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A Survey of Filter Design for Audio Noise Reduction

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Abstract – *Over the past few decades, the problem of controlling noise level in the environment especially in the audio processing field has been the focus of a tremendous amount of research. Many papers have dealt with noise reduction in audio applications to improve the quality of audio signals. The purpose of this paper is to study the methods and techniques used for noise reduction in audio applications depending on certain types of scenario and audio signals. The approach of this study also include on improving speech signals in the various speech enhancement methods.*

Keywords: Audio filter design; Noise reduction, Kalman Filter, Quantization noise, Amplifier noise

1.0 INTRODUCTION

Over the past few decades, the problem of controlling noise level in the environment especially in the audio processing field has been the focus of a tremendous amount of research. Many papers have dealt with noise reduction in audio applications to improve the quality of audio signals. The large majority of papers consider the usage of low-pass filters (LPFs) using the Infinite Impulse Response (IIR) filters [1]. Steiglitz-McBride designed the modifications of the traditional IIR by reducing the filter order by 5 factors [6]. Another research proposed the usage of sampling capacitor to rotate charge between the capacitors and increase the sampling rate by using a pipelining technique [1].

The third approach using a LPF is the Cascaded Intergrator-Comb (CIC) filter by providing the basis of four over-sampling to improve the over-sampling rate designed for multi stage CIC filter [7]. LPF using three-section stub and Z-transform technique is improved by employing a discrete-time technique to overcome the drawback of the frequencies [5]. Furthermore, the design of a LPF based on a parallel-coupled line and transmission line theory is proposed to reduce the large size of interference in the wireless communications systems [4].

Another technique to reduce the noise signals in audio application includes the design of all-pass filter especially in the musical sound based environments [3]. An approach to provide variable of low-pass or high-pass responses with fine control over cut-off frequency is proposed and this design shows in savings of power consumption over other approaches [9]. Besides that, the Kalman filter is known as an effective and popular technique in the field of speech enhancement and overcome the problem of musical noise present in Wiener filtering technique [12].

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Moreover, in the field of speech recognition, the hidden Markov Model method is the most successful and flexible approach using the beamforming technique [11]. McCowan et al. applied the diffuse acoustic field model to reduce the ambient noise in this approach [10]. Other than that, a new family of filters has been proposed by allowing the tuning of the cut-off frequency to optimize the finite dimensional approximations in the filters [2]. Another design using electronic work bench (EWB) is applied in detecting the audio signal of communication systems to minimize the gap of the simulation results obtained with the theoretical analysis [8].

Based on these researches, the design of filters of noise reduction is based on several types of interference and sound sources. These include:

- Properties of the audio signal: low frequency or high frequency
- Properties of the background noise in the signal: spatially coherent or incoherent
- Amplifier noise: producing additional thermal noise
- Quantization noise: transformation of analogue to digital signal
- Loss of signal quality cause by coding and speech compression

In all practical situations, the received signal contains some form of noise components as stated above. The noise may be a result of the finite precision involved in coding or due to the addition of acoustically background noise. The problem of removing these unwanted noises from a received signal has been the subject of numerous investigations. The pioneering work from previous researchers especially in the previous ten years has led to this study.

The purpose of this paper is to study the methods and techniques used for noise reduction in audio applications depending on certain types of scenario and audio signals. The approach of this study also include on improving speech signals in the various speech enhancement methods.

2.0 METHODOLOGY

Various researches and development has been done is recent years to design a noise filter for audio for the enhancement of audio signals. However, there are many properties that need to be considered before designing a filter for the audio enhancement. In this section, we present a general survey of the properties included and the filtering techniques used by each of the different condition and scenarios.

2.1 Types of Filters according to Frequency Signal

There are two types of audio signals that are commonly categorized to design a filter, a low frequency (LF) signal and a high frequency (HF) signal. For these signals, there are four types of filters:

i. High-pass filter (HPF): attenuates content below a cutoff frequency, allowing lower frequencies to pass through the filter.
ii. Low-pass filter (LPF): attenuates content above the cutoff frequency and allowing lower frequencies to pass through the filter.
iii. Band pass filter (BPF): A circuit that passes only a band of frequencies while attenuating all frequencies outside the band.
iv. Band reject filter (BRF): attenuates only a band of frequencies while passing all frequencies outside the band.

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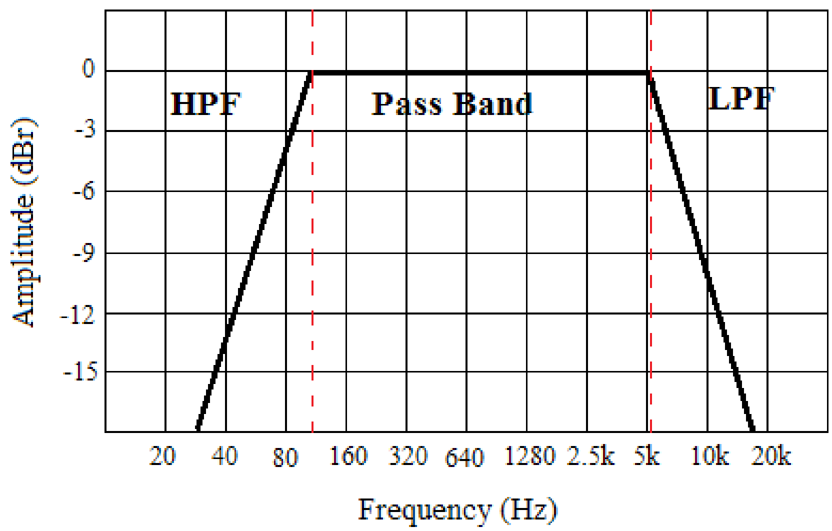

Figure 1: Frequency specific attenuation from a high-pass filter at 100Hz and a low-pass filter at 5.000 Hz. The unaffected middle band is called the pass band.

An implementation of a low-pass filter is done in previous researches. In [1], a discrete-time infinite impulse response (IIR) LPF that achieves a high-order of filtering through a charge-sharing rotation is proposed. It has a very low input-referred noise due to only one sampling capacitor for all the filtering stages. The filter consists of switches, capacitor, a clock waveform generator and a simple gm-cell.

There are three types of Discrete-Time Low-Pass IIR Filters:

i. First-Order Filter:
RC Filter
RL Filter

ii. Second-Order Filter:
RLC Filter

iii. Standard Filter:
Butterworth
Chebyshev Type I
Chebyshev Type II
Elliptic

Another type of filter used based on the types of frequencies is warped digital filters. An efficient implementation to reconfigure warped digital filters with variable low-pass, high-pass and band-pass is done by various researchers. However, there are many disadvantages in the warped filter design. The warped filters require first-order all-pass transformation to obtain the low-pass or high-pass responses, and second-order all-pass transformation to obtain the band-pass or band-stop responses. To overcome this, the implementations include a variable digital filter (VDF) whose frequency selectivity can be controlled though a small number of parameters [9]. The implementation includes by combining warped filters with coefficient decimation (CDM).

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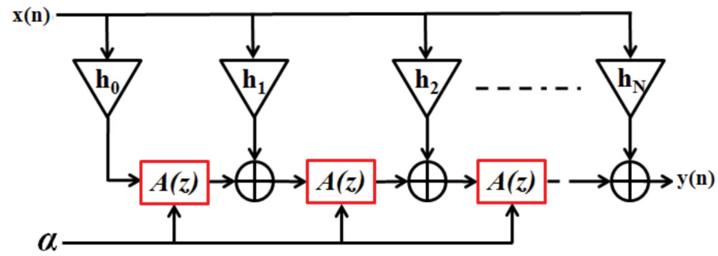

Figure 2: Warped digital filter

In the proposed design of VDF in [9], the filter coefficients are fixed and can be hardwired. The CDM is implemented using the multiplexers controlled by a signal.

Filters behaving as all-pass with time-varying characteristics are called as all-pass filter. It has a direct generalization of time-invariant designs that can lose the important norm-preserving property. A proposed filter that do preserves signal energy and reduce to simple first- and second-order all-pass filter is proposed in [3]. There are two types of all-pass filters:

I. First-Order All-Pass Filter
An approach to extend the all-pass to the time-varying case is by using the recursion to the coefficients.

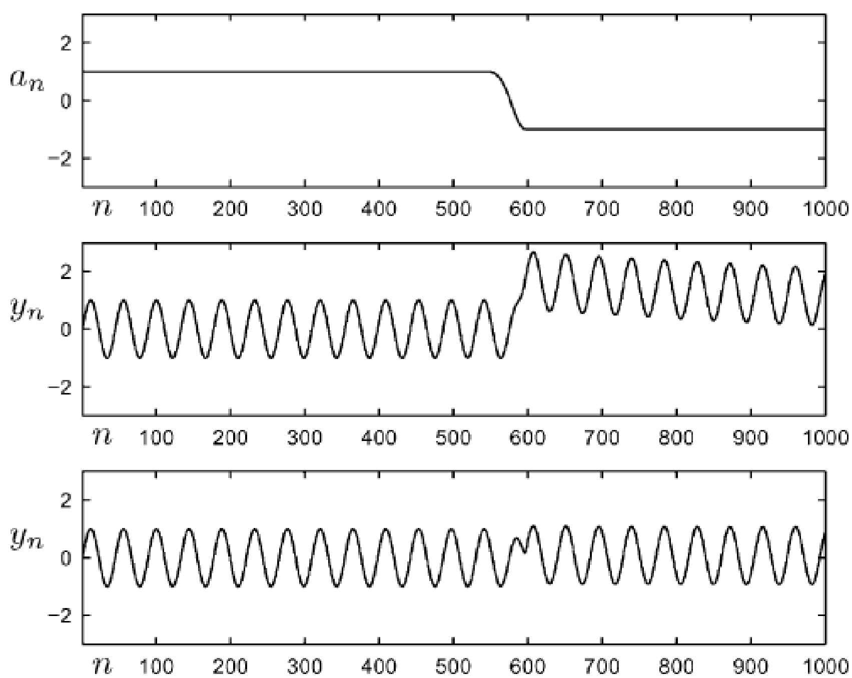

Figure 3: The results of the recursion in the first-order design in [3].

II. Second-Order All-Pass Filter:
An approach to generalize the structure to have a digital one-port form wave. Based on figure 4, the sequences M1,n and M2,n are always positive and interpreted as port resistances (for series adaptor) and as port conductances (for parallel adaptor) [3].

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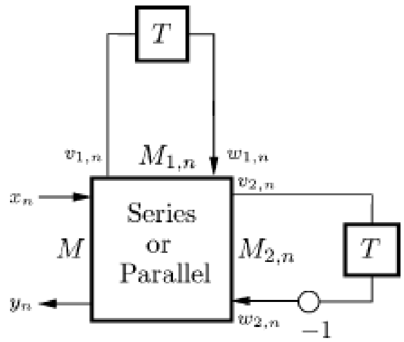

Figure 4: One-port digital wave, the generalization of a second-order all-pass filter.

2.2 Properties of the Best Method to Apply to the Filter in Noise Reduction

There are several properties of method that are apply with the filter in reducing noise in the audio. The method that are apply in the filter basically have their own pros and cons in producing the output. The method used to the filter are:

I. Cascade Integrator Comb (CIC) Filter with Delta Sigma Modulation:
A Cascaded Integrator Comb (CIC) filter is a special class of linear phase, finite impulse response (FIR) filter. Third stage of delta sigma modulator reduces the noise from the external source and feed to the input of CIC filter along with the source input. While recording is in progress external noise added in the same and make the sound noisy and it also degraded the voice quality [15]. Thus this methods need some modification in the structure to remove the external noise problem and enhance the sound quality. In order to overcome the problem faced, the adaptive delta sigma modulator with CIC filter is used for implementation. Delta sigma modulator based CIC filter gives better hardware utilization and lesser power consumption [15].

II. Adaptive Filter with Delta Sigma Modulator:
Adaptive Delta Sigma Modulator consists of two stages, the Modulator Stage which shapes the noise power spectrum by moving it as much as possible outside of the signal bandwidth in order to decrease the in band noise power, and adaption stage. Changes are made in the adaptive feedback signal based on the changes detected in the input signal power so that it can continue tracking the input signal [15].

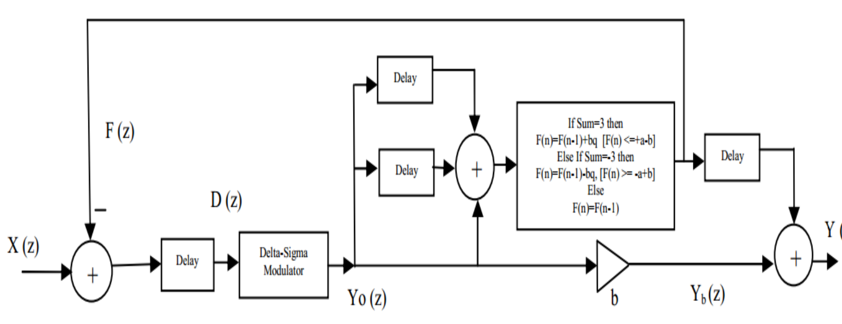

Figure 5: Adaptive Filter Delta Sigma Modulator Design

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The adaptive filter delta sigma techniques is used to enhance the stability and the dynamic range of the delta sigma modulators. The Adaptive Filter with Delta Sigma Modulation are work best in noise shaping signal.

III. Acoustic Noise Cancellation (ANC) with FIR Filter:
FIR filters are digital filters with finite impulse response. They are also known as non-recursive digital filters as they do not have the feedback (a recursive part of a filter), even though recursive algorithms can be used for FIR filter realization. ANC using FIR filter has the advantage of simplicity and stability during adaption [21]. The IIR filter is proposed as an alternative to FIR. If FIR and IIR filter uses same number of coefficients, the frequency response of IIR filter can better approximate a desired characteristic. Therefore implementing IIR filter is desirable compared to hundreds of taps in FIR filter for some application. However IIR filters are seldom used because they have the problem of instability, slow convergence and phase distortion [21].

IV. Discrete Cosine Transform (DCT)-LMS Algorithm:
DCT-LMS algorithm is very useful in speech filtering and (DCT) has higher energy compaction property for highly correlated signal which is required for noise removal purpose. So it is possible to use DCT form of input signal when it is highly correlated. If speech energy is highly concentrated in few coefficients and noise energy remained white then reduction of noise is easier with DCT. Due to use of DCT noise corrupted signal undergoes some noise reduction but not significantly. So if this signal again passed through LMS algorithm then it gives successful results. DCT provides higher spectral resolution and have real coefficients so it can be considered that they have binary phase value. DCT-LMS has high energy compaction property, faster convergence rate, provides better SNR improvement and it is easy to implement. Thus it fulfils almost all needs of noise cancellation [20].

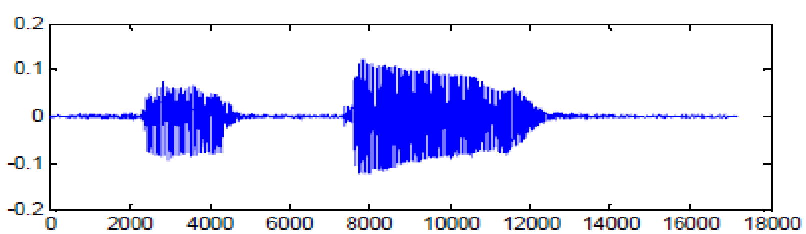

Figure 6: clean speech signal

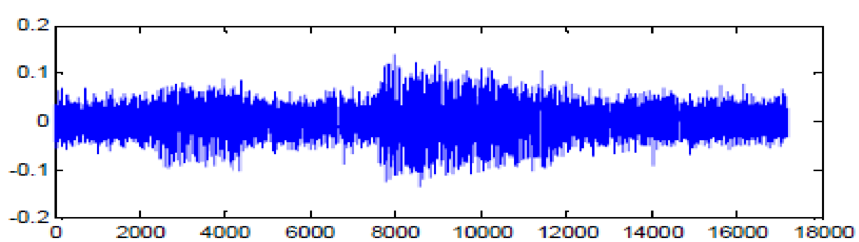

Figure 7: noise speech signal 0 Db (without using RLS)

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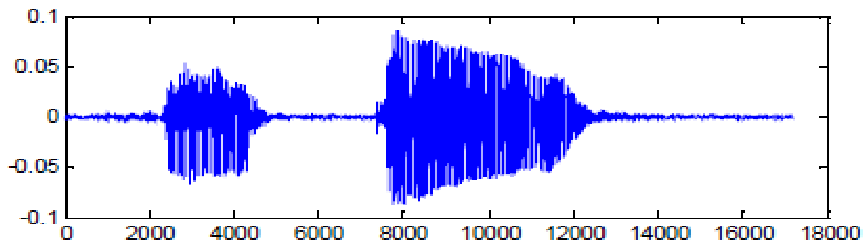

Figure 8: filter speech signal 0 db (RLS)

V. Recursive Least Square (RLS) Adaptive Filter:
The Recursive least squares (RLS) is an adaptive filter which recursively finds the coefficients that minimize a weighted linear least squares cost function relating to the input signals. Using RLS adaptive filtering is to improve the Hidden Markov Model (HMM) performance in noisy environments [11]. RLS adaptive filter is used to cancel or clean the noisy signals. The graph show that the HMM model is not producing a clear speech when there is noisy present but the RLS can improve the result of the by filtering some of noise away.

VI. Pre-filter by employing the Masking Properties in Kalman Filter:
Kalman's filtering, also known as linear quadratic estimation (LQE), is an algorithm that uses a series of measurements observed over time, containing noise (random variations) and other inaccuracies, and produces estimates of unknown variables that tend to be more precise than those based on a single measurement alone. We propose the use of a pre-filter to reduce the noise and subsequently obtain better estimates of the AR parameters. The proposed pre-filter is based on the masking properties of the human auditory system [12]. Both time domain forward masking effects and the frequency domain simultaneous masking effects are considered in the proposed system. The overall masking threshold is determined by combining the individual temporal and simultaneous masking thresholds. The results show that Kalman's filter with masking threshold are the best method can be used in enhancement speech compare to the other method show in the figure 9 below.

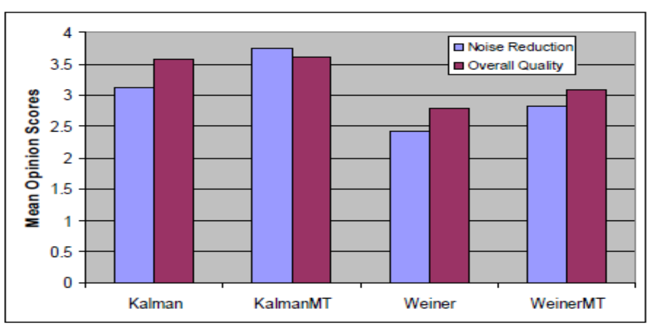

Figure 9: MOS obtained from the subjective tests for the proposed technique compared to standard speech enhancement techniques.

VII. Wiener Filter followed by Adaptive LMS algorithm:
The Wiener filter is a filter used to produce an estimate of a desired or target random process by linear time-invariant filtering of an observed noisy process, assuming known stationary signal and noise spectra, and additive noise. The Wiener filter minimizes the mean square error between the estimated random process and the desired process. Wiener filter

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will work best as long as the signal is stationary but when the signal transmitted to Wiener filter is non stationary, Wiener filter coefficient have to be adjusted based in the nature of signal. Thus, LMS Adaptive filter is needed to remove the noise in non – stationary music signal. The received signal is send to a Wiener FIR Filter. The goal of Wiener Filter is to filter out the noise that has corrupted the original signal [17]. A feedback which is based on the LMS is required which introduces adaptation to the input. In this way, a new sequence of filter coefficients is determined for the Wiener Filter. Thus, the coefficients of the Wiener Filter will adapt according to the variations in the input received signal, which will be more effective in creating an exact replica of the original signal [17]. This is represented in the simplified block diagram shown below.

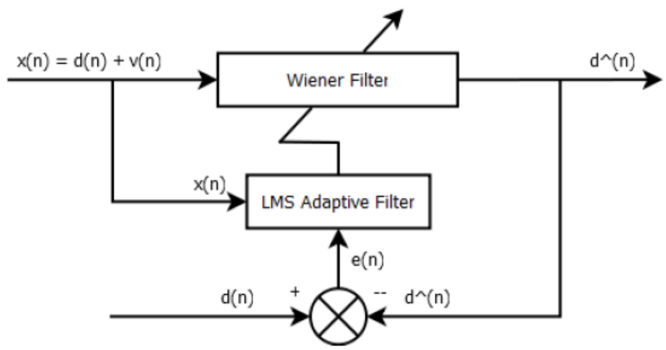

Figure 10: Simplified Block Diagram of Wiener Filter and LMS Adaptive Filter.

2.3 Transformation of Analogue to Digital System

Usually the first step is transformation of analogue to digital filter is by sampling and then digitizing it using ADC. Analogue to digital audio converter converts an incoming electrical sound wave (essentially a changing pattern of electrical voltage) into binary (1s or 0s) in order that the audio signal is recorded.

This process is sometimes referred to a Pulse Code Modulation (PCM) [13] and digital audio (PCM audio).This three types of commonly used analog filter are Gm-C, active RC and active switched-capacitor (SC) filter.

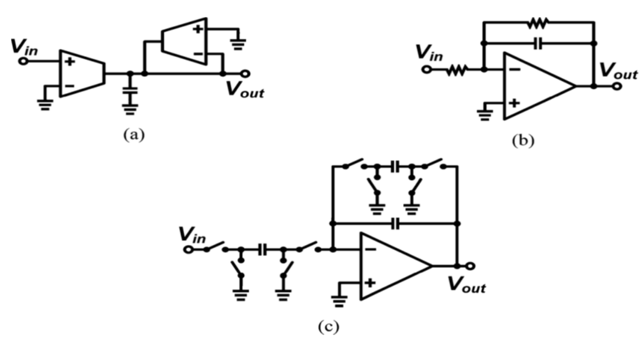

Figure 11: Conventional analog filters: (a) Gm-C, (b) active-RC, and (c) active switched-capacitor

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Analog-intensive receivers use Chebyshev [5] type of filters with complex conjugate poles to select the wanted channel out of adjacent channels while filtering out interferers and blockers. In this way, most of the filtering is done in the CT analog domain [1], and also low dynamic range ADC can be used afterwards. These transformations include very low power consumption, lowest input-referred noise and excellent linearity.

This analogue to digital works by repeatedly measuring the amplitude (volume) of an incoming electrical pressure sound wave (an electrical voltage), and outputting these measurements as a long list of the binary bytes. In this way, a mathematical 'pictures' of the shape of the wave is created.

This principal theory of conversion (sample rate and bit depth) is essential knowledge for music technologists and sound recordists who need to control quality of their audio files. The quality of analogue to digital transformation is paramount if we want to record the best quality with as little distortion as possible. The same technique is also used in noise reducing in order to get the clear audio speech by reducing the unwanted sound of surrounding.

2.4 Latest design of audio filter that was invented to improve the audio filter

Based on our survey, the latest design includes:

1) Seventh order low pass filter
2) Dynamical decoupling
3) Wiener Filter based LMS Algorithm using LabVIEW
4) Discrete wavelet technique

I. Seventh order low pass filter
Seventh order low past filter is designed based on parallel-coupled line and transmission line theory. This new invented low pass filter suppressed the harmonic signal using attenuation poles in the stop band.

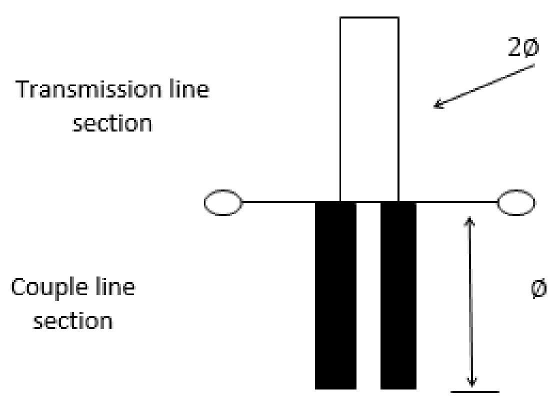

Figure 12: Schematic of coupled line low-pass filter unit [4]

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The HTS seventh-order parallel-coupled line low-pass filter has excellent attenuation characteristics up to seven times the cutoff frequency. The presented filter exhibits an excellent performance in terms of harmonics and spurious suppression. The proposed structure of the filter has an advantage in size, loss, and isolation from harmonics and spurious frequency

II. Dynamical decoupling (DD)
This filter is used to suppress decoherence of quantum systems by eliminating the system-environment coupling. Uhrig DD (UDD) is used as a benchmark sequence to illustrate the filtering process. We find that UDD sequence works as a perfect dynamical filter for low-frequency noise. For spectrum with hard cutoff, UDD increases the performance order by order with linearly increased number of pulses. This filter used the long-established spin-boson model [16].

$$H = \sum_i \omega_i b_i^\dagger b_i + \frac{1}{2}\sigma_z \sum_i \lambda_i \left(b_i^\dagger + b_i\right)$$

Where b_i (B^+_i) are canonical annihilation (creation) operators for the *i*th oscillator. UDD filter function as a flat in low-frequency region, and effectively suppresses environmental noise within low-frequency band. The UDD sequence performs well for hard cutoff spectrum, while for soft cutoff the required pulse number increases exponentially

III. Wiener Filter based LMS Algorithm using LabVIEW
This filter used least men square (LMS) adaptive filter in order to remove the unwanted noise that might occur.

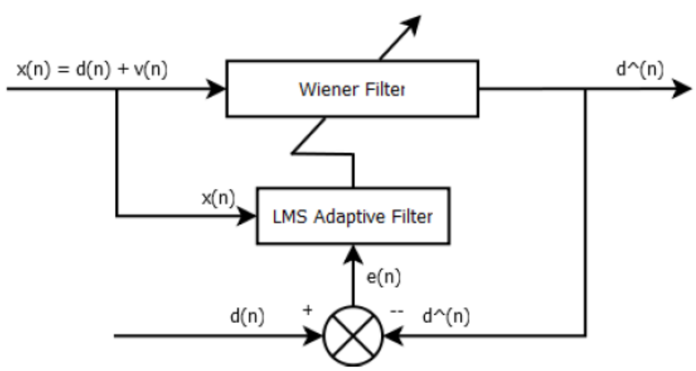

Figure 13: Simplified Block diagram

Table 1: Comparison of wavelet transforms techniques with AWGN as 5 at hard threshold technique.

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Sr. No.	Types of Wavelet	Noisy SNR	Denoised SNR	Total Elapsed time in seconds	Threshold
1	Coif5	4.99911	14.9132	42.2529	0.224576
2	Db9	5.00173	15.0536	27.3599	0.262682
3	Db10	4.99452	15.0841	28.1884	0.251217
4	Sym4	5.00548	14.7002	28.3055	0.231492
5	Sym8	5.00368	15.0206	29.1295	0.268752

The advantages of using this graphical programming (LabVIEW):

1) Easier programming and understanding of the code
2) Reduction in code size
3) Faster execution
4) Better debugging

IV. Discrete wavelet technique (DWT)

This technique is used to reduce the unwanted higher or lower order frequency components in speech signal. This filter basically used to modify the characteristic of a filter to meet new specifications without repeating the filter design procedures [18].

From the above results the DWT Coif wavelet with hard threshold and soft threshold and Sym4 hard and soft threshold is implemented and compared. In this Coif wavelet with soft threshold is best as compared to coif hard threshold and Sym4 wavelet with hard and soft threshold. In DWT soft threshold results are has been best as compared to hard threshold.

C. Adaptive Filter for Fully Digital Audio Amplifier

Recently, the fully digital audio amplifiers become used in some commercial products because of its merits [13]. The power efficiency is very high since the class-D amplifiers are used for the final stage. In the class-D amplifiers, the output signals are produced by switching operation and identically there are no power losses [13].

The merits of the high efficiency are not the only the reduction of the consuming power but also the small heat generation which enables a small size of the signal processing is achieved in digital manner and the only analog part is the output stage of the amplifiers. It does not requires any digital-to-analog converter if the source signals are digital ones [13].

In the fully digital audio amplifiers, the final stage switching amplifiers driven by the PWM signals generate distortions [13]. There are several reasons for the distortion, and the dead times of the full bridge circuits are difficult to be fixed. To reduce the quantization noise in the frequency region lower than 20 kHz, the noise-shaping filter detects the quantization error and feedbacks to quantize input signal. Digital filters are used to eliminate undesired signal and hence to extract required information of the signal [15]. These signal processes are performed in digital manner until the PWM signal is generated.

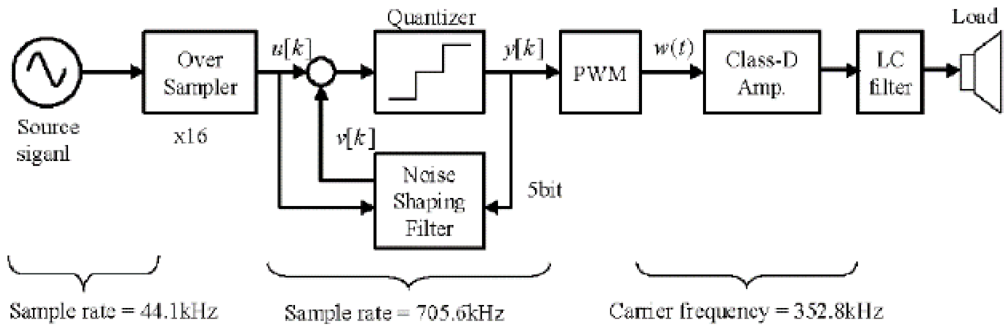

Figure 14: Structure of fully digital audio amplifier.

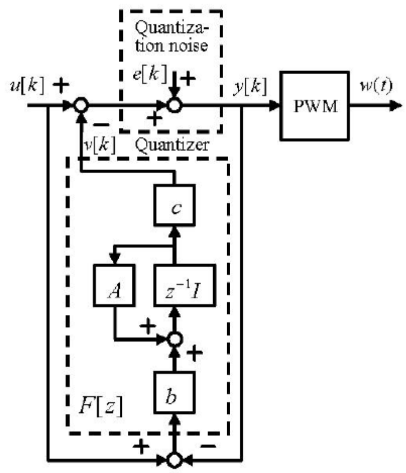

Figure 15: Structure of an ordinary noise shaping filter.

Table 1: Adaption algorithm

Sample values			Adaptive feedback Signal f(n)
ω(n)	ω(n-1)	ω(n-2)	
+1	+1	+1	f(n)= f(n-1)+bq
-1	-1	-1	f(n)= f(n-1)-bq
All other combinations			f(n)= f(n-1)

Adaptive filter gives better noise shaping performance for fully digital audio amplifier. Adaptive filtering can be considered as a process in which the parameters used for the processing of signals changes according to estimated mean squared error or the correlation [15]. Various adaptive filter algorithms have been proposed by researchers. There are four types of adaptive filters [15,21]:

i. Adaptive Delta Sigma Modulator: performs quantization operation within a feedback loop, which computes the difference between the instantaneous input signal and the quantized output, followed by an integrator.

ii. Subband Adaptive Filters (SAF): to remove echo in telephone communication. With increased number of subbands in the filter, convergence rate improves considerably at the cost of the computational complexity.

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iii. Adaptive Filter Length selection: a short adaptive filter converges faster than a long, although a long adaptive filter is necessary to model real systems.
iv. Adaptive Line Enhancer (ALE): known as traditional Acoustic Noise Cancellation (ANC), yield the cancellation of the wide band noise from the corrupted speech signal but did not address the problems of sinusoidal noise.

Among the five type of adaptive filter above, only two are consider using for fully digital audio amplifier. To determine the appropriate updating of the filter coefficients, the general set up of adaptive filtering environment is shown in figure 17.

The noise shaping filter for fully digital audio amplifiers are considered and the adaptive filter is proposed for the PWM distortion compensate.

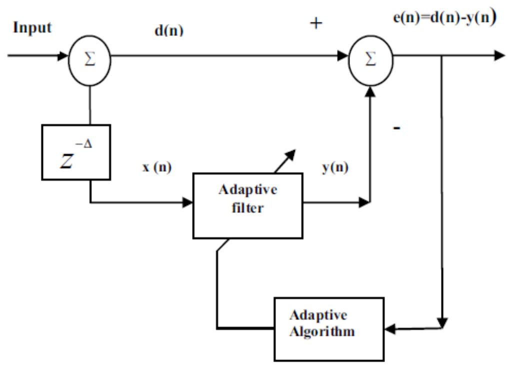

Figure 16: Block diagram of the adaptive line enhancer.

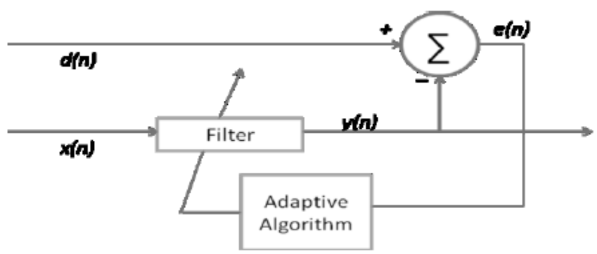

Figure 17: General setup of adaptive filter.

3.0 RESULTS

3.1 FIR and IIR Filters

Based on the different techniques of filter design from different types of frequencies, each of the filters are suitable with different frequencies. The IIR filters can achieve a given filtering characteristic using less memory and calculations than a similar FIR filter [12]. However, when

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comparing IIR with FIR filters, the FIR filter is more advantageous than IIR filter. The disadvantageous of IIR filters are as follows:

-They are more susceptible to problems of finite-length arithmetic, such as noise generated by calculations, and limit cycles

-They are harder to implement using fixed-point arithmetic.
-They don't offer the computational advantages of FIR filters for multirate (decimation and interpolation) applications.

Furthermore, the different types of filter in the Discrete-Time Low-Pass IIR filter has different advantageous and disadvantageous. Based on this study, the results are as follows:

Butterworth Filter
Advantages:
- Maximally flat magnitude response in the pass-band.
- Good all-around performance.
- Pulse response better than Chebyshev.
- Rate of attenuation better than Bessel.

Disadvantages:
Some overshoot and ringing in step response. This filter has the flattest possible pass-band magnitude response. Attenuation is -3dB at the design cutoff frequency. Attenuation beyond the cutoff frequency is a moderately steep -20dB/decade/pole [9]. The pulse response of the Butterworth filter has moderate overshoot and ringing.

Chebyshev Filter
Advantages:
- Better rate of attenuation beyond the pass-band than Butterworth.

Disadvantages:
- Ripple in pass-band. Considerably more ringing in step response than Butterworth.

This filter response has the steeper initial rate of attenuation beyond the cutoff frequency than Butterworth. This advantage comes at the penalty of amplitude variation (ripple) in the pass-band. Unlike Butterworth and Bessel response, which have 3dB attenuation at the cutoff frequency, Cebyshev cutoff frequency is defined as the frequency at which the response falls below the ripple band [3].

Inverse Chebyshev Filter
Advantages:
- Flat magnitude response in pass-band with steep rate of attenuation in transition-band.

Disadvantages:
- Ripple in stop-band. Some overshoot and ringing in step response.

The Inverse Chebyshev filter is confined to the stop-band. This filter type has a steep rate of roll-off and a flat magnitude response in pass-band. Cutoff of the Inverse Chebyshev is defined as the frequency where the response first enters the specified stop-band. Step response of the Inverse Chebyshev is similar to the Butterworth.

It has been found that the Butterworth filter is the best compromise between attenuation and phase response [3]. It has no ripple in the pass band or the stop band. The speech signals have

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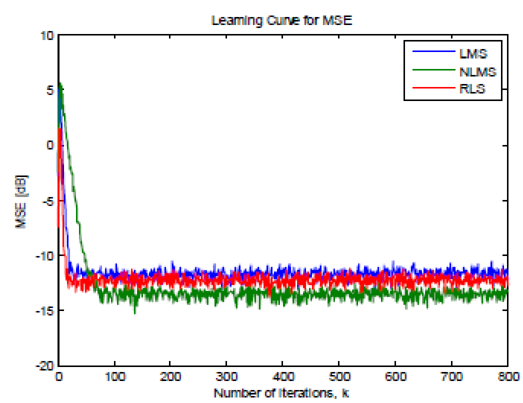
also been encountered using simulation, which was the special consideration, and compared the input and output spectrum of the signal.

3.2 Best Methods Applied

Based on the properties of method apply to the filter in reducing noise, the best method that has been verified is using Hidden Markov Model – Recursive Least Square Algorithm (HMM-RLS) followed by Discrete Cosine Transformation–Least Mean Square (DCT-LMS). RLS and LMS are both from the Adaptive filter.

Recursive Least Squares (RLS) algorithm is capable of realizing a rate of convergence that is much faster than the LMS algorithm, because the RLS algorithm utilizes all the information contained in the input data from the start of the adaptation up to the present.

There are some criteria performances that have to be analyzed to show that the RLS algorithm in Adaptive Filter is the best method in filtering noise. The first criteria using the filter in minimize the Mean Square Error (MSE). The performance of RLS for MSE is compared to LMS. The graph below shows the MSE curve outcome when using RLS and LMS.



0

Figure 18: comparison MSE curve between LMS and RLS algorithms.

From the plots it is clear that the RLS achieve faster initial convergence speed than LMS in comparison of MSE. RLS algorithms although converges faster but it is computationally more complex as matrix inversion is involved. The results show that the RLS algorithm outperforms the LMS algorithm in terms of convergence rate, MSE and the learning behavior.

Table 2: The performance comparison of LMS and RLS

Algorithms	MSE	Complexity	Stability
LMS	1.5*10^{-2}	2N+1	Less stable
RLS	6.2*10^{-3}	$4N^2$	High stable

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From the table, it is shown that the performance of RLS is high as compared to LMS due to less MSE. LMS also show less stability compared to RLS. The second performance criteria in comparing the RLS and LMS are the algorithm execution time and the required filter order based on signal to noise ratio (SNR) improvement.

Table 3: SNR Improvement in DB

Noise variance	Sampling rate (kHz)	SNR improvement (Db) LMS	SNR improvement (Db) RLS
0.02	1.5	9.85	9.91
0.05	1.5	7.55	8.89
0.10	1.5	5.12	7.02

The SNR improvement of LMS and RLS adaptive algorithm shown in Table 3 at 1.5 kHz sampling rate and different noise variance, according to the table the RLS adaptive algorithm has improved SNR in dB. It concludes that the best adaptive algorithm is Recursive Least Square according to the SNR improvement table and graph of MSE.

3.3 Adaptive Filter

Adaptive filter is compatible for unknown environment so it plays an important role in noise cancellation for full digital audio amplifier. There are many applications of adaptive filtering, for examples: Adaptive Delta Sigma Modulator, Subband Adaptive Filter (SAF), Adaptive Filter Length selection and Adaptive Line Enhancer. Different types of adaptive filters are evolved as per requirement of applications. With the help of above theoretical explanation, it can be said that Adaptive Delta Sigma Modulator performs well in most of the applications. It performs quantization operation within a feedback loop, which computes the difference between the instantaneous input signal and the quantized output, followed by an integrator that can reduce the quantization noise in the frequency region of audio amplifier and it is easy to implement. Thus it fulfills almost all needs of noise cancellation for audio amplifiers against other methods by above description.

3.4 Latest design of audio filters

Based on all different types of audio filter that discussed above, one of the best audio filter that is best in performance, easier to use and the most important is suitable for modern communication systems is High-Temperature Superconducting Low-Pass Filter for Broad-Band Harmonic Rejection.

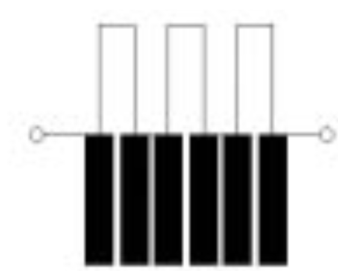

Figure 19: The schematics of the proposed seventh-order low-pass filter

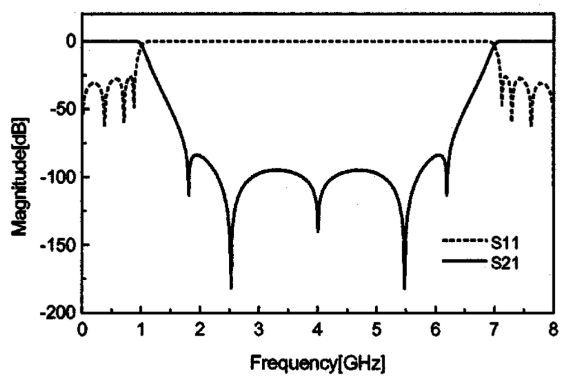

Figure 20: Simulation result of the seventh-order low pass filter by circuit simulator.

This new low-pass filter suppressed the harmonic signal using attenuation poles in the stopband. Its cut-off frequency is 1 GHz with a 0.01dB ripple level. The electrical length of the coupled line has been chosen to be 90 at 4 GHz. Five attenuation poles were located at 1.8, 2.5, 4, 5.5, and 6.2 GHz. This filter also has five symmetrical attenuation poles in the stopband, so it has an improved attenuation characteristic up to seven times that of the cut-off frequency.

The symmetry of attenuation poles as shown in figure above is broken because transmission lines are bended in real device. The fabricated filter exhibits a characteristic stop band in the range from 1 to 7 GHz. Changing the measurement temperature from 77 K to 30 K does not affect microwave properties of the HTS filter. The fabricated low-pass filter exhibits an improved attenuation characteristic up to seven times the cut-off frequency.

4.0 CONCLUSION

This paper has provided an overview and comparisons between various types of audio filter design. Many contributions have already been proposed, and a number of techniques are identified. It is necessary to choose a suitable frequency range in order to design the filters like low pass and high pass filters. The best filter for the IIR classic filter is the Butterworth filter and the best method applied to filter design is by using Hidden Markov Model – Recursive Least Square Algorithm (HMM-RLS). For adaptive filters, the Adaptive Delta Sigma Modulator performs the best quantization operation and the most suitable filter for modern systems is High-Temperature Superconducting Low-Pass Filter. Therefore, the best filter which is suitable for the noise filtering is the adaptive filter. Based on the result, is it proven that adaptive filter can be used with different method and different algorithm to improve the audio signal and removing noise. However, a detailed discussion about other filter types is beyond the scope of this paper.

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