

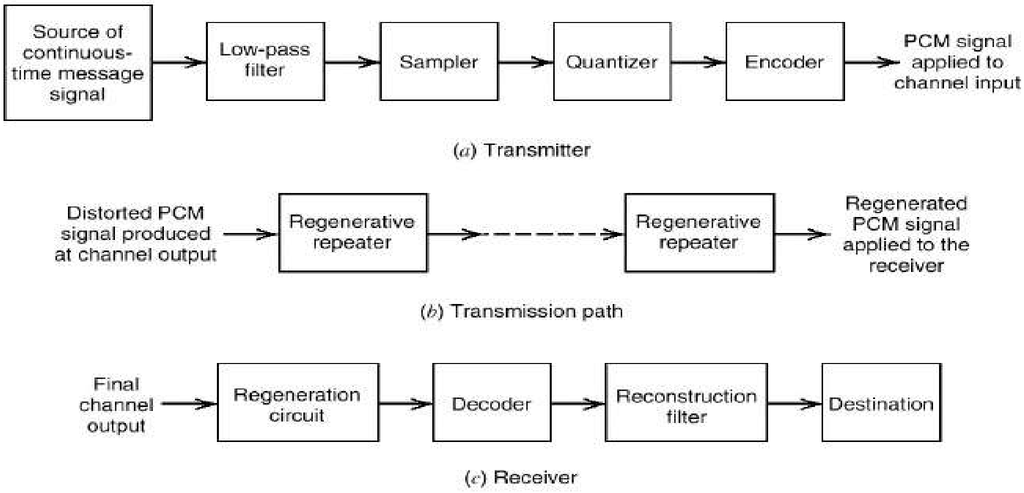

Fig 2.2: Basic Elements of Pulse Code Modulation System

At the receiver end, a pulse code demodulator decodes the binary signal back into pulses with the same quantum levels as those in the modulator. By further processes, we can restore the original analog waveform.

- **Pulse Code Modulation Theory:**

The above block diagram describes the whole process of PCM. The source of the continuous-time message signal is passed through a low pass filter and then sampling, Quantization, Encoding will be done. We will see each in detail step by step.

1) Sampling:

Sampling is a process of measuring the amplitude of a continuous-time signal at discrete instants, converts the continuous signal into a discrete signal. For example, conversion of a sound wave to a sequence of samples. The Sample is a value or set of values at a point in time or it can be spaced. Sampler extract samples of a continuous signal, it is a subsystem ideal sampler produces samples that are equivalent to the instantaneous value of the continuous signal at the specified various points. The Sampling process generates a flat-top Pulse Amplitude Modulated (PAM) signal.

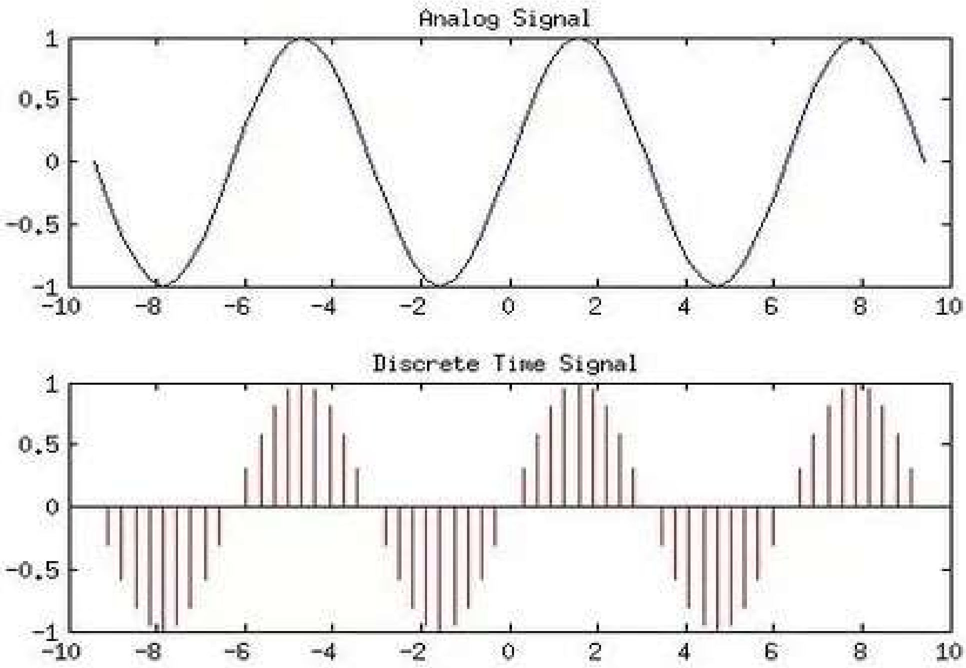

Fig 2.3: Analog and Sampled Signal

Sampling frequency, Fs is the number of average samples per second also known as the Sampling rate. According to the Nyquist Theorem, the sampling rate should be at least 2 times the upper cut-off frequency. Sampling frequency, Fs>=2*fmax to avoid Aliasing Effect. If the sampling frequency is very higher than the Nyquist rate it becomes Oversampling, theoretically a bandwidth-limited signal can be reconstructed if sampled above the Nyquist rate. If the sampling frequency is less than the Nyquist rate it will become Under sampling.

Basically, two types of techniques are used for the sampling process. Those are 1. Natural Sampling and 2. Flat- top Sampling.

2) Quantization:

In quantization, an analog sample with an amplitude that converted into a digital sample with an amplitude that takes one of a specifically defined set of quantization values. Quantization is done by dividing the range of possible values of the analog samples into some different levels and assigning the centre value of each level to any sample in the quantization interval. Quantization approximates the analog sample values with the nearest quantization values.

So almost all the quantized samples will differ from the original samples by a small amount. That amount is called quantization error. The result of this quantization error is we will hear a hissing noise when playing a random signal. Converting analog samples into binary numbers that are 0 and 1.

In most cases, we will use uniform quantizers. Uniform quantization is applicable when the sample values are in a finite range (Fmin, Fmax). The total data range is divided into 2n levels, let it be L intervals. They will have an equal length Q. Q is known as Quantization interval or quantization step size. In uniform quantization, there will be no quantization error.

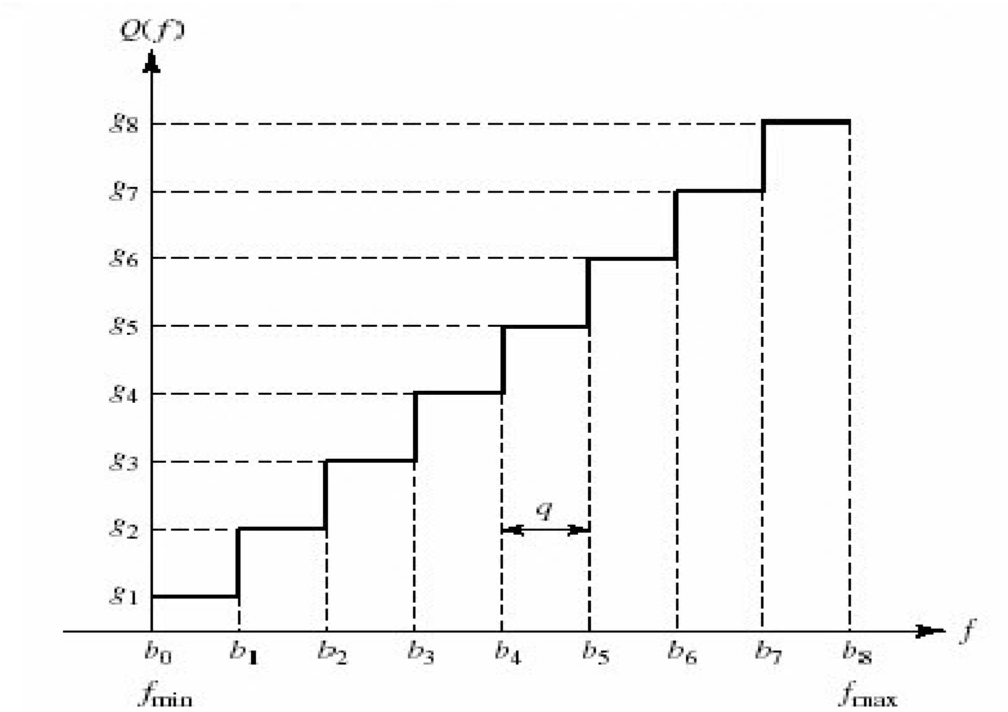

Fig 2.4: Uniformly Quantized Signal

L=2n, then Step size Q = (Fmax – Fmin) / L

Interval i is mapped to the middle value. We will store or send only the index value of quantized value.

An Index value of quantized value Qi (F) = [F – Fmin / Q]

Quantized value Q (F) = Qi (F) Q + Q / 2 + Fmin

But there are some problems raised in uniform quantization those are

- Only optimal for the uniformly distributed signal.

- Real audio signals are more concentrated near zeros.
- The Human ear is more sensitive to quantization errors at small values.

The solution to this problem is using non-uniform quantization. In this process, the quantization interval is smaller near zero.

3) Low Pass Filter (LPF):

This LPF is used to remove the high frequency (HF) components that are present within the input analog signal. Here this signal is higher as compared to the highest frequency message signal so that it avoids aliasing of the message signal.

4) Regenerative Repeater:

The signal strength can be enhanced through this regenerative repeater. So, the channel's output also includes a regenerative repeater circuit to balance the signal loss, renovate the signal & also increases the signal strength.

5) Decoder:

The main function of a decoder circuit is to decode the pulse-coded signal to repeat the actual signal. This circuit works like a demodulator.

6) Reconstruction Filter:

After the conversion of DAC (digital-to-analog conversion) is done with the help of the decoder and regenerative circuit, then an LPF (low-pass filter) is used to get back the original signal. So, this is known as the reconstruction filter.

Therefore, the Pulse Code Modulator circuit (PCM) is used to digitize the specified analog signal, code it, sample it & after that, it transmits in the form of analog. So, this entire procedure can be repeated within a reverse model to get the actual signal.

7) Coding:

The encoder encodes the quantized samples. Each quantized sample is encoded into an 8-bit codeword by using A-law in the encoding process.

- Bit 1 is the most significant bit (MSB), it represents the polarity of the sample. "1" represents positive polarity and "0" represents negative polarity.
- Bit 2,3 and 4 will defines the location of the sample value. These three bits together form a linear curve for low-level negative or positive samples.
- Bit 5,6,7 and 8 are the least significant bits (LSB) it represents one of the segments' quantized value. Each segment is divided into 16 quantum levels.

Pulse code modulation is similar to PWM, PAM otherwise PPM however there is a significant disparity among them that is they are analog pulse modulation systems but Pulse code modulation is a digital pulse modulation system. So, the output of PCM is in the form of coded digital and it is in the form of digital signals of stable width, position & amplitude. The data can be transmitted in code words format. A pulse code modulation system includes a transmitter like a PCM encoder & a receiver like a PCM decoder. The important operations within the transmitter of pulse code modulation mainly include sampling, quantizing, and encoding. Generally, these operations are performed within a similar circuit namely ADC.

- **Types of PCM:**

PCM is two types of Differential Pulse Code Modulation (DPCM), Adaptive Differential Pulse Code Modulation (ADPCM) & Linear Pulse Code Modulation.

1) Differential Pulse Code Modulation (DPCM):

In DPCM only the difference between a sample and the previous value is encoded. The difference will be much smaller than the total sample value so we need some bits for getting the same accuracy as in ordinary PCM. So that the required bit rate will also reduce. For example, in 5-bit code 1 bit is for polarity, and the remaining 4 bits for 16 quantum levels.

The **advantages of differential pulse code modulation** include the following.

- The requirement of bandwidth is low as compared to pulse code modulation.
- Due to the prediction filter, the quantization error can be decreased.
- As compared to PCM, the number of bits that are used to represent one sample value can also be reduced.
- Low signalling rate.

The **disadvantages of differential pulse code modulation** include the following.

- Bit rate is high.
- Practical usage is restricted.
- A Predicator circuit needs to be used which is extremely complex.

2) Adaptive Differential Pulse Code Modulation (ADPCM):

ADPCM is achieved by adapting the quantizing levels to analog signal characteristics. We can estimate the values with the preceding sample values. Error estimation is done as same as in DPCM. In the 32Kbps ADPCM method difference between the predicted value and sample, the value is coded with 4 bits, so that we'll get 15 quantum levels. In this method data rate is half of the conventional PCM.

3) Linear Pulse Code Modulation:

The term LPCM stands for "Linear pulse code modulation". This is one kind of modulation technique, used to encode uncompressed audio data digitally, wherever audio signals are signified through a series of amplitude values from a model on a linear scale where these values are comparative to the amplitudes. So, the amplitude values are quantized linearly, therefore similar to a very large set of feasible values through a quite small set of values that may be discrete symbols or integers.

This kind of modulation is also used for audio formats like a collective reference that occurs when using the result of this encoding technique. PCM or Pulse code modulation is a general method of encoding and the main function of this is to describe LPCM frequently and it is capable of extremely high throughput.

- **Pulse Code Demodulation:**

Pulse Code Demodulation will be doing the same modulation process in reverse. Demodulation starts with the decoding process, during transmission the PCM signal will be affected by noise interference. So, before the PCM signal sends to the PCM demodulator, we have to recover the signal to the original level for that we are using a comparator. The PCM signal is a series pulse wave signal, but for demodulation, we need a wave to be parallel. By using a serial to parallel converter, the series pulse wave signal will be converted into a parallel digital signal. After that the signal will pass through the n-bits decoder, it should be a Digital to Analog converter. Decoder recovers the original quantization values of the digital signal. This quantization value also includes a lot of high-frequency harmonics with original audio signals. For avoiding unnecessary signals, we utilize a low-pass filter at the final part.

- **Advantages:**

The advantages of pulse code modulation include the following.

- Analog signals can be transmitted over a high-speed digital communication system.
- The probability of occurring error will reduce by the use of appropriate coding methods.
- PCM is used in Telkom system, digital audio recording, digitized video special effects, digital video, voice mail.
- PCM is also used in Radio control units as transmitters and also a receiver for remote-controlled cars, boats, planes.
- The PCM signal is more resistant to interference than normal signals.

- **Pulse Code Modulation Applications:**

  The applications of PCM include the following.

  - PCM technique is mainly used to change the signal from analog to digital signal so that an analog signal which is changed can be broadcasted throughout the digital communication network. This modulation is available in binary form, so the available possible states will be two types like high & low.
  - Pulse-code modulation (PCM) is a technique used to represent sampled analog signals digitally. It is the normal form of digital audio within computers, digital telephony, compact discs & other digital audio applications.
  - These modulations can be used for temperature regulation, cold or heat storage through high storage density & thermal comfort within buildings that need a narrow range of temperature. Thus, if the solar energy is stored efficiently, then it can be used for night cold. The pulse code modulation refers to the utilization of a precise set of rules for changing a signal into a stream of digits.

- **Limitations of PCM**

  - The sampling theorem like Nyquist–Shannon illustrates the operating of pulse code modulation devices can be done without establishing distortions in their frequency bands if these bands offer a sampling frequency as a minimum twice that of the maximum frequency included within the i/p signal.

  - For instance, the voiceband frequency which is used mainly ranges from 300 Hz -3400 Hz. For the efficient renovation of the voice signal, the applications of telephony normally utilize a sampling frequency of 8000 Hz which is twice the maximum working voice frequency.

  - Apart from in any PCM system, there are impairment implicit possible sources like the following. Selecting a separate value that is close but not precisely at the analog signal range for every sample guide to quantization error.

  - In between the samples, no signal measurement can be made; so, the sampling theorem assurances non-ambiguous depiction & signal recovery simply if it has no energy at 'fs/2' frequency, high frequencies will not be properly signified otherwise recovered & include aliasing distortion toward the signal under the Nyquist frequency.

  - Because samples are reliant on time, so a precise clock is necessary for precise reproduction. If any of the encodings otherwise decoding CLK is not steady, these defects will directly influence the output of the device quantity.

2.4 Types of noises and noise removal techniques:

Virtually all audio recordings contain some amount of noise. This noise may join audio signal during recording process or due to long media storage, which is not acceptable by sound engineers. To produce best quality audio recordings these unwanted audio noises must be removed to the greatest extent possible.

- **Importance of Digitization in Noise Removal:**

  - Sound inherently begins and ends as an analogue signal. Few years ago, removing noise from the audio was a very difficult task because of the high cost involved which was not affordable for a typical user.
  - The sound digitization and digital signal processing technologies changed this scenario dramatically. The new generation high-speed PCs equipped with quality sound cards and software applications has made audio noise removal work even more affordable.
  - Any vinyl surface is subject to micro fissures, scratches, and soiling. Over a period of time, the result is a constant deterioration in the quality of the audio. This deterioration shows itself in unwanted noise, clicks, and crackles.
  - Digitization process itself does not remove any type of the distortion or noise, but it allows us to suppress or eliminate such distortions or noise by applying special soft Noise Removal tools/algorithms to a digital sound.

- **Classification of Noise**

  - **Broadband Noise:** The noise in which the acoustic energy is distributed over a relatively wide range of frequencies called broad band noise or continuous noise. The sounds such as hiss and static fall in this category. It is measured in Signal to Noise Ratio (SNR). The average level of noise is called noise floor
  - **Narrow Band Noise:** It is limited to a narrow range of frequencies. Usually, this kind of noise has a constant level and frequency and normally caused by incorrect grounding and poorly shielded cables. It includes single frequency such a 50Hz or 60 Hz induced from the mains supply and its harmonics at 100 Hz, 150 Hz and so on.
  - **Impulse Noise:** It includes sharp sounds such as clicks and pops. The pops are high amplitude audio impulses compared to the original audio signal and persist for a longer duration that is usually more than 2ms whereas clicks are lower in amplitude, higher in frequency and shorter in duration.
  - Digital clicks also exist which are caused by processor overload as we make a digital recording. The recording "stops" for a moment and the resulting skip creates a very short click.
  - In audio editing software we can see Clicks and Pops as below:
    Irregular Noise: it includes sounds such as background conversation, traffic and rain. These types of sounds are very difficult to remove because they are made up of many random sounds that vary in frequency and loudness.

- **Noise removal strategies:**

   - Minimize Noise Before Recording Before digitizing a LP record, we should clean the record with a soft micro fiber brush and mild cleaning solution to remove dust and then it may be vacuum dried. The stylus and cartridge of Turntable should also be in good condition. While recording from a tape deck we should ensure that heads are clean and demagnetized. Good quality shielded cables can further reduce noise from electrical interference.
   - While transferring a LP or a tape, a few seconds of silence may be recorded from the source which will show the amount of existing hum, hiss & static noise as well as it will help to create noise profile.
   - Computer Noise the quality of sound card plays a big role on the quality of recordings. When we record through a sound card the analog to digital conversion process adds distortion from quantization errors and electrical noise can be picked up from other components in the computer.
   - Lower priced sound cards are poorly shielded which makes them more susceptible to noise. Poor quality sound cards will have low resolution analog to digital converters which will introduce more distortion from quantization errors.
   - Moreover, sound card should be placed in the slot farthest away from the computer's power supply and processor. Video card should also be placed as far away from the sound card as possible. The best way to avoid picking up electrical noise from inside the computer is to use an external audio interface which connects to a computer via USB or firewire port.
   - Set Appropriate Level During recording, it is important to set the recording levels as high as possible to obtain a good Signal to Noise Ratio and maximum Dynamic range. However, it should not be set so high to produce clipping and signal distortion.
   - Level meters are usually labeled in dB with 0dB equal to the maximum level. Levels below the maximum level are shown in negative dB. Normally recording level peak should average around -6 dB and shouldn't exceed -3 dB. This will avoid clipping, simultaneously maintains a good signal to noise ratio. If recording level is set too low, any noise picked up from the analog circuits in sound card will be more apparent, because it will be proportionally louder.
   - Work In Stages It is always good to work on a copy of the original file and to experiment with different settings until satisfying results occur. All noise cannot be removed in one pass, we should work in stages and backup files should always be kept.
   - Preview The Result The audio restoration programs and plug-ins normally have `Preview', 'Bypass' and 'Noise only' options. The preview option makes us to listen to the result and fine tune the setting before applying the changes. While the noise only option makes us to hear exactly what is going to be removed.
   - Record at Higher Resolution For best possible fidelity, we should record at 24-bit resolution and after removing noise we can convert back to 16 bit. Although it takes up more disk space but it is for temporary. It is better to record in an uncompressed format; it gives a chance to clean up the file before it is encoded.

- Select Right Software It is very important to select right kind of software depending on the type of Noise. There are lots and lots of features in all software's, so after proper comparison of software features and pricing we should go for buying. It is also advisable to first use evaluation copy and then purchase the actual software.

- **Noise removal techniques:**

  - There are many different methods of audio noise reduction. Method one adopts must be based upon the nature of noise found in the audio signal. Basically, Noise reduction is a series of filters which allow us to remove specific frequencies of audio, that is, the frequencies at which the noise occurs.
  - For obtaining best results the methods/tools should be used in following sequence which is based on the category of noise: – Reduce Narrow band noise – Reduce impulse noise – Reduce crackle and distortion – Reduce Broad band noise
  - **Narrow Band Noise:** It is simply any unwanted signal that remains steady over time (repeated). It includes DC offset, hum and buzz from ground loops and acoustical noise from air handlers or motors.
  - Apply a high pass filter which removes rumble and DC offset with a proper slope setting and frequency control to determine the cut off frequency. To eliminate DC offset only set the frequency control at around 10 Hz. The harmonic notch filter set at 50 Hz or 60 Hz will be able to ensure AC hum easily removed. There are Q & gain controls, which set the width and depth of the notch filter, may also be taken care.
  - **Clicks and Pops** – De clicking Clicks and pops removal tool is generally available in all software. It scans the audio signal for spikes, remove them, then recreates the missing sound wave by analyzing pre and post click samples and interpolating the result using high order algorithms.
  - The bigger the click, the easier it is to identify and remove. At extreme settings, de clicking also may remove musical transients, like snare, drum attacks, so it is best to make two light passes by adjusting the software for shorter and longer clicks.
  - **Distortion and Crackle:** Once the larger pops, clicks and hum are removed, we should try to remove continuous crackle and distortion.
  - Crackle is a series of pops and clicks that appears continuously. It can't be identified individually from the signal. It is a form of signal distortion. De crackle tool which is directly available in Audio software's works very well.
  - The other common type of distortion is clipping (overloading) typically caused when a digital recording is made of an analogue source and the input level is too high. The extreme volume overloads the digital converters and the signal is clipped once it exceeds the peak value. This squaring produces additional frequencies in the signal which are noise or distortion.
  - Clip removal tool is commonly found to fix this kind of noise. This effect repairs audio by replacing clipped sections of static like noise with new audio data.

- **Broadband Noise:** The noise reduction effect for broadband noise works by capturing a noise profile from file and then using this profile to subtract the noise from the rest of the signal.

CHAPTER 3

SD-ROM ALGORITHM

3.1 Introduction to SD-ROM algorithm:

The main objective of the noise removal technique is to modify only corrupted samples and the uncorrupted samples are left unchanged. The SD-ROM algorithm is built in such a way that to remove the samples which are modified during transmission and to retain the samples which are not modified. The techniques like median filter and other order static filters modifies uncorrupted samples also. The SD-ROM technique 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 based on rank-ordered differences and threshold values using the sliding window mechanism.

3.2 SD-ROM algorithm in non-recursive and 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 3.1. 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.

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

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

Fig 3.1: Filter window

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

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

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

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

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

Fig 3.2: Recursive approach

1	2	3	4	5	6

Fig 3.3: Non-Recursive approach

Figure 3.2 shows recursive approach where samples no. 1*,2* are previously filtered samples while sample no. 3 is under inspection. Figure 3.3 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 3.4 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

Fig 3.4: 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 3.4.

CHAPTER 4

MEDIAN FILTERING TECHNIQUE

4.1 Introduction to Median Filtering technique:

The median filter is a non-linear ordered statistic digital filtering technique which is normally used to reduce noise drastically in an audio signals. It is one of the best windowing operators out of the many windowing operators like the mean filter, min and max filter and the mode filter. The simple idea is to examine a sample value of the input signal and decide if it is representative of the signal. Normally, instead of replacing the inspecting sample value with the mean of neighbouring sample values, the values from the surrounding neighbourhood are first sorted into numerical order, and then that inspecting sample value is replaced with the middle (median) sample value. The neighbourhood is referred to as the window.

4.2 General Median Filtering technique:

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 4.1. 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.

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

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

Fig 4.1: Filter window

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

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

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.

7) Then the center sample X(n) is replaced by the median value (or) middle value of that sliding window. Here the sample no. 1 value is replaced by the value of the sample no. 2 as shown in Figure 4.2.

1	2	3	4	5	6

Fig 4.2: Median filter window

8) Next the sliding window moves one step forward as shown in figure 4.3 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

Fig 4.3: 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 4.3.

CHAPTER 5

MATLAB

5.1 Introduction to MATLAB

MATLAB is a high-performance accent for abstruse computing It integrates computation, visualization, and programming in an easy-to-use ambiance area problems and solutions are bidding in accustomed algebraic notation. Typical uses include

- Math and ciphering
- Algorithm development
- Data accretion
- Modeling simulation, and prototyping
- Data analysis, exploration, and accommodation
- Scientific and engineering cartoon
- Application development, including graphical user interface building

MATLAB is an alternate arrangement whose basal abstracts aspect is an arrangement that does not crave dimensioning. This allows you to break abounding abstruse accretion problems abnormally those with cast and agent formulations, in atom of the time it would yield to address affairs in a scale non alternate docent such as C .The name MATLAB stands for cast laboratory MATLAB has acquired over a icon of years with a scribe from abounding users. In university environments, it is the accepted advisory apparatus for anterior and Avant Garde courses in mathematics, engineering, and science. In industry, MATLAB is the apparatus of best for high-productivity research, development, and analysis.

MATLAB actualizes the ancestors of add-on application-specific solutions alleged toolboxes. Very important to a lot of users of MATLAB, toolboxes acquiesce you to apprentice and administer specialized technology Toolboxes are absolute collections of

MATLAB functions (M-files) that extend the MATLAB ambiance to break accurate classes of problems. Areas in which toolboxes are accessible cover arresting processing, ascendancy systems, neural networks, down-covered logic, wavelets, simulation, and abounding others

5.2 The MATLAB System

The MATLAB arrangement consists of 6 capital parts

5.2.1 Development Environment

This is the set of accoutrement and accessories that advise you to use MATLAB functions and files. Abounding of these accoutrements are graphical user interfaces. It includes the MATLAB desktop and Command Window, a command history, an editor and debugger, and browsers for examination help, the workspace, files, and the seek path.

5.2.2 The MATLAB Algebraic Function

This is an all-inclusive accumulating of computational algorithms alignment from elementary functions like sum, sine, cosine, and circuitous arithmetic, to added adult functions like cast inverse, cast eigenvalues, Bessel functions, and fast Fourier transforms.

5.2.3 The MATLAB Language

This is a high-level matrix/array accent with ascendant breeze statements, functions, abstracts structures, input/output, and acquisitive programming features. It allows both programming in the small to rapidly actualize quick and bedraggled departure programs and programming in the large to actualize complete ample and circuitous appliance programs.

5.2.4 Graphics

MATLAB has all-encompassing accessories for announcement vectors and matrices as graphs, as able-bodied as annotating and pressing these graphs. It includes high-level functions for two-dimensional and three-dimensional abstracts visualization, angel processing, animation and presentation graphics it as well includes low-level functions that acquiesce you to absolutely adapt the actualization of cartoon as able-bodied at to body complete graphical user interfaces on your MATLAB applications.

5.2.5 The MATLAB Appliance Affairs Interface (API)

This is a library that allows you to address C and Fortran programs that collaborate with MATLAB. It includes accessories for calling routines from MATLAB (dynamic linking), calling MATLAB as a computational engine, and for account and autograph MAT-files.

5.2.6 MATLAB Desktop

MATLAB Desktop is the capital MATLAB appliance window. The desktop contains 6 sub windows, the command window, the workspace browser, the accepted agenda window, the command history window, and one or added amour windows, which are apparent alone if the user displays a graphic.

The command window is the area the user types MATLAB commands and expressions at the alert) and area the achievement of those commands is displayed MATLAB defines the workspace as the set of variables that the user creates in a plan session the workspace browser shows these variables and some advice about them. Double beat on a capricious in the workspace browser launches the Management Editor, which can be acclimated to access advice and assets instances adapt assertive backdrop of the variable.

The accepted Agenda tab aloft the workspace tab shows the capacity of the accepted directory, whose aisle is apparent in the accepted agenda window. For example, in the windows operating arrangement the aisle ability be as follows: CAMATLAB Work, advertence that agenda work" a subdirectory of the capital agenda "MATLAB"; WHICH IS INSTALLED IN DRIVE C beat on the arrow in the accepted agenda window shown an account of afresh acclimated paths Beat on the button to the appropriate of the window allows the user to change the accepted directory.

MATLAB is a deck aisle to accretion M-files and added MATLAB accompanying files, which are adapted in directories in the computer book system, any book run in MATLAB has to abide in the accepted agenda or in an agenda that is on seek path.

5.2.7 Using the MATLAB Editor to actualize M-Files

The MATLAB editor is both an argument editor specialized for creating M-files and a graphical MATLAB debugger. The editor can arise in a window by itself, or it can be a sub window in the desktop M-files are denoted by the addendum m, as in pixelup.m. The MATLAB editor window bus abundant pull-down airheaded for tasks such as saving, viewing and debugging files Because it performs some simple checks and as well uses blush to differentiate amid assorted elements of code, this argument editor is recommended as the apparatus of best for autograph and alteration M-functions. To access the editor. blazon adapter at the alert opens the M-file filename.m in an editor window, accessible for editing as acclaimed earlier, the book has to be in the accepted directory, or in an agenda in the seek path.

5.2.8 Getting Help

The arch way to get advice online is to use the MATLAB advice browser opened as an abstract window either by beating on the catechism mark attribute() on the desktop toolbar, or by accounting advice browser at the alien in the command window. The advice Browser is a web browser chip into the MATLAB desktop that displays a Hypertext Markup Language (HTML) document. The Advice Browser consists of two panes, the advice navigator pane, acclimated to accretion information, and the affectation pane, acclimated to appearance the information Self- explanatory tabs added than navigator area is acclimated to accomplish a search.

5.3 Communication

The Communications Arrangement Toolbox provides algorithms and accoutrement for the design, simulation, and assay of communications systems. These capabilities are provided by the MATLAB function, MATLAB Arrangement objects and Simulink block. The arrangement toolbox includes algorithms to antecedent coding, access coding, interleaving modulation equalization, synchronization, and access modelling, Accoutrement are provided for bit absurdity amount analysis, breeding eye and afterlife diagrams, and visualizing access characteristics. The arrangement toolbox as well provides adaptive algorithms that let you archetypal activating communications systems that use OFDM. OFDMA, and MIMO techniques. Algorithm's abutment fixed-point abstracts accession and C or HDL cipher generation.

5.4 Key Features

Algorithms for designing the concrete band of communications systems, including antecedent coding access coding, interleaving modulation, access models. MIMO equalization and synchronization.

- GPU-enabled Arrangement tar for computationally accelerated algorithms such as Turbo, LDPC, and Viterbi decoders Alternate accommodation tools, including eye diagrams, constellations and access drop functions
- Graphical apparatus for comparing the apish bit absurdity amount of an arrangement with analytical results.
- Access models, including AWGN. Multipath Rayleigh Fading. Rician Fading. MIMO Multipath Fading, and LTE MIMO Multipath Fading
- Basal RF impairments, including nonlinearity, actualization noise, thermal noise and actualization and abundance offsets
- Algorithms accessible as MATLAB function MATLAB Arrangement objects, and Simulink blocks
- Abutment for fixed-point clay and C and HDL cipher generation

5.5 System Design

The architecture and simulation of a communications arrangement requires allegory its acknowledgment to the babble and arrest inherent in real world environments, belief its behaviour appliance graphical and quantitative means, and free whether the consistent achievement meets standards of acceptability. Communications Arrangement Toolbox accoutrements a array of tasks for communications arrangement architecture and simulation. Abounding of the functions, Arrangement objects and blocks in the arrangement toolbox accomplish computations associated with an accurate basic of a communications system, such as a demodulator or equalizer. Added capabilities are advised for accommodation or analysis.

5.6 System Characterization

The arrangement toolbox offers several accepted methods for quantitatively anecdotic arrangement performance:

- Bit absurdity amount (BER) computations
- Adjoining access ability arrangement (ACPR) measurements
- Absurdity agent consequence (EVM) measurements
- Accentuation absurdity arrangement (MER) measurements

Because BER computations are axiological to the assumption of any communications system. the arrangement toolbox provides the afterward accoutrement and capabilities for configuring BER analysis scenarios.

5.7 BER Simulations

BER apparatus - A graphical user interface that enables you to BER achievement of communications systems. You can assay achievement visa simulation-based, semi analytic, or abstract approach. Error Amount Analysis Console - A MATLAB article that runs simulations for communications systems to measure absurdity amount performance It supports specified analysis credibility and bearing of parametric achievement plots and surfaces Accelerated achievement can be accomplished if active on a multicore accretion platform.

Multicore and GPU dispatch - A adequacy provided by Parallel Accretion Toolbox that enables you to advance simulation achievement appliance multicore and GPU accoutrements aural your computer. Distributed accretion and billow accretion abutment Capabilities provided by Parallel Accretion Toolbox and MATLAB Distributed Accretion Server that accredit you to advantage the accretion ability of your server farms and the Amazon EC2 Web service.

5.8 Performance Visualization

The arrangement toolbox provides the afterward capabilities for visualizing arrangement performance: Access accommodation apparatus -For visualizing the characteristics of a crumbling access Eye diagrams and arresting after life be sprinkle plots- For a qualitative, beheld compassionate of arrangement behaviour that enables you to accomplish antecedent architecture decisions Signal aisle plots - For a connected account of the signal's aisle amid accommodation points.

BER plots - For visualizing quantitative BER achievement of an architecture candidate, parameterized by metrics such as SNR and fixed-point char size.

5.9 Analog and Digital Modulation

Analog and digital modulation techniques encode the information stream into a signal that is suitable for transmission Communications System Toolbox provides a number of modulation and corresponding demodulation capabilities. These capabilities are available as MATLAB functions and objects MATLAB System objects and Simulink blocks Modulation types provided by the toolbox are

Analog: AM, FM, PM, SSB, and DSBSC.

Digital: FSK, PSK, BPSK, DPSK, OQPSK, MSK, PAM, QAM and TCM.

5.10 Source and Channel Coding

The Communications System Toolbox provides source and channel coding capabilities that let you develop and evaluate communications architectures quickly enabling you to explore what if scenarios and avoid the need to create coding capabilities from scratch.

5.10.1 Source Coding

Source coding, also known as quantization or signal formatting is a way of processing data in order system toolbar provides a variety of types of algorithms for implementing source coding and decoding, including

- Quantizing
- Companding A-law and Mu-law
- Differential pulse code modulation (DPCM)
- Huffman coding
- Arithmetic coding

5.10.2 Channel Coding

To combat the effects of noise and channel corruption, the system toolbox provides block and convolutional coding and decoding techniques to implement error detection and correction. For simple error detection with no inherent correction, a cycle

redundancy check capability is also available Channel coding capabilities provided by the system toolbox include:

- BCH encoder and decoder
- Reed-Solomon encoder and decoder
- LDPC encoder and decoder
- Convolutional encoder and Viterbi decoder
- Orthogonal space-time block code (OSTBC) (encoder and decoder for MIMO channels)
- Turbo encoder and decoder examples

CHAPTER 6

SIMULATION RESULTS

6.1 Signal to noise ratio (SNR):

Signal to noise ratio is the ratio of signal power to noise power. Signal to noise ratio should be as high as possible So noise power should be very less. While calculating the snr only the internal noise is considered in its calculation because they exist for long time. Most dominant internal noises are additive white Gaussian noise and impulsive noise. So, the snr should be as high as possible at the output of the receiver so that signal can be retrieved back. So, snr is a very important parameter.

SNR (dB) = $10 \log_{10}(s/n)$

- s = signal power (W), n = noise power (W).

6.2 Peak Signal to noise ratio (PSNR):

Peak Signal to Noise Ratio is a comparison between the maximum intensity of the signal over time compared to the noise floor. This can be a useful metric to approximate quality of compression by comparing the source "signal" to the encoded target and reporting how much compression "noise" affected the signal in the output.

PSNR (dB) = $20 \log_{10} (MAX/\sqrt{MSE})$

- MAX means maximum value = 255 (8 bits per sample).
- More generally, when samples are represented using linear PCM with B bits per sample, MAX_I is 2^B-1.
- MSE means mean square error
  Error (E) = (true value – observed value), Square (S) = E^2,

  Mean (M) = S/no. of samples, Hence MSE = S/no. of samples

6.3 SNR Enhancement:

Filtering techniques like low pass filtering and median filtering as the filtering is applied uniformly over the whole signal so it modifies both corrupted and uncorrupted samples. So, the SNR value is low compared to SD-ROM technique. In SD-ROM technique only the corrupted samples would be replaced, so that's why SNR is more. In SD-ROM technique, it also depends on threshold values i.e.; based on the window size.
For a window of size five SNR would be more.

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 T1=4 and T2=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 (dB)	PSNR (dB)
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 T1=6, T2=8, T3=14.

6.4 Performance Comparison:

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, 3, 4 at 5%, 10%, 50% noise rate.

Table 2: Performance comparison table at 5% noise rate (5001 samples)

Filter Type	Window Size	SNR (dB)	PSNR (dB)
SD-ROM (recursive)	5	34.0517	43.7349
	7	31.6198	41.4244
SD-ROM (non-recursive)	5	32.9723	42.6593
	7	29.9993	39.8651
Median Filter	5	30.1229	43.1060
	7	27.9793	37.2521

Table 3: Performance comparison table at 10% noise rate (12001 samples)

Filter Type	Window Size	SNR (dB)	PSNR (dB)
SD-ROM (recursive)	5	32.2083	41.3815
	7	29.0828	38.2625
SD-ROM (non-recursive)	5	30.0129	39.1714
	7	29.5542	37.8631
Median Filter	5	27.4623	38.1994
	7	25.6667	36.9978

Table 4: Performance comparison table at 50% noise rate (400001 samples)

Filter Type	Window Size	SNR (dB)	PSNR (dB)
SD-ROM (recursive)	5	27.6748	35.9990
	7	25.0689	33.4801
SD-ROM (non-recursive)	5	24.7260	32.0145
	7	20.0016	29.6532
Median Filter	5	23.3215	31.4222
	7	19.8686	28.6882

By this Table 2, 3, 4 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.

6.5 GRAPHS

For 10% corrupted signal (12001 samples):

1) SD-ROM algorithm in recursive version graphs:

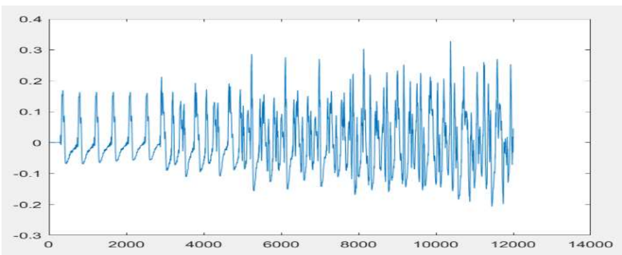

Fig 6.1: Original audio signal

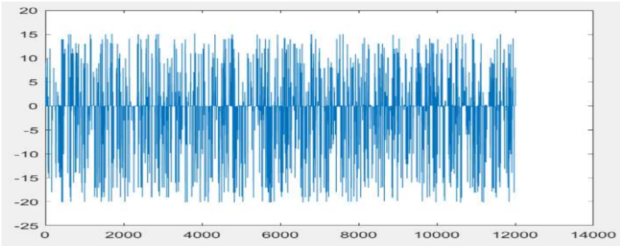

Fig 6.2: Noisy audio signal (10% noise rate)

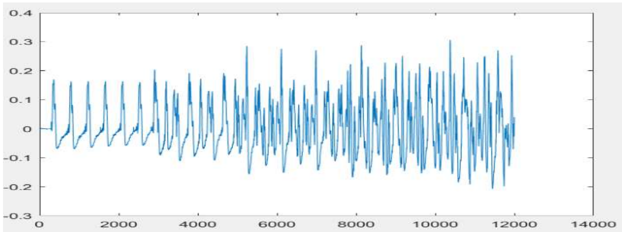

Fig 6.3: SD-ROM recursive version

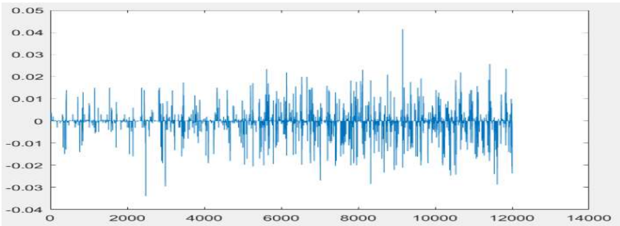

Fig 6.4: Difference between original audio and SD-ROM signal

2) Median filter Graphs:

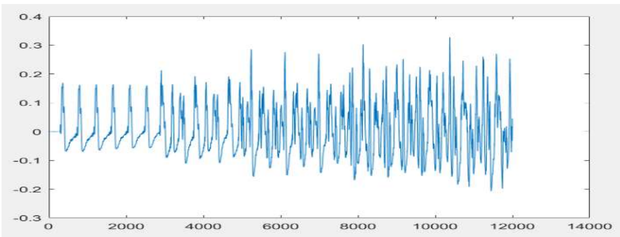

Fig 6.5: Original audio signal

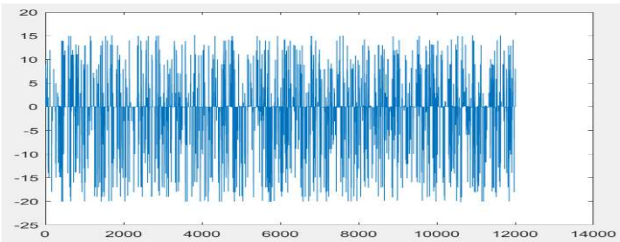

Fig 6.6: Noisy audio signal (10% noise rate)

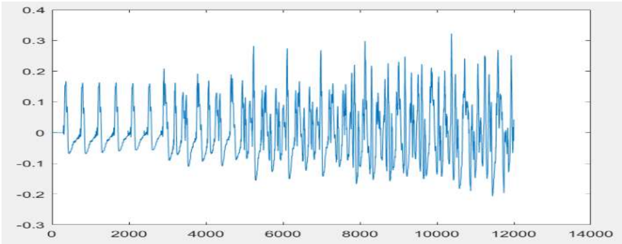

Fig 6.7: MEDIAN FILTER

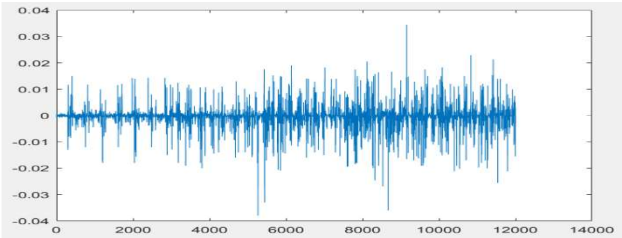

Fig 6.8: Difference between original audio and MEDIAN FILTER signal

Figure 6.1 shows the original sampled audio signal and this signal is corrupted at 10% noise rate as shown in figure 6.2, this signal is filtered using SD-ROM recursive version and the output is shown in figure 6.3 and the figure 6.4 shows the difference between original audio and SD-ROM signal.

Figure 6.5 shows the original sampled audio signal and this signal is corrupted at 10% noise rate as shown in figure 6.6, this signal is filtered using general MEDIAN FILTER and the output is shown in figure 6.7 and the figure 6.8 shows the difference between original audio and MEDIAN FILTER signal.

% Noise Rate: Consider an audio signal of 12001samples is artificially corrupted at 10% noise rate means the 1200 samples are corrupted with impulse noise.

CHAPTER 7

CONCLUSION AND FUTURE SCOPE

7.1 Conclusion:

SD-ROM algorithm in recursive version is an efficient method in removing impulsive noise from audio signals than other impulse noise removal techniques. 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, scratches and static on these discs are essentially modeled as impulse noise, which are detected and removed by our technique. Other applications of this technique could be in telecommunications, where noise both in regular and cellular telephones and could be reduced without sacrificing tonal quality with a fast hardware implementation of the algorithm., in telecommunication systems, etc.

7.2 Future Scope:

Our sense of hearing provides us rich information about our environment with respect to the locations and characteristics of sound producing objects. For example, we can effortlessly assimilate the sounds of birds twittering outside the window and traffic moving in the distance while following the lyrics of a song over the radio sung with multi-instrument accompaniment. The human auditory system is able to process the complex sound mixture reaching our ears and form high-level abstractions of the environment by the analysis and grouping of measured sensory inputs. The process of achieving the segregation and identification of sources from the received composite acoustic signal is known as auditory scene analysis. It is easy to imagine that the machine realization of this functionality (sound source separation and classification) would be very useful in applications such as speech recognition in noise, automatic music transcription and multimedia data search and retrieval. In all cases the audio signal must be processed based on signal models, which may be drawn from sound production as well as sound perception and cognition. While production models are an integral part of speech processing systems, general audio processing is still limited to rather basic signal models due to the diverse and wide-ranging nature of audio signals. Important technological applications of digital audio signal processing are audio data compression, synthesis of audio effects and audio classification. While audio compression has been the most prominent application of digital audio processing in the recent past, the burgeoning importance of multimedia content management is seeing growing applications of signal processing in audio segmentation and classification. Audio classification is a part of the larger problem of audio-visual data handling with important applications in digital libraries, professional media production, education, entertainment and surveillance. Speech and speaker recognition can be considered classic problems in audio retrieval and have received decades of research attention. On the other hand, the rapidly growing archives of digital music on the internet are now drawing attention to wider problems of nonlinear browsing and retrieval using more natural ways of interacting with multimedia data including, most prominently, music. Since audio records (unlike images) can be listened to only sequentially, good indexing is valuable for effective retrieval. Listening to audio clips can actually help to navigate audio-visual material more easily than the viewing of video scenes. Audio classification is also useful as a front end to audio compression systems where the efficiency of coding and transmission is facilitated by matching the compression method to the audio type, as for example, speech or music.

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.

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. 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]

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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.

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

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,

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

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

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

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

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

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 (dB)	PSNR (dB)
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 (dB)	PSNR (dB)
SD-ROM (recursive version)	5	32.2083	41.3815
	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:

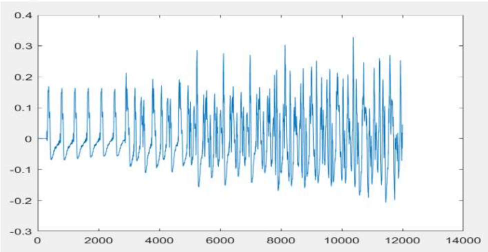

Figure 2.1: Original audio signal

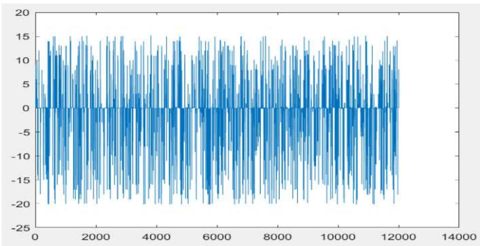

Figure 2.2: Noisy audio signal (10% noise rate)

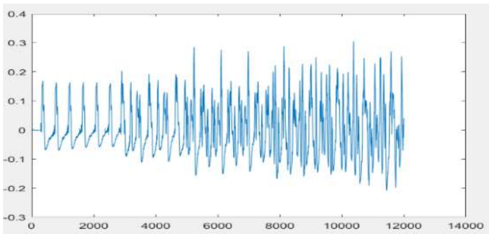

Figure 2.3: SD-ROM recursive version

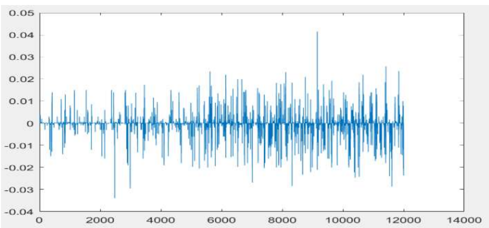

Figure 2.4: Difference between original audio and SD-ROM signal

Median filter Graphs:

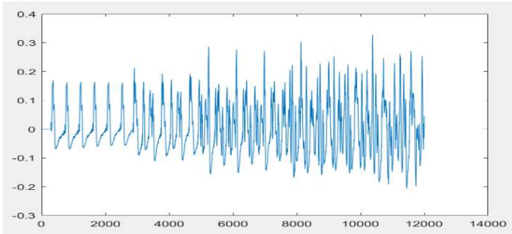

Figure 3.1: Original audio signal

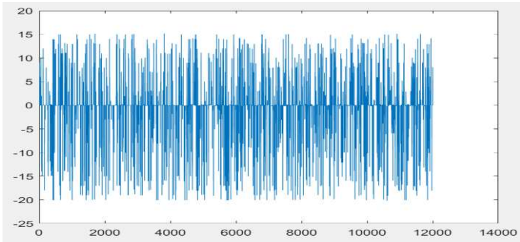

Figure 3.2: Noisy audio signal (10% noise rate)

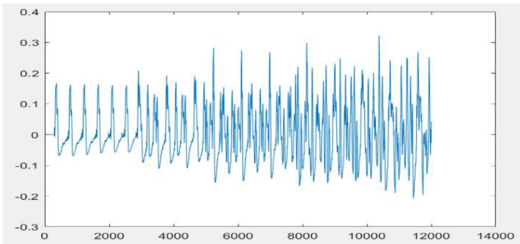

Figure 3.3: MEDIAN FILTER

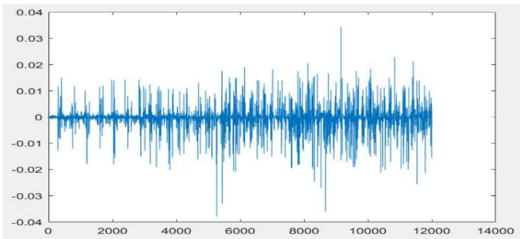

Figure 3.4: Difference between original audio and MEDIAN FILTER signal

V. CONCLUSION

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