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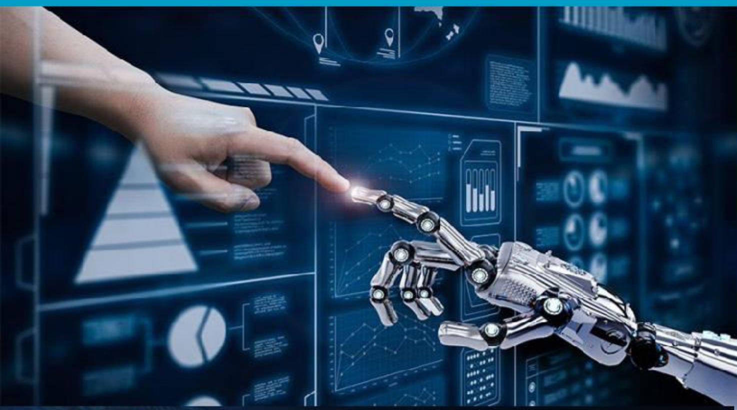

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A Comprehensive Summary of Digital Signal Distortion Mitigation

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ABSTRACT: An undesirable signal that obstructs the assessment or communication of another signal is referred to as noise. A signal that communicates information about the source of the noise is a noise itself. New concepts for wideband stationary/non-stationary elimination of noise for audio signals are presented in this research. Although current noise reduction approaches have usually been shown to be successful, they frequently display some unwanted qualities. A common issue is the distortion and/or change of the audio properties of the primary audio sound. The majority of commercially accessible noise mitigation programmes employ a spectrum exclusion methodology based in the frequency domain. The concept of spectral subtraction entails estimates of the noise spectrum and subtraction on a frame-by-frame basis across distinct frequency bins. In overall, noise spectrum estimates errors and noise variance within a frame create traces of audible artefacts in the processed audio that vary in frequencies for each frame, such as grating melodic noise or brief sinusoidal impulses. Alternatively, adaptive filters like the Kalman and Adaptive Weiner filters are used in time domain noise filtering approaches.

KEYWORDS: Audio signal Distortion, Noise Filtering
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I. INTRODUCTION

People are pursuing automatic and integrated home audio experiences as the use of intelligent digital audio technology increases. Due to their sophisticated and useful features, gadgets like intelligent speakers and the social network portal are in popularity. The audio sector is at the cutting-edge of technologies like speech recognition, 360-degree audio, and wireless audio. Communication has become more natural and surreal for listeners thanks to the idea of immersive audio. These high-quality audio experiences are the result of a number of tools and methods that polish the sound and play a big part in producing excellent audio.

The primary constraints on communication and measuring systems are noise and distortion. As a result, the foundation of communications and signal processing theory and practise has been the modelling and elimination of the effects of noise and distortion. Areas where the signals cannot be separated from noise and distortion include cellular mobile communication, speech recognition, picture processing, medical signal analysis, radar, and sonar. Noise mitigation and distortion mitigation are significant issues in all of these applications.

Face-to-face meetings are giving way to remote communication, such virtual video conferences, as the preferred means of communication. However, every communication mechanism will inevitably include acoustic noise, distortion, and echo. Imagine someone conversing on the phone or pacing the streets. Traffic noise, ambient noise from those around him, wind noise, etc. would all interfere with his ability to speak. It becomes essential to eliminate this distortion for a clear and perfect sound. A variety of methods are employed to raise the audio quality.

Although there is some overlap between the ideas of distortion and noise in signal processing, they are two different phenomena. The alteration or modification of the waveform's shape or additional feature is referred to as distortion in a signal. Contrarily, noise is a random signal that is added to the original signal from outside. The impacts of noise are more difficult to eliminate than the consequences of distortion. In comparison to distortion, noise is likewise more stochastic.

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II. AUDIO SIGNAL PROCESSING

The process of applying complex methods and algorithms to audio signals is known as audio signal processing. Sound is represented through audio signals, which come in both digital and analogue formats. Our ears can only hear sounds within 20 and 20,000 Hz, which is the range of their frequencies. Electrical signals contain analogue signals, whereas binary representations contain digital signals. By combining digital and analogue signals, this method involves eliminating undesirable noise and harmonising the time-frequency bands. It emphasises computational techniques for modifying the sounds. By incorporating several strategies, it reduces or eliminates undesirable noise like as echo and overmodulation.

III. TYPES OF NOISE

White Noise: White is the culmination of all colours in the visible colour spectrum. White in an electrical spectrum is noise that is present over a range of frequencies. Perfect white noise suggests the well-known Gaussian distribution in the form of a bell

Spurious signals: There can be a signal at a certain frequency originating from somewhere that is unrelated to the signal of relevance. Soon after purchasing them, the subwoofer started to have an unpleasant hum. A rattle was produced by a small thump, and this rattle flowed back into a resonant tone. Fix it by tightening the screws. Other instances include EMI-induced erroneous signals, component resonances, and uncontrolled sources of capacitance and inductor.

Harmonics: Harmonics often have little effect on a sluggish sensor, such as one that takes a temperature measurement once per second or a reading of pressure ten times per second. As the signal's frequency increases, as in audio applications, things start to happen. A sine wave treated with precisely linear components and then flawlessly sampled at the A/D conversion would be an ideal analogue signal. A signal is distorted by any nonlinearity. Sinusoidal waves begin to exhibit harmonics when multiple integers of the fundamental wave added together. More potent second, third, fourth, and higher-order harmonics are present in squarer waves.

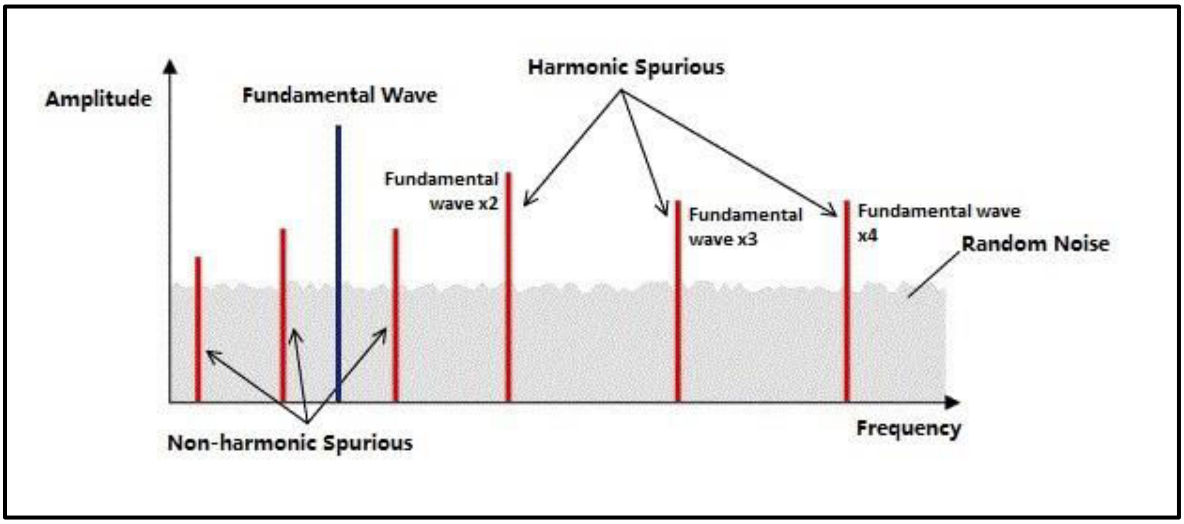

Fig. 1: A signal's harmonics, Spurious signal, and noise are all displayed alongside the signal in its frequency-domain [1].

Base-line wander (BW): BW is a low-frequency (LF) distortion that is mostly brought on by breathing, movement of the body, inadequate electrode contact, and skin-electrode impedance [5]. The electrolyte, electrode, skin, and movement characteristics all affect the wander's amplitude and duration [6]. By raising the wandering frequency and producing motion artefacts, respectively, abnormal breathing rate and electrode movement affect the ECG. BWs cause errors in the diagnosis of myocardial infarction, Brugada syndrome, and other ST-segment related disorders by distorting the ST-segment and other LF elements of the ECG signal [7].

Power-line interference: When the acquisition of the ECG signal, PLI sounds are brought on by the inductive and capacitive couplings of the 50/60 0.2 Hz power lines. It has a bandwidth of about 1 Hz, a maximum amplitude of 50%

FSD, and it is narrowband [8]. The shape of the signals is distorted when the PLI contents and the ECG are combined. Due to P-wave distortions caused by this, atrial arrhythmias such atrial enlargement and fibrillation are incorrectly diagnosed [9].

IV. DISTORTION

The inaccurate recreation of an input signal at an amplifier's output is known as distortion. Push-pull amplifiers experience crossover distortion of the output wave near its zero crossover point because of its two-stage construction. The Class-A amplifier configuration's low power effectiveness rating as a result of being biassed around its core Q-point is one of its key drawbacks.

Crossover Distortion: Simply by switching the amplifier's output stage to a Class B push-pull arrangement, we may enhance the amplifier and nearly double its effectiveness. The majority of contemporary Class B amplifiers are transformerless or complementary versions with two transistors in their output stage, which is wonderful from an efficiency standpoint. Due to its special zero cut-off biassing configuration, push-pull amplifiers have one significant fundamental issue in which the two transistors do not come together completely at the output of both sides of the waveform. Due to this issue, the output wave form exhibits some "distortion" as the signal "crosses-over" from one transistor to the other at the zero-voltage point.

As a result, a problem known as Crossover Distortion is created. Push-pull amplifiers have a severe fundamental flaw in that the two transistors do not fully converge at the receiving ends of both sides of the waveform due to their unique zero cut-off biassing configuration. This problem causes some "distortion" in the output wave form as the signal "crosses-over" from one transistor to the next at the zero voltage point. We have to presume that each transistor begins to conduct when its base to emitter voltage increases just above zero in order to ensure that the output waveform is not distorted. However, we understand that this is not the case since, in the case of silicon bipolar transistors, the base-emitter voltage has to reach at least 0.7 volts before the transistor begins to carry out owing to the forward diode voltage drop of the base-emitter pn-junction, generating this flat spot. This crossover distortions effect also lowers the output waveform's total peak to peak value, resulting in the highest power output.

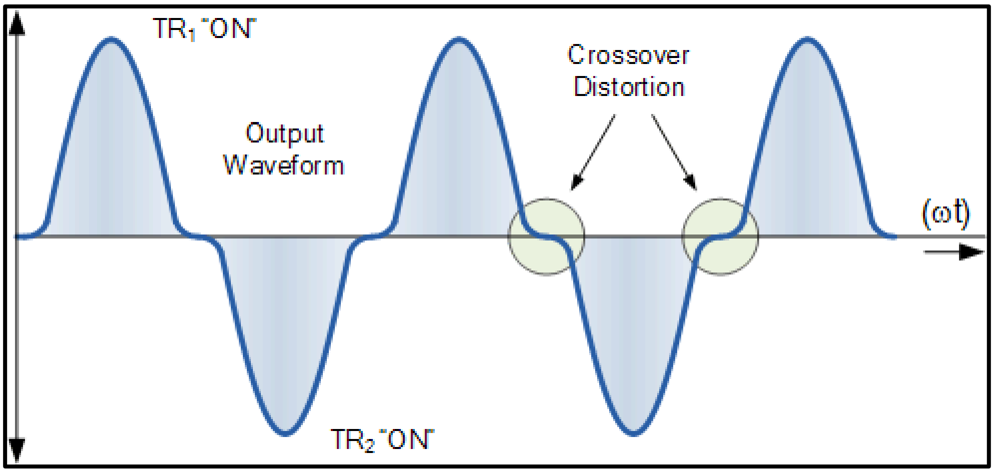

Fig.2: Crossover Distortion [4].

Crossover Distortion Reduction Using Pre-Biasing: By implementing a small forward base bias voltage to the bases of the two transistors using the input transformer's centre tap, the issue of crossover distortion can be greatly minimised. As a result, the transistors are not anymore biassed at the zero the threshold point but rather are "Pre-biased" at a level set by the new biassing voltage.

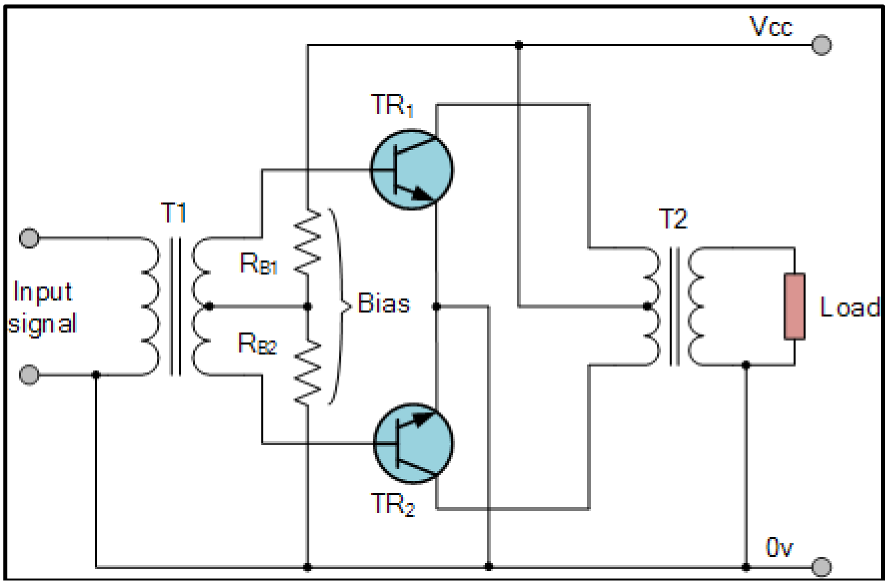

Fig.3: Push-pull Amplifier with Pre-biasing [4]

Because both transistors are now biassed just a little bit above their initial cut-off point, this kind of resistor pre-biasing enables one transistor to turn "ON" precisely at the same moment that the other transistor turns "OFF." The bias voltage, yet, has to be at least twice as high as the base to emitter voltage required to switch the transistors "ON" in order accomplish this. Transformatorless amplifiers that use complement transistors can also use this pre-biasing by simply substituting biassing diodes for the two potential divider resistors, as shown beneath.

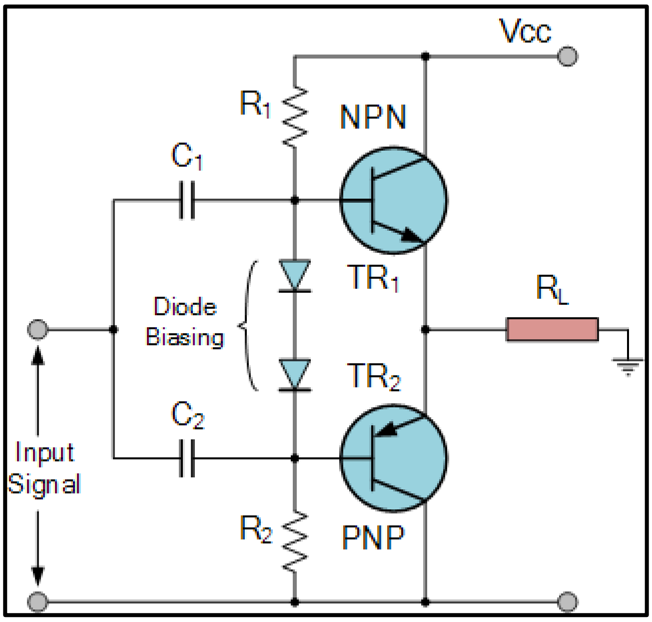

Fig.4: Pre-biasing with Diodes [4]

This pre-biasing voltage, whether used in a transformer-based or transformerless amplifier circuit, has an impact of moving the amplifiers' Q-point past the original the threshold point, enabling each transistor to function in its active area for a little more than half or 180 degrees of each half cycle. By connecting more diodes in series, the diode biassing voltage that is already present at the transistor base terminal can be multiplied. This results in an amplifier circuit that is frequently referred to as a Class AB Amplifier, and its biassing configuration is shown beneath.

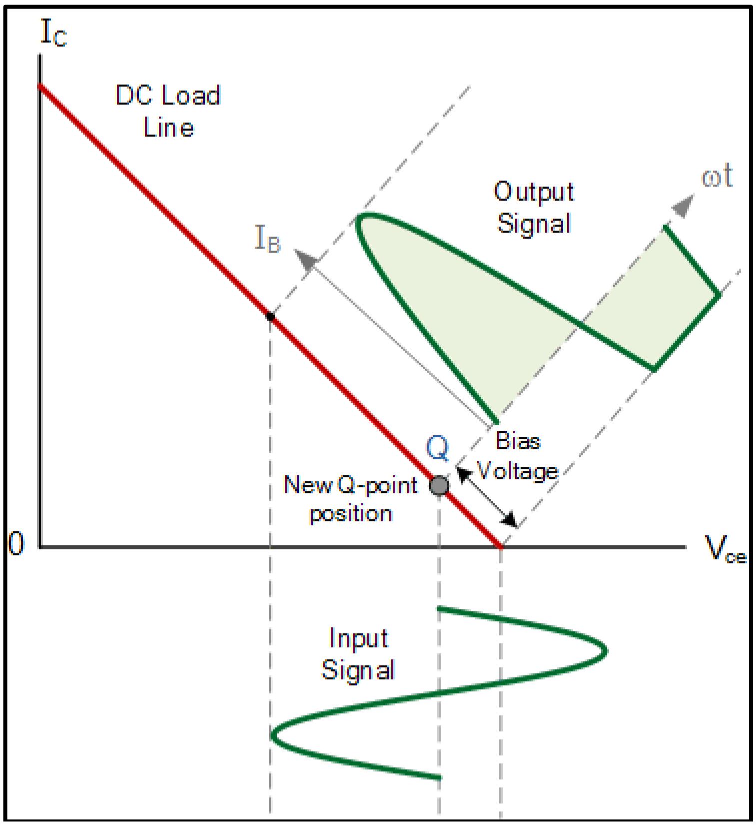

Fig.5: Class AB Output Values [4]

Class B amplifiers experience crossover distortion because the amplifier is biassed at the cut-off. Consequently, as the waveform crosses the zero axis, BOTH transistors are switched "OFF" at the same time. This crossover distortion can be considerably decreased or even totally removed by adding a modest base bias voltage, either by utilising a resistive potential divider circuit or diode biassing, and getting the transistors to the point of being just switched "ON." Another sort or class of amplifier circuit, known as a Class AB Amplifier, is created by applying a biassing voltage. The biassing level applied to the output transistors is what differentiates a pure Class B amplifier from an enhanced Class AB amplifier. Diodes have several benefits over resistors, one of which is that their PN-junctions can account for changes in the transistors' heat. Since the Class AB amplifier is basically a Class B amplifier with additional "Bias," we may state with accuracy that it is the following:

Class A Amplifiers: Since they are biassed towards the middle of the load line, there is no crossover distortion.

Class B Amplifiers: large levels of biassing at the threshold point causing crossover distortion**.**

Class AB Amplifiers: If the biassing level is adjusted too low, there may be some crossover distortion.

V. ANALOG TO DIGITAL CONVERTER (ADC)

Noise and distortion are more likely to affect analogue audio signals. They may be conveniently handled, stored, and transmitted without losing any quality by being converted into digital signals. It translates electrical impulses into binary bits resolution using a predetermined sampling rate. The quality increases with sampling rate and measurement accuracy. The signal-to-noise ratio (SNR) and bandwidth of an ADC determine its performance. SNR varies when there is a shift in resolution, accuracy, aliasing (occurs when encoded signal varies from original signal), etc. Bandwidth is defined by sampling rate. When the SNR of the ADC is higher than the input signal, the ADC is said to be at its optimum state.

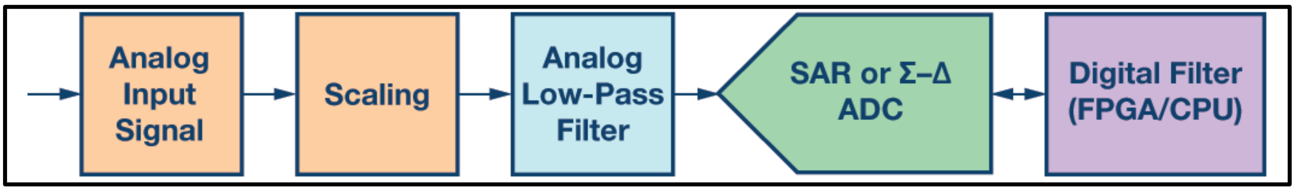

Fig. 2: simple A/D filtering [2]

VI. AUDIO PRE/POST PROCESSING SYSTEMS

The noise and any artefacts produced during the initial processing stage are suppressed using post-processing methods. Echo reduction, distortion elimination, and voice improvement are its main objectives. Popular post-processing methods to create reverberation and reduce noise include equalisation and filtering.

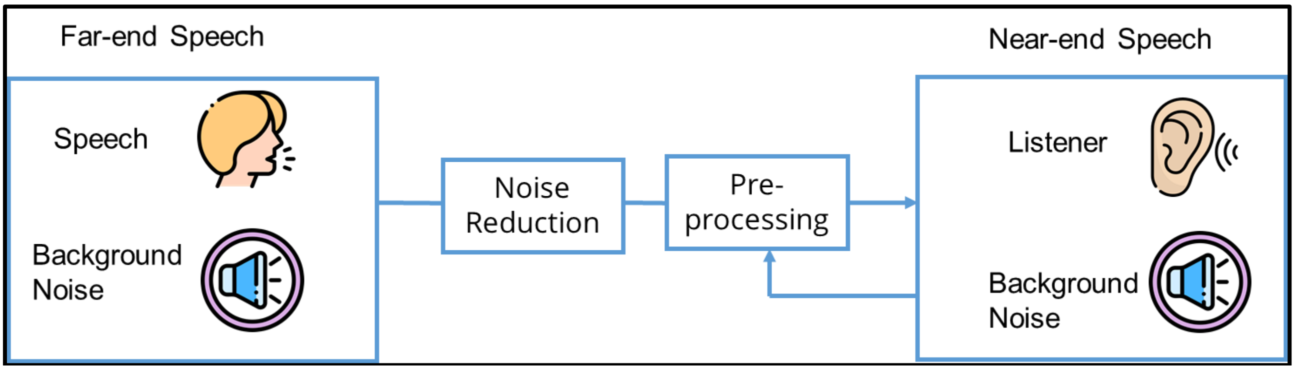

Fig.3: Audio Cleaning [3]

VII. AUDIO COMPRESSION AND DECOMPRESSION

One of the most potent mixing tools is compression, which lowers the dynamic range of audio sources. The variance between an audio signal's greatest and lowest range is known as dynamic range. For instance, whether someone is screaming or whispering, the pitch is probably too high or too low; in this instance, the sound will be distorted if it is recorded without compressing. By reducing the loudest sound and enhancing the slowest sound, the compressor resolves this issue. It gives us a more realistic sound without distortion and aids in finding the ideal audio track balancing. To preserve storage space and speed up transmission, it also decreases the bandwidth of digital audio streams and file size.

There are two distinct forms of audio compression: lossless and lossy. Because lossy methods have substantially higher compression ratios than their original data, they are among the most utilised audio compression techniques. It can get rid of any deterioration in quality and irrelevant information. The two most often used audio compressions are MP3 and AAC.

VIII. AUTOMATIC ECHO CANCELLATION

Acoustic couplings between the earpiece and speaker causes echo, reverberation, and unwanted noise, which is why it is necessary for microphones to catch far-end speech. An Acoustic Echo Canceller reduces these effects. Assume you are on a voice call with a person on the phone. The other person's speech, known as far-end speech, would be broadcast by a speaker, while your voice, known as near-end communication, would be recorded by a microphone. After certain delay, the other party would be able to hear their voice if the far-end speech were to be sent back to the other side of the conversation. The data transfer of far-end back to the other caller is prevented by AEC.

IX. RESAMPLING

The number of samples produced overall each second is referred to as resampling. The frequency of these samples is expressed in (kHz). Frame rates and sampling rates used by audio systems vary. It calculates the audio signals' frequencies. It operates on the oversampling and transcoding theory, which reduces noise and distortion. The benefits of a greater sampling rate is that it provides more accurate information about signal peaks and troughs, which enhances the sound clarity.

X. FILTERATION

The majority of signal processing applications utilise signal filtering as their main pre-processing step. The raw signal sometimes contains several types of noise, making it unusable for performing sophisticated analysis. As part of the pre-processing procedure, researchers must apply a filter to the signal to lower the noise. De-noising, one of numerous relevant pre-processing stages, is crucial when signals are collected from the environment around them. One such filter used to lower random noise in the majority of time-domain signals is a moving average filter.

The most fundamental signal processing circuit, filters are employed in virtually every step of processing. It permits the filtered data to flow through while eliminating the undesired noise, echo, and distortion.
Low-pass filter: Low-pass filters block frequencies above the threshold range while allowing frequencies below the set the threshold level.

High-pass filter: The reverse side of a low-pass filter is a high-pass filter. It minimises the frequency less than the the threshold range while filtering and passing the frequency higher than the the threshold range.

Bandpass Filter: Band pass filters are used to reduce excess noise after resampling the signals, and they are thought to be the best filters for signal processing. Only frequencies that fall within the threshold range are passed; frequencies that are more or less than that range are attenuated.

Notch Filter: Most frequencies are not affected, and those that fall within a certain range are attenuated to extremely low levels.

Moving Average Filter: A Finite Impulse Response (FIR) smoothing filter known as a moving average filter is used to remove noise and short-term overshoots from signals while preserving the underlying signal representations or sharp step response. It is a straightforward yet sophisticated statistical method for de-noising time-domain signals.

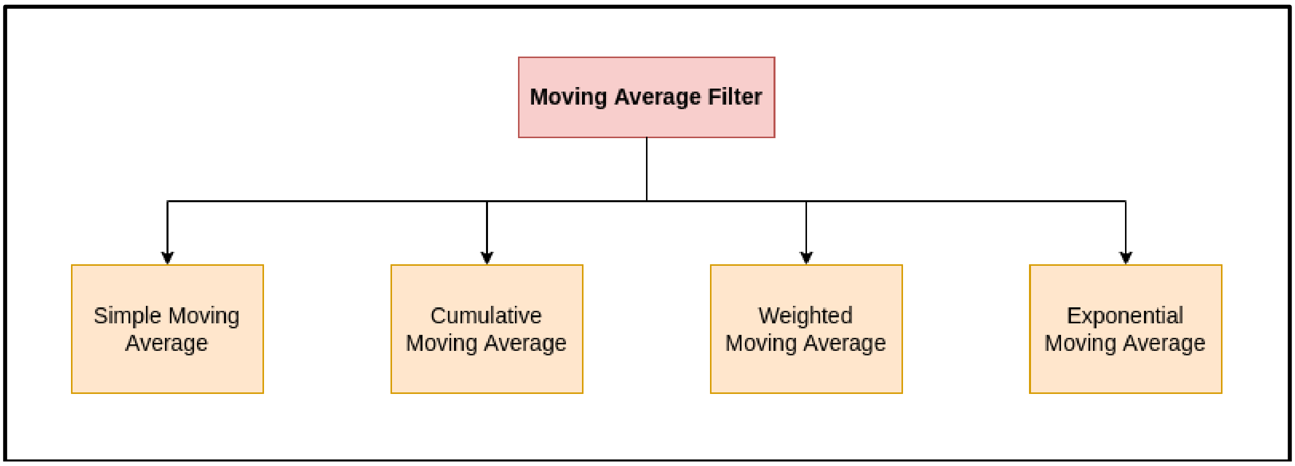

Fig.4: Types of Moving Average Filter

Simple Moving Average (SMA): This is one of the most straightforward moving average filter configurations that is simple to comprehend and use for the intended application. The fundamental benefit of the simple moving average is that its formula can be used to comprehend it, thus we don't need to know complex mathematics to understand it. One feature of the SMA is that implementing a SMA for a certain period removes any periodic fluctuations in the data because the average always contains one full cycle. A totally regular cycle, however, is quite uncommon.

Cumulative Moving Average (CMA): The CMA deviates slightly from other varieties in the moving average family, and therefore is useless for noise reduction. Contrary to SMA, which will simply be an average of past data points within a set sliding window size and equal weights, CMA also takes into account the latest data point, which has the advantage of significantly accounting for prior data.

Weighted Moving Average: When using a weighted average, one are taking into account the proportion of each of each data point inside a window length, similar to how a basic moving average filter operates. While WMA provides weight for relative proportion changes, SMA is unweighted, thereby rendering it more appealing for application. Giving data points weights aims to elevate some data points over others in importance. And the weight varies according on the use. Image processing is one of the applications, specifically the filters we employ. The local patch of pixel values on which the filters are fitted acts as the data points, and the combination serves as the weighted average. Those filter coefficients are nothing more than weights.

Exponentially Weighted Average (EWA): Current data points are given greater weight in an exponentially weighted average than overall previous data items. By still taking into account a sizable amount of the reactive nature of recent data points, this ensures that the trend is preserved. In comparison to SMA, this filter effectively and efficiently de-noises time-domain data by suppressing noisy components. It is an infinite impulse response (IIR) filter of first order.

XI. EQUALIZATION

Equalisers are used to change or regulate the frequency so that the frequency of the sound spectrum at the transmitter and receiver should match. With the use of low-pass, high-pass, and band-pass filters, frequency ranges can be made high or low. It achieves the desired output by eliminating the lag between various components of frequency.

XII. LOUDNESS CONTROL

It gives a constant output despite having various input signals. It shows the amount of gain or attenuation applied to the input signals to get the target input signal. If the input signal is higher than the target input, then AGC subtracts the gain, and if it is lower than the target input level then AGC (Automatic Gain Control) adds the gain. Gain shows the loudness of the input of the channel, which controls the tone.

XIII. BEAMFORMING

Beamforming is a signal processing method used in microphone array processing. It is often referred to as spatial filtering. Beamforming uses the spatial variety of the microphones in the array to find and retrieve the necessary source signals while suppressing undesired interference. Based on the direction of the signal source, beamforming is employed to focus and guide the directivity beam of the composite microphones. This method aids in improving the signal-to-noise (SNR) ratio and the composite range of mics.

XIV. CONCLUSION

Thanks to developments in digital audio technology, we now have speech processing technologies that are both highly effective and of very high resolution. These techniques are used when audio content is being recorded, stored, and transmitted. To achieve a desired audio quality, undesirable echoes, disruption, and distortions must be eliminated from audio content. To provide a smooth and error-free voice quality, it operates on the premise of converting audio signals between analogue and digital formats, modifying frequency ranges, eliminating undesirable noise, and introducing sound effects.

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