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Comprehensive study on effectiveness of various Channel Equalization techniques in retrieval of audio signal over a noisy environment

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Abstract: In this paper, an effort has been made to make a performance simulative study on judging the robustness of various channel equalization techniques on transmission of audio signal in a noisy SISO channel. Three channel equalization techniques such as Ideal Inverse Channel Filtering, Tikhonov Regularization based Wiener Equalization and Blind Equalization Using Channel Input Power Spectrum have been implemented. With consideration of time varying characteristics of both input signal and the channel in MATLAB based simulation study, it has been explored that the Tikhonov Regularization aided Wiener Equalization scheme is very much effective to retrieve A/D converted audio signal in spite of signal distortion incurred by the channel and noise contamination

Keywords: Ideal Inverse Channel Filtering, Tikhonov Regularization based Wiener Equalization and Blind Equalization Using Channel Input Power Spectrum, noisy channel, AWGN and SNR

I. Introduction

In real-time system, signal processing is a most important component concerned with the analysis and modeling of signals, extraction of information from signals, classification and recognition of patterns, synthesis and morphing of signals viz. creation of a new signal out of existing samples. In real-time signal processing system, it is noticeable that the observation signals are often distorted, incomplete and noisy. The noise reduction and the removal of channel distortion and interference are treated as important challenging tasks in signal processing system(s). The digital signal processing methodologies are widely adopted in hi-fi audio, TV and radio, cellular mobile phones, voice recognition, vision, antenna-arrays, radar, sonar, geophysical exploration, medical electronics, bio-medical signal processing, adaptive noise reduction, channel equalization, pattern classification/recognition, audio signal coding etc. [1].

Channel Equalization is a powerful digital signal processing scheme used at the receiver of a typically assumed SISO (Single input-single output)/MISO (Multiple input-single output)/MIMO (Multiple input-multiple output) system. Most equalizers are implemented digitally after A/D conversion. In digital signal processing, any kind of information are represented as digital symbols and Channel equalization alleviates inter symbol interference (ISI) and designed typically to balance ISI mitigation with noise enhancement as both the signal and the noise pass through the equalizer [2,3].

II. Channel Equalization schemes

In my present study, three channel equalization schemes have been used. A brief overview of these schemes is given below:

A. Ideal Inverse Channel Filtering

The fundamental concepts behind the Ideal Inverse Channel Filtering aided channel equalization scheme implementation can be demonstrated using a simplified signal model in terms of transmitted signal vector X , independent and identically distributed (i.i.d.) additive white Gaussian Noise V , rapidly varying channel coefficient matrix H and received signal vector Z as:

$$Z = HX + V \quad (1)$$

In Frequency domain, the Equation (1) can be written as:

$$Z(f) = X(f)H(f) + V(f) \quad (2)$$

where, $Z(f)$, $X(f)$, $H(f)$ and $V(f)$ are the frequency spectra of the channel output, the channel input, the channel response and the additive noise respectively. In inverse-channel filtering technique, an effort is taken to recover the

original input from the channel output signal. In the frequency domain, the ideal inverse channel filter can be expressed as

$$H(f) H^{\text{inv}}(f) = 1 \quad (3)$$

In Equation (3), $H^{\text{inv}}(f)$ is used to denote the inverse channel filter. For the ideal equalizer, we have,

$$H^{\text{inv}}(f) = H^{-1}(f) \text{ or expressed in Log frequency domain, } \ln H^{\text{inv}}(f) = -\ln(H(f)) \text{ which can further be written as: } H^{\text{inv}}(f) = \exp(-\ln(H(f))) \quad (4)$$

Using the inverse channel filter from Equation (4), the recovered channel input signal can be written in the frequency domain as

$$\tilde{X}(f) = H^{\text{inv}}(f) Z(f) \quad (5)$$

In signal detection/channel equalization technique, Inverse Fourier Transformation of Equation (4) is taken to retrieve the transmitted signal vector $\hat{X}$ as:

$$\hat{X} = \text{IFFT} [H^{\text{inv}}(f) Z(f)] \quad (6)$$

B. Tikhonov Regularization based Wiener Equalization

In Wiener Equalizer aided signal detection scheme, the autocorrelation matrix of the channel output can be obtained from Equation (1) as

$$R_{zz} = E[Z Z^T] = H R_{xx} H^T + R_{vv} \quad (7)$$

The cross-correlation matrix R_{xz} between the input and output signals is given by

$$R_{xz} = E[X Z^T] = H R_{xx} \quad (8)$$

where, $E[\cdot]$ is the expectation operator, R_{xx} and R_{vv} are the autocorrelation matrices of the input signal and noise respectively. As noise vector V is uncorrelated with signal vector X , additional terms in both Equation (6) and (7) have been omitted.

In many cases, it is observable that encountered ill-conditioning during matrix inversion in addition to noise contamination interrupts from producing exact solution of Least Square problem. Such problem can be alleviated using Tikhonov Regularization parameter and singular value decomposition (SVD) scheme. On singular value decomposition of matrix R_{zz} , we get three matrices U , Σ and V in its transposed form.

$$R_{zz} = U \Sigma V^T \quad (9)$$

The Tikhonov Regularization parameter α is selected as 0.1 times the maximum singular value of the pseudo-diagonal matrix Σ .

The Wiener Equalizer inverse filter using the Tikhonov Regularization parameter and identity matrix I can be written as:

$$\hat{h}^{\text{inv}} = R_{xz} R_{zz}^{-1} = H R_{xx} (H R_{xx} H^T + R_{vv} + \alpha I)^{-1} \quad (10)$$

The estimated transmitted signal vector $\hat{X}$ can be written as

$$\hat{X} = \hat{h}^{\text{inv}} Z \quad (11)$$

In MATLAB notation using for end loop with specification of 125 iterations, the estimated Wiener Equalizer inverse filter ($\hat{h}^{\text{inv}}$) can be written as:

```
for kkk=1:125
Rzz(:, :, kkk)=Z(:, kkk)*Z(:, kkk)';
Rxz(:, :, kkk)=X(:, kkk)*Z(:, kkk)';
[S(:, :, kkk) V(:, :, kkk) D(:, :, kkk)] = svd(Rzz(:, :, kkk));
singular_value(:, :, kkk)=(diag(V(:, :, kkk)));
alpha(kkk)=max(singular_value(:, :, kkk));
Rzzinv(:, :, kkk)= inv(Rzz(:, :, kkk)+0.1*alpha(kkk)*eye(1024));
Hhatinv(:, kkk)= Rxz(:, :, kkk)* Rzzinv(:, :, kkk)
end
```

C. Blind Equalization Using Channel Input Power Spectrum

Prior to the development of the conceptual idea on Blind channel Equalization scheme using Channel Input Power Spectrum, we are in need of taking multiplication of Equation (2) with their complex conjugates and also taking the expectation

$$E[Z(f)Z^*(f)] = E[(X(f)H(f) + V(f))(X(f)H(f) + V^*(f))] \quad (12)$$

Assuming the signal $X(f)$ and noise $V(f)$ are uncorrelated, Equation (12) becomes

$$P_{zz}(f) = P_{xx}(f)|H(f)|^2 + P_{vv}(f) \quad (13)$$

where, P_{zz} , $P_{xx}(f)$ and $P_{vv}(f)$ are the power spectra of the distorted signal, the original audio signal and the noise respectively. From Equation (13), an estimate of the spectrum of the channel response can be obtained as

$$|H(f)|^2 = H(f)H^*(f) = \frac{P_{zz}(f) - P_{vv}(f)}{P_{xx}(f)} \quad (14)$$

From Equation (14), the estimated inverse filter $\hat{h}^{inv}(f)$ in frequency domain is

$$\hat{h}^{inv}(f) = \frac{1}{H(f)} = \frac{P_{xx}(f)H^*(f)}{P_{zz}(f) - P_{vv}(f)} \quad (15)$$

Using the inverse channel filter from Equation (15), the recovered channel input signal can be written in the frequency domain as

$$\hat{X}(f) = \hat{h}^{inv}(f)Z(f) \quad (16)$$

Taking Inverse Fourier Transformation of Equation (16), the retrieved transmitted signal vector $\hat{X}$ can be written as [1,4]:

$$\hat{X} = \text{IFFT}[\hat{h}^{inv}(f)Z(f)] \quad (17)$$

III. System Description and Signal processing

A comprehensive block diagram to explain the present simulation study is depicted in Figure 1.

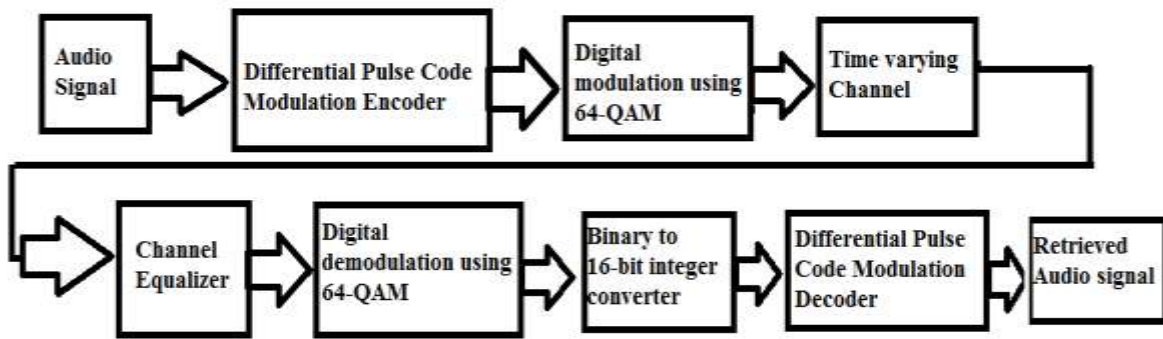


Figure 1: A simplified block diagram to illustrate various signal processing schemes

A segment of digitally recorded audio signal of 1 second duration is processed for analysis. In Differential pulse code modulation encoding, the sampled analog values of audio signal are converted into corresponding integer values under consideration of 65536 quantization levels. The extracted binary bits in 0/1 format are modulated using 64-QAM digital modulation scheme [3]. It is rearranged into transmitted signal vector $X \in C^{1024 \times 125}$. The noise V , Channel H and received signal vectors Z are also $\in C^{1024 \times 125}$. In Equalization section, signal distortion incurred by the channel is removed and the channel equalized signal is undergone to digital demodulation. The digitally demodulated binary bits are blocked into 16 bits and each block is converted into 16-bits integer and

eventually, all the integers are fed into Differential pulse code modulation decoding to retrieve sampled values forming the desired signal.

IV. Results and Discussion

In this section, simulation results are presented to illustrate the significant impact of various channel equalization schemes on effective retrieval of audio signal passing through a noisy channel. The simulation study has been made using MATLAB 2014a based on the parameters given in Table 1.

Data Type	Audio signal
No. of samples	48000
Sampling Frequency (Hz)	48000
No of binary bits for a single sample	16
Total number of binary bits from audio samples	768000
Digital modulation	64-QAM
No. of digitally modulated symbols	128000
Noise type	AWGN
Signal to Noise-Ratio, SNR in dB	0,4,8,12,16 and 20
Channel Equalization	Ideal Inverse Channel Filtering, Tikhonov Regularization based Wiener Equalization and Blind Equalization Using Channel Input Power Spectrum.

Table 1: Summary of the Simulated Model Parameters

It is quite noticeable from graphical illustrations presented in Figure 2 through Figure-4 that such results are clearly indicative of showing distinguishable difference between system performance under various channel equalization schemes. In perspective of spectral representation, the amplitude of different significant frequency components for transmitted and retrieved audio signals are shown in Figure 5 through Figure-7. In Figure 2 with Inverse-Channel Filtering aided signal detection technique implementation, the transmitted audio signal and its various retrieved forms processed in noisy channel at three SNR values (4dB, 8dB and 12 dB) are presented. It is observable that the detected audio signal at 4 dB is highly contaminated with noise and is not recognizable. The estimated number of erroneous samples values (2.18%) in term of analog to digitally converted integer values is found to be 1045 out of 48000 samples. In 8 dB, the quality of the retrieved audio signal is improved and the number of erroneous samples (0.74%) reduces from 1045 to 357. In 12dB SNR value, the scenario of the effectiveness of the signal detection technique is totally changed viz. all the 48000 samples have been detected correctly. In Figure 3 with Power spectrum based channel equalization technique implementation, the estimated number of erroneous samples values is 4.2% at 12 dB SNR value (Erroneous sample: 2016). In 16 dB SNR value, The estimated number of erroneous samples values reduces from 4.2% to 2.16% (Erroneous sample: 1037). At very high SNR value of 20 dB, the transmitted audio signal is correctly retrieved. In Figure 4, it is ratified that the Wiener Equalizer aided signal detection technique using Tikhonov Regularization parameter is very effective and robust in retrieving data at various SNR values. In all cases, all the transmitted samples have been detected correctly. It is observable from spectral curves of Figure 5 under a scenario of Inverse-Channel Filtering aided signal detection technique implementation that the high frequency components are significantly attenuated in a specific case where the signal power is greater than noise power by 4 dB. The significant frequency components of the transmitted audio signal is found to have concentrated around the frequency range of 0 Hz to 50 Hz. At 8dB SNR, the rate of heavily high frequency component attenuation reduces showing a significant frequency component in the range of 0 Hz to 4000Hz. On crucial observation, it seen that the transmitted audio signal contains significant frequency components in the range of 0 Hz to approximately 8000Hz. In case of Power spectrum based blind channel equalization scheme implementation, the spectral curves (Figure 6) shows that such scheme is not suitable for effective audio signal retrieving. In Figure 6, it is quite obvious that high frequency components are significantly attenuated at considerable SNR values of 12 dB and 16 dB. In Figure 7, unique spectral response is observable in case of Tikhonov Regularization based Wiener Equalization scheme implementation.

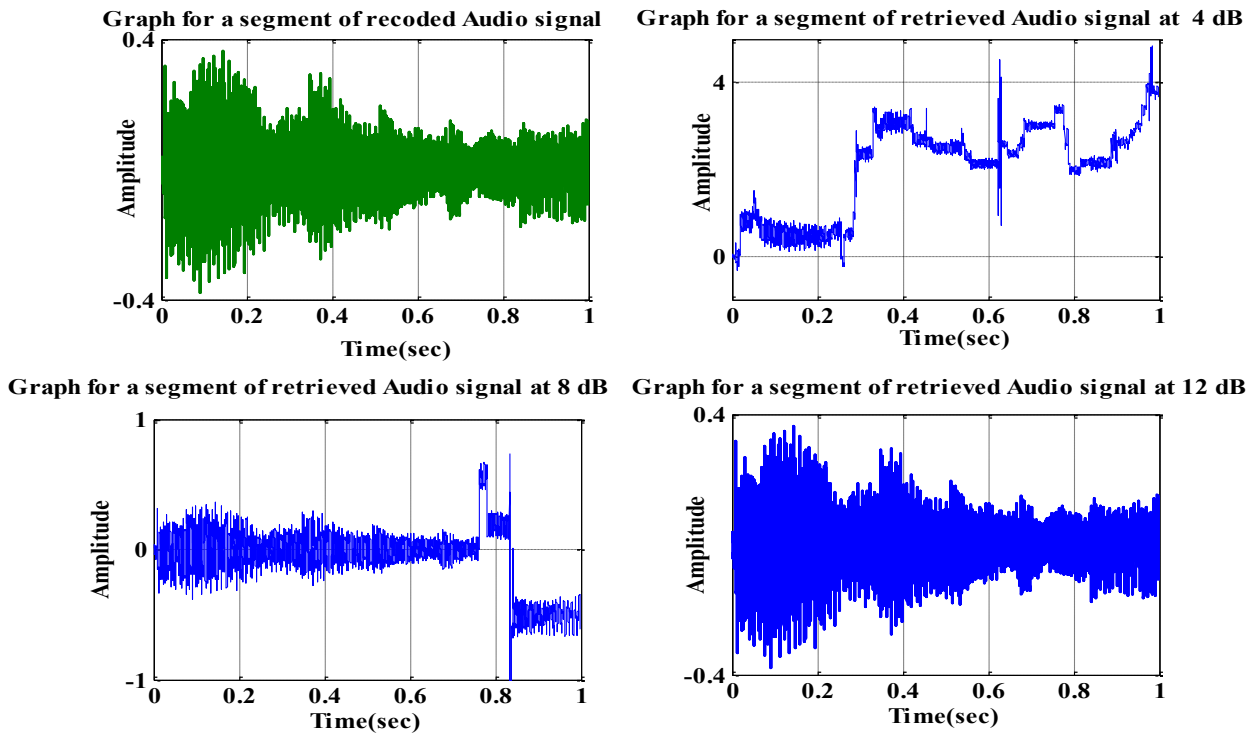


Figure 2: Transmitted and Retrieved audio signals at different values of Signal to Noise power (SNR) ratio under implementation of Inverse-Channel Filtering aided signal detection technique.

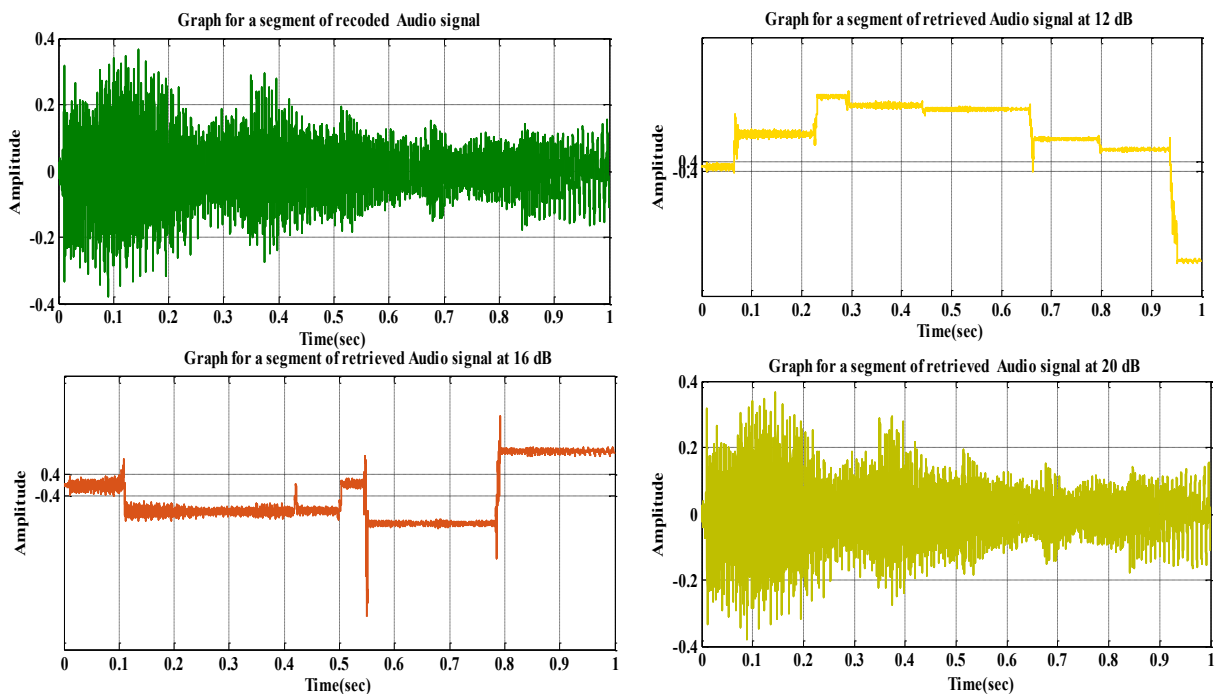


Figure 3: Transmitted and Retrieved audio signals at different values of Signal to Noise power (SNR) ratio under implementation of Power spectrum based blind channel equalization technique.

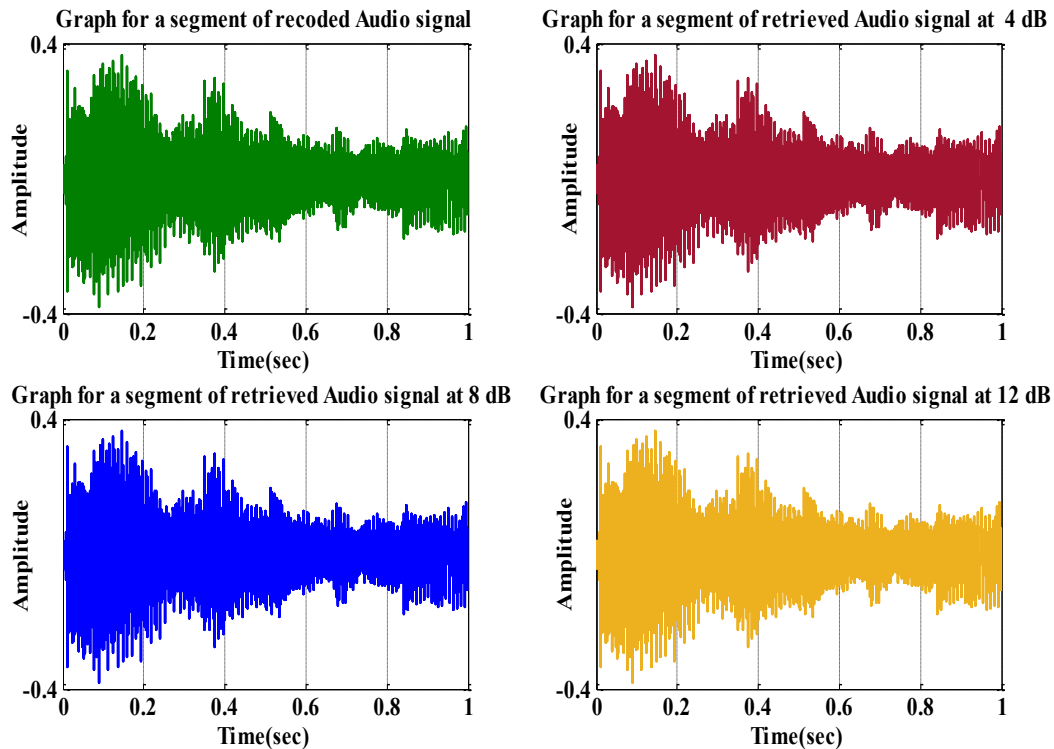


Figure 4: Transmitted and Retrieved audio signals at different values of Signal to Noise power (SNR) ratio under implementation of Wiener Equalizer aided signal detection aided signal detection technique.

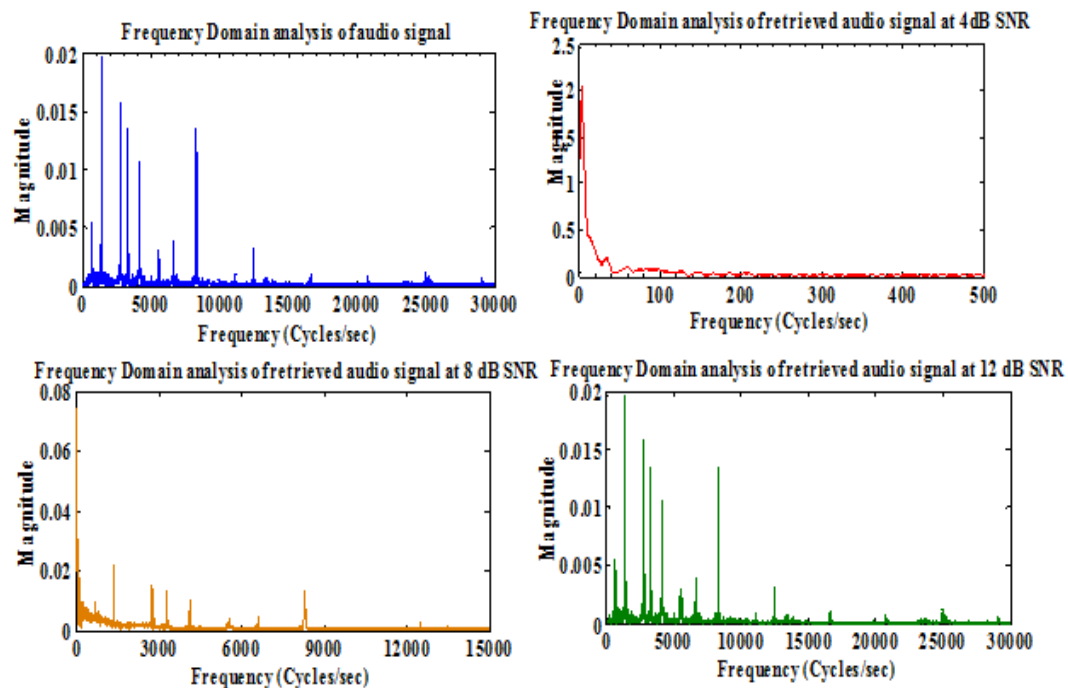


Figure 5: Spectral analysis of Transmitted and Retrieved audio signals at different values of Signal to Noise power (SNR) ratio under implementation of Inverse-Channel Filtering aided signal detection technique.

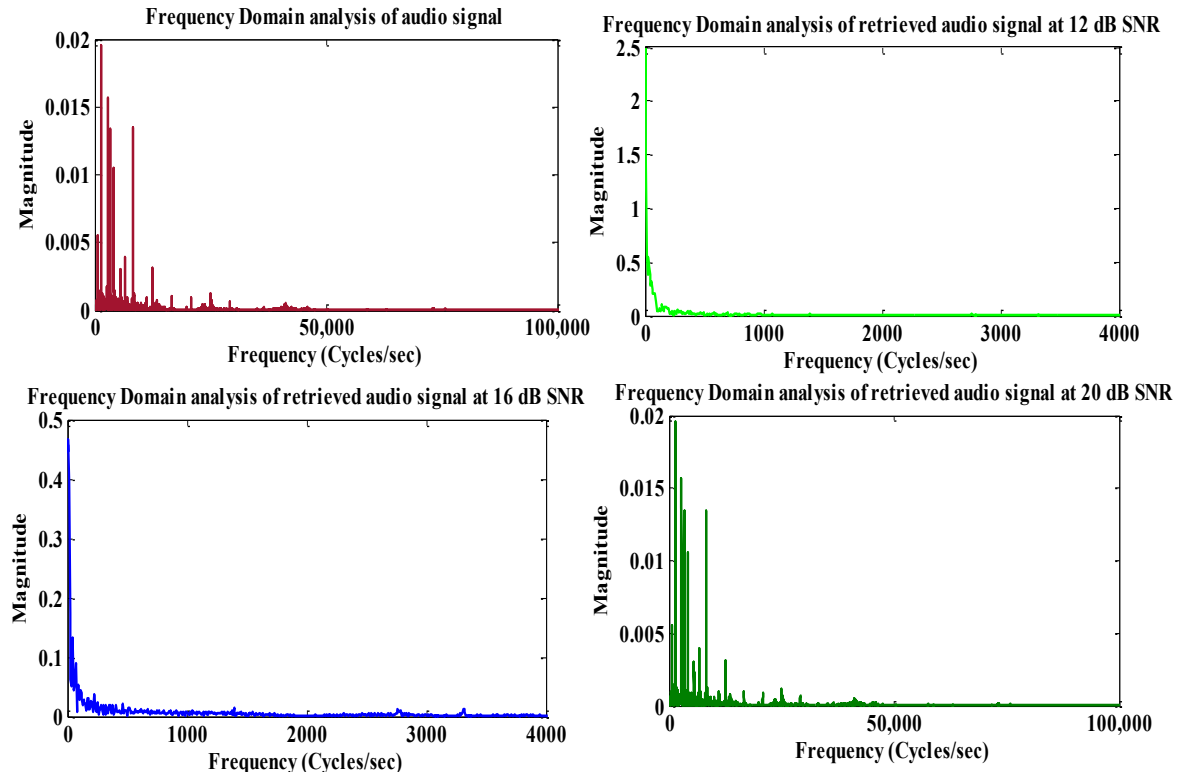


Figure 6: Spectral analysis of Transmitted and Retrieved audio signals at different values of Signal to Noise power (SNR) ratio under implementation power spectrum based blind channel equalization.

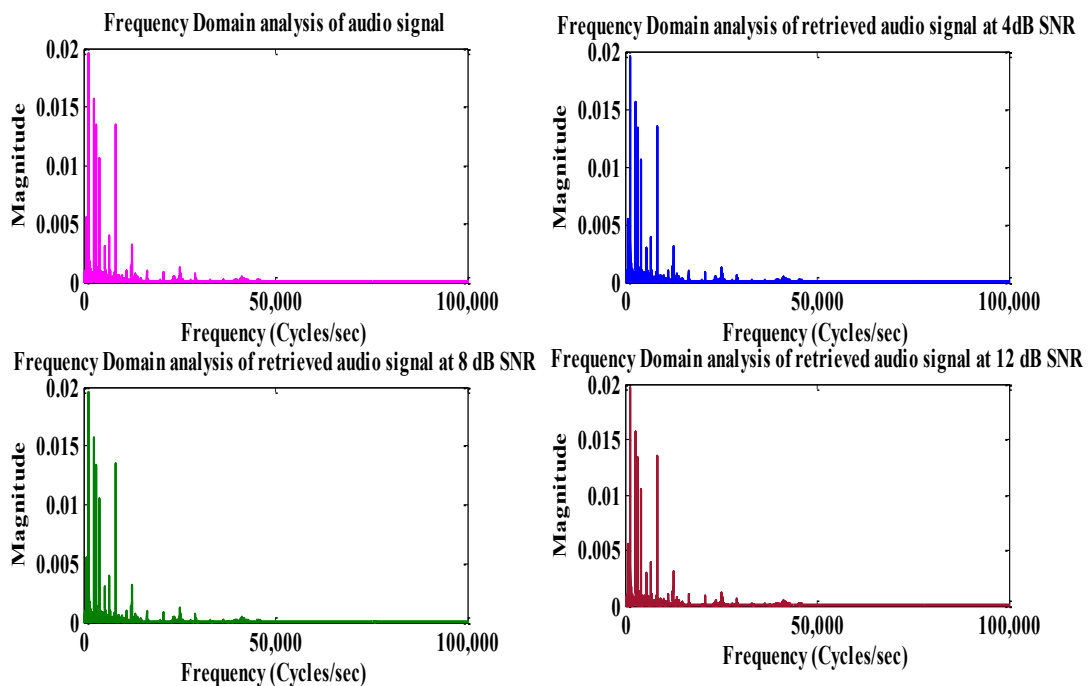


Figure 7: Spectral analysis of Transmitted and Retrieved audio signals at different values of Signal to Noise power (SNR) ratio under implementation Wiener Equalizer aided signal detection aided signal detection technique.

V. Conclusions

In this paper, various digital signal processing technique oriented Channel Equalization schemes have been implemented merely to observe critically the quality of the transmitted audio signal in a noisy channel. Simulation results indicate that the Tikhonov Regularization aided Wiener Equalization scheme is very much effective in significant reduction of distortion inferred from the time varying channel. Additionally, Insertion of Tikhonov Regularization parameter has successfully eliminated the ill-conditioning problem during matrix inversion. However, Based on the simulation results, it can be concluded that Tikhonov Regularization aided Wiener channel Equalization scheme is very much robust and reliable in retrieving data in noisy channel.

V1. References

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