# ELIMINATION OF BURST NOISE USING MODIFIED CHEBYSHEV FIR FILTERS AND ITS GUI WITH MATLAB

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***Abstract:*** **In this manuscript we discuss the implementation and use of Modified Chabyshev FIR filter for filtering speech signals consisting of burst or continuous pitch noise, a phenomenon which takes place when we transmit speech over a channel. The Chebyshev filter is designed in such a way that in removes the burst noise completely.**

***Key-Words*****: FIR Filter, Speech Processing, Noise Reduction, Tschebyshev FIR Filter, White Noise.**

## I. INTRODUCTION

Speech filtering is one of the classical topics of speech signal processing and several methods have been suggested as well as employed. The filters used for this are mainly divided into two categories viz. analog filters and digital Filters. The digital filters have many advantages over traditional analog filters such as accuracy, ease of design. For speech filtering FIR is most commonly used due to the advantages such as linear phase and guaranteed stability.

Noises are of to two types of noise. One is the continuous noise. It is added to the whole speech and does not change so radically. The other is the burst noise. It is characterized by the large occasional burst of energy. Continuous noise can be easily estimated comparing to burst noise. Noise robust LPC analysis [1], [2], Hidden Markov Model (HMM) decomposition and composition [3], [4], and the extraction of dynamic cepstrum, [5] etc. can be used to retrieve the original speech signal. In real environment, burst noise when combined with continuous noise and added into input speech seriously degrades the speech, and it is next to impossible to recognize it. The prediction of burst noise is so difficult that the extraction of speech components from burst noise periods is very difficult.

In this article we implement the Modified Chabyshev FIR filter [6], [7] on real speech signal. Modified Chabyshev FIR filter characteristics discussed by authors [6], [7] makes this filter really useful for eliminating burst noise off line while maintaining the phase exactly linear for speech signals. In case of continuous noise the filter can be used recursively to eliminate it. We will implement our design for both types of noises.

## II. CHABYSHEV FILTER DESIGN

Taking the case of a linear equispaced antenna array with n elements of equal amplitude and spacing, labeled from left to right where n is a positive integer. The total field 'E' at a large distance in the direction $\varnothing$ is given as

$$E=1+e^{j} + e^{j2\psi} +.........+e^{j(n-2)\psi} + e^{j(n-1)\psi} \qquad (1.\ 1)$$

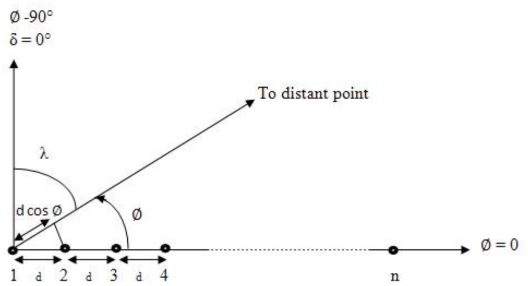


Figure 1 : Linear Arrays Of n Isotropic Point Source

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Where ψ the total phase difference of the fields from adjacent sources as given by

$$\psi = \frac{2\pi d}{\lambda} \cos \varnothing + \delta = kd \cos\theta + \delta \quad \text{where} \qquad (1.2)$$

$$k = \frac{2\pi}{\lambda}$$

Where $\delta$ is the phase difference of adjacent sources; i.e. source 2 with respect to 1 and 3 with respect to 2 etc. Considering the case of a linear equispaced antenna array with n elements with different amplitude, labeled from left to right.

$$|E| = |A_0 e^{j0} + A_1 e^{j\psi} + A_2 e^{j2\psi} + \dots\dots\dots + A_{n-2} e^{j(n-2)\psi} + A_{n-1} e^{j(n-1)\psi}| \qquad (1.3)$$

Where ψ the total phase difference of the fields from adjacent sources as given by

$$\psi = \frac{2\pi d}{\lambda} \cos \varnothing + \delta = kd \cos\theta + \delta$$

Where $|E|$ is the magnitude of the far field , k = 2**π** / λ ,$\lambda$ is the free space wavelength, $d$ is the spacing between elements,$\varnothing$ is the angle from the normal to the linear array, $\delta$ is the progressive phase shift from left to right, and $A_0, A_1, A_2$,..... are complex amplitudes which are proportional to the current amplitudes.

**Dolph-Chabyshev Array Design:** Taylor developed an aperture distribution based on Dolph's use of the Tschebyshev polynomials to produce the narrowest beamwidth for a specified sidelobe level for an array. The Tschebyshev array design produces equal-amplitude sidelobes that we discover to be undesirable for large arrays because the equivalent aperture distribution peaks at the ends and the average value of the sidelobes limits the directivity to 3 dB above the sidelobe level. Taylor used the zeros of the Tschebyshev array to alter the positions of the inner nulls of the uniform distribution to lower sidelobe levels.

$$T_m(x) = \cos(m\cos^{-1}x) \quad 0 < |x| < 1$$
$$T_m(x) = \cos(m\cos^{-1}x) \quad 1 < |x| \qquad (1.4)$$

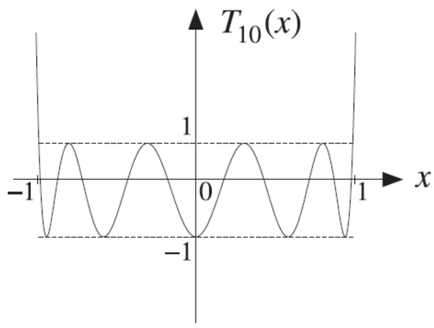


Figure 2 : Tschebyshev polynomials of order ten

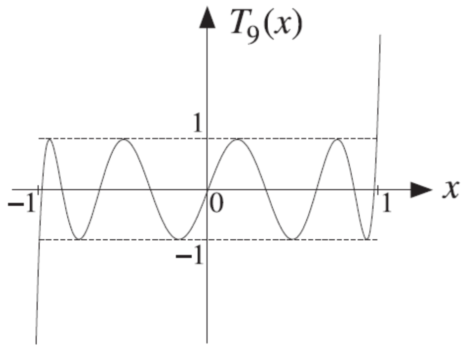


Figure 3 : Tschebyshev polynomials of order nine

Consider the Chabyshev polynomial of m[th] degree

$T_m(x) = \cos(m \cos^{-1} x) = \cos(m\delta)$

where $\cos\delta = x$

The nulls of the pattern are given by the roots $\cos(m\delta) = 0$

That is by $\delta_k^0 = \frac{(2k-1)\pi}{2m}$ k=1,2,........m

Now consider the function $\psi$, for a broadside array $\psi = \beta d \cos \varnothing$

As $\varnothing$ varies from $0$ to $\pi/2$ to π, ψ goes from βd to 0 to – βd and the range of $\psi$ is $2\beta d$. Now let $x = x_0 \cos \psi/2$ . Then as $\varnothing$ varies from $0$ through $\pi/2$ to $\pi$, $\psi$ varies from $\beta d$ through $0$ to $-\beta d$, and $x$ will vary from $x_0$ cos $\pi d/\lambda$ to $x_0$ back to $x_0 \cos(-\pi d/\lambda) = x_0 \cos \pi d/\lambda$. For example, if $d=\lambda/2$, $\psi$ will range from $\pi$ through zero to $-\pi$ and $x$ will range from $0$ to $+x_0$ and back to $0$. Again if $d=\lambda$, $\psi$ will range twice around the circle from $2\pi$ through $0$ to $-2\pi$,(two major lobes) and $x$ will range from $-x_0$ to $x_0$ and back to $x_0$.

The nulls of the Tschebyshev pattern occur at values of x given by $x_k^0 = \cos\delta_k^0$. So the corresponding position for the nulls on the unit circle will be given by $x_k^0 = x_0 \cos\frac{\psi_k^0}{2}$ or

$$\psi_k^0 = 2\cos^{-1}\frac{x_k^0}{x_0} \quad (1.5)$$

$$\psi_k^0 = 2\cos^{-1}\left[\frac{\cos\delta_k^0}{x_0}\right]$$

where $\delta_k^0 = \frac{(2k-1)\pi}{2m}$ k =1,2,3,..........m (1.6)

The above equation gives the required spacing of nulls on the unit circle for a pattern whose side lobes are all equal.

Substituting $z = e^{j\psi}$ then Equaltion (1.3) becomes

$$H(z) = A_0 + A_1 z + A_2 z^2 + \ldots\ldots. + A_{n-2} z^{n-2} + A_{n-1} z^{n-1} \quad (1.4)$$

which is the equation of a FIR Filter. For the FIR filter $\psi_k^0 = \omega_m$ and $\delta_k^0 = \omega_k$ and m is the order of the filter. Thus

$$\omega_m = 2\cos^{-1}\{\cos(\frac{(2k-1)\pi}{2m})/x_0\} \quad (1.5)$$

$\delta_k^0 = \omega_k = (2k-1)\,\pi/2m$,Thus

$$\omega_m = 2\cos^{-1}\{\cos(\omega_k)/x_0\} \quad (1.6)$$

It can be calculated by noting that if $b = \cosh\rho$, then

$x_0 = \cosh(\rho/m)$

Substituting $x_0$ in equation (2.1.6) we have

$$\omega_m = 2\cos^{-1}\{\cos(\omega_k)/\cosh(\rho/m)\} \quad (1.7)$$

*and* $\rho = \cosh^{-1} b$

substituting $\rho$ in equation (1.7) the location of zeros, $\omega_m$, on unit circle can be calculated by the following equation

$$\omega_m = 2\cos^{-1}\{\cos(\omega_k)/\cosh(\cosh^{-1} b/m)\} \quad (1.8)$$

Similarly

$$\omega_s = 2\cos^{-1}[1/\{\cosh(1/m\cosh^{-1} b)\}] \quad (1.9)$$

$$\omega_p = 2\cos^{-1}\left[\frac{\cosh\{(1/m)\cosh-1(b/\sqrt{2})\}}{\cosh(1/m\cosh-1b)}\right] \quad (1.10)$$

where

m is the order of the filter

b is the absolute value of attenuation in the stop band,

$\omega_s$ is the stopband frequency,

$\omega_p$ is the passband frequency.

Using the relation $z_m = e^{j\omega_m}$ , we can write equation (1.4) as

$$H(z) = (z-z_1)(z-z_2)\ldots\ldots.(z-z_m) \quad (1.11)$$

where,

$z_1, z_2, \ldots.$ are location of zeros,

H(z) is the frequency response in z-transform domain.

Replacing z by $e^{j\omega}$ and $z_m$'s by $e^{j\omega_m}$'s in equation (1.11)

$$H(z) = (e^{j\omega} - e^{j\omega_1})(e^{j\omega} - e^{j\omega_1})\ldots.(e^{j\omega} - e^{j\omega_m}) \quad (1.12)$$

Order of the filter m=6 and b=100

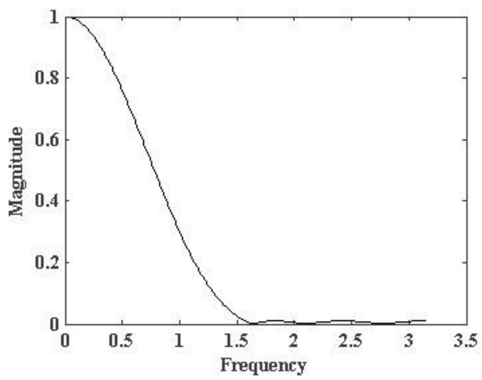


Figure 4 : Magnitude Response Of 6th order FIR filter

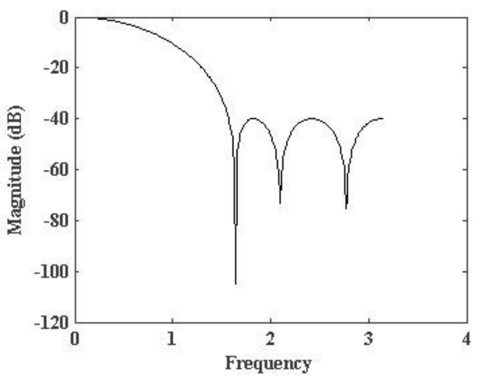

Figure 5 : Magnitude Response Of 6th order FIR filter in dB

**Limitations**: For a given order filter, we cannot change the zeroes of the filter. So we cannot change the cutoff frequency for a given order of filter.

## III. MODIFIED TSCHEBYSHEV FIR FILTER

To overcome this limitation we are introducing a new parameter '$\alpha$'. In the original Tschebyshev polynomial we will multiply a new parameter '$\alpha$' with parameter 'x'.

$$T_m(\alpha x) = \cos(m\cos^{-1} \alpha x) \qquad 0 < |x| < 1$$

$$T_m(\alpha x) = \cosh(m\cosh^{-1} \alpha x) \qquad 1 < |x| \qquad (1.13)$$

Then $\omega_s$, $\omega_m$ and $\omega_p$ becomes

$$\omega_s = 2\cos^{-1}[1/\alpha\{\cosh(1/m\cosh^{-1}b)\}] \qquad (1.14)$$

$$\omega_m = 2\cos^{-1} [\cos(\omega_k)/\{\alpha(\cosh(1/m\cosh^{-1} b))\}] \qquad (1.15)$$

$$\omega_p = 2\cos^{-1}\left[\frac{\cosh\{(1/m)\cosh-1(b/\sqrt{2})\}}{\cosh(1/m\cosh-1b)}\right] \qquad (1.16)$$

where
$\omega_k = (2k-1)\pi/2m$,
and k=1....m.
and

$$H(z) = (e^{j\omega} - e^{j\omega_1})(e^{j\omega} - e^{j\omega_1})....(e^{j\omega} - e^{j\omega_m}) \qquad (1.17)$$

Using this modified FIR filter we can change the cuttoff frequency and bandwidth of the filter.

Order of the filter m=4 and b=100

We can see that the position of zeroes change with varying the value of alpha.

## IV. APPLICATION

Now we design a Graphical User Interface (GUI) (Complete code is available at *http://spgjuit.googlepages.com/reports*) for the application of modified Chabyshev FIR filter using MATLAB®.

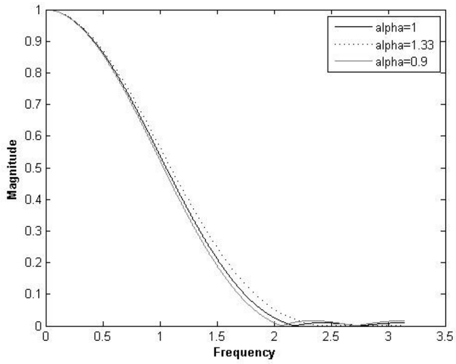


Figure 6 : Magnitude Response Of 4th order FIR filter

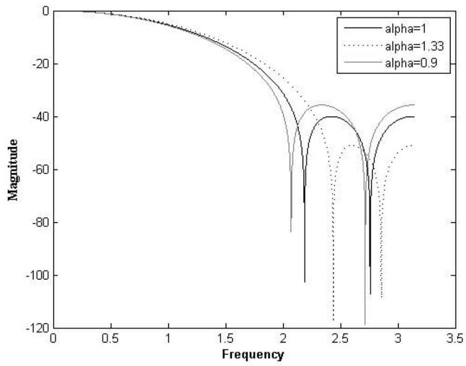


Figure 7 : Magnitude Response Of 4th order FIR filter in dB

Recording a voice**:** Done by Sound Recorder in a file of 'DSP.wav'
Length: 4.00 seconds
Data Size: 32,058 bytes
Audio Format & Sampling frequency
PCM,8 Bits, Mono, 8KHz

Adding noise to the sound file**:** High frequency sinusoidal noise 3000 Hz and
White Noise is added to the signal.
Passing Speech Signal through the FIR Filter Designed Earlier
Order of Filter :6
Location Of First Zero :3000Hz (To cancel the sinosoidal noise at 3000Hz)
Alpha :1.78
Attenuation :40dB

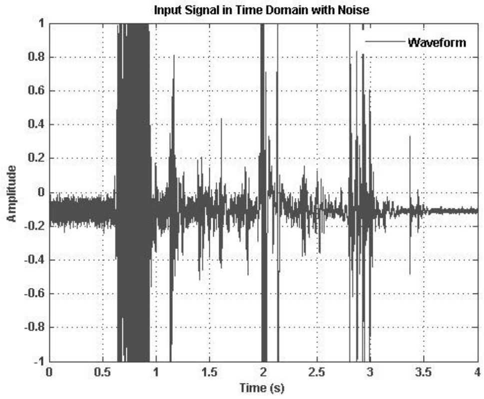


Figure 8 : Input Voice Signal in Time Domain with white noise

1.

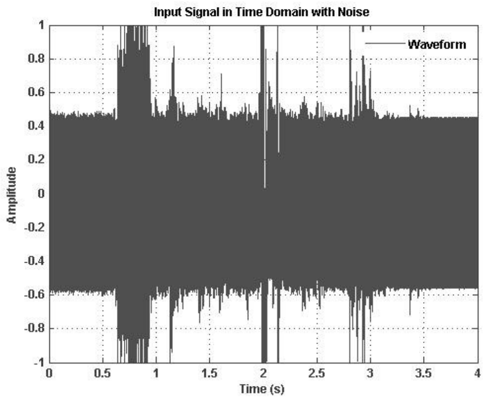


Figure 9 : Input Voice Signal in Time Domain with white noise and constant pitch noise

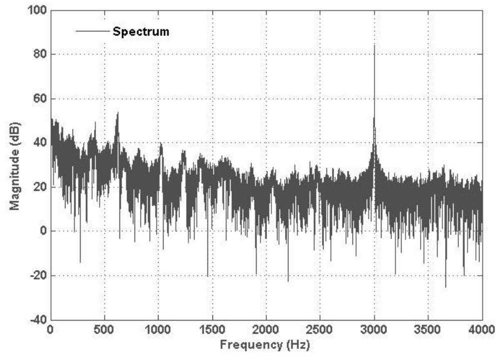


Figure 10 : Frequency Spectrum of Voice signal with noise

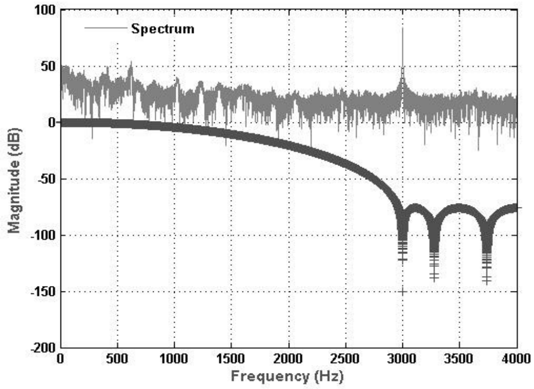


Figure 11 : Frequency Spectrum with Magnitude response of Filter

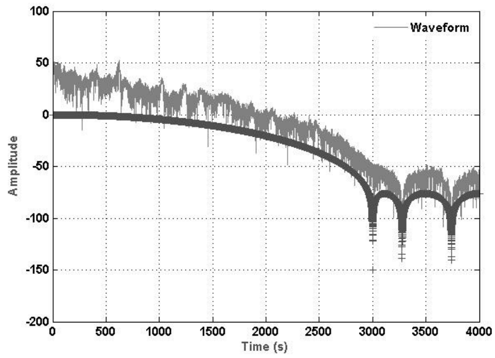


Figure 12 : Output Spectrum of signal with reduced noise

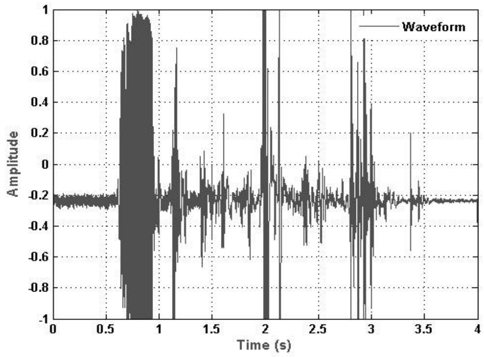


Figure 13 : Output Voice Signal with reduced noise

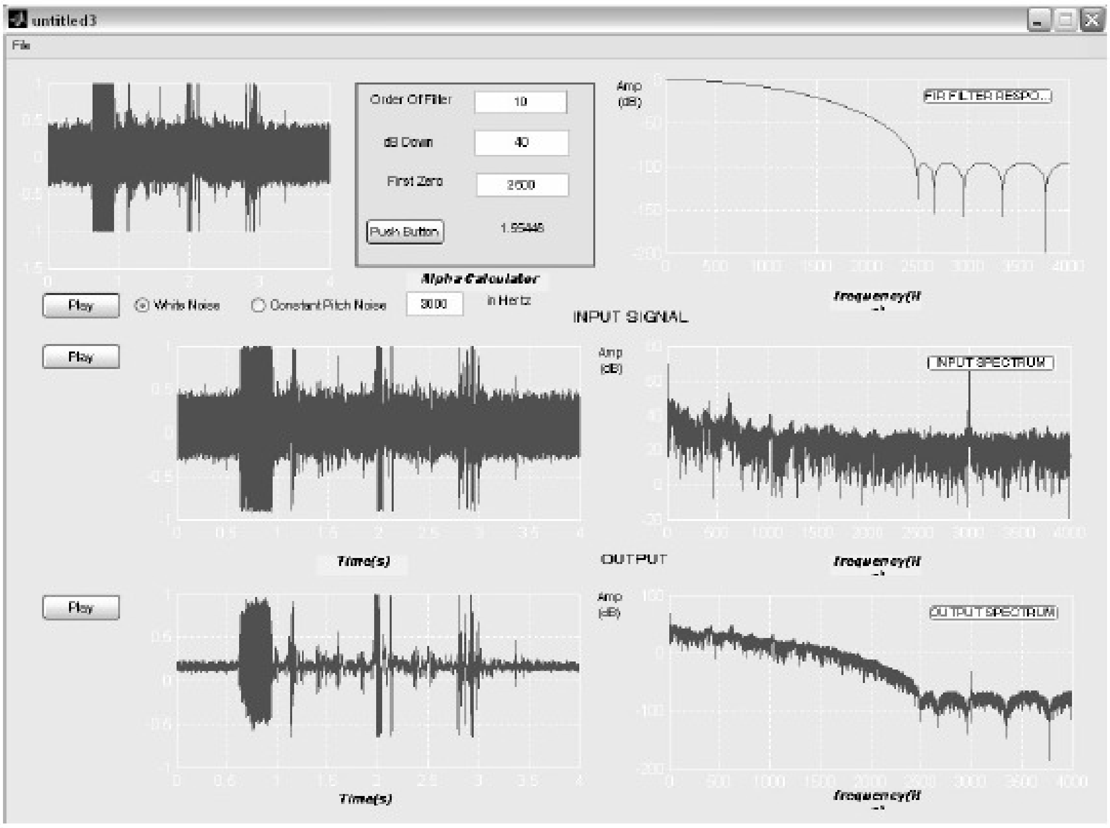


Figure 14 : Partial Elimination of Burst Noise

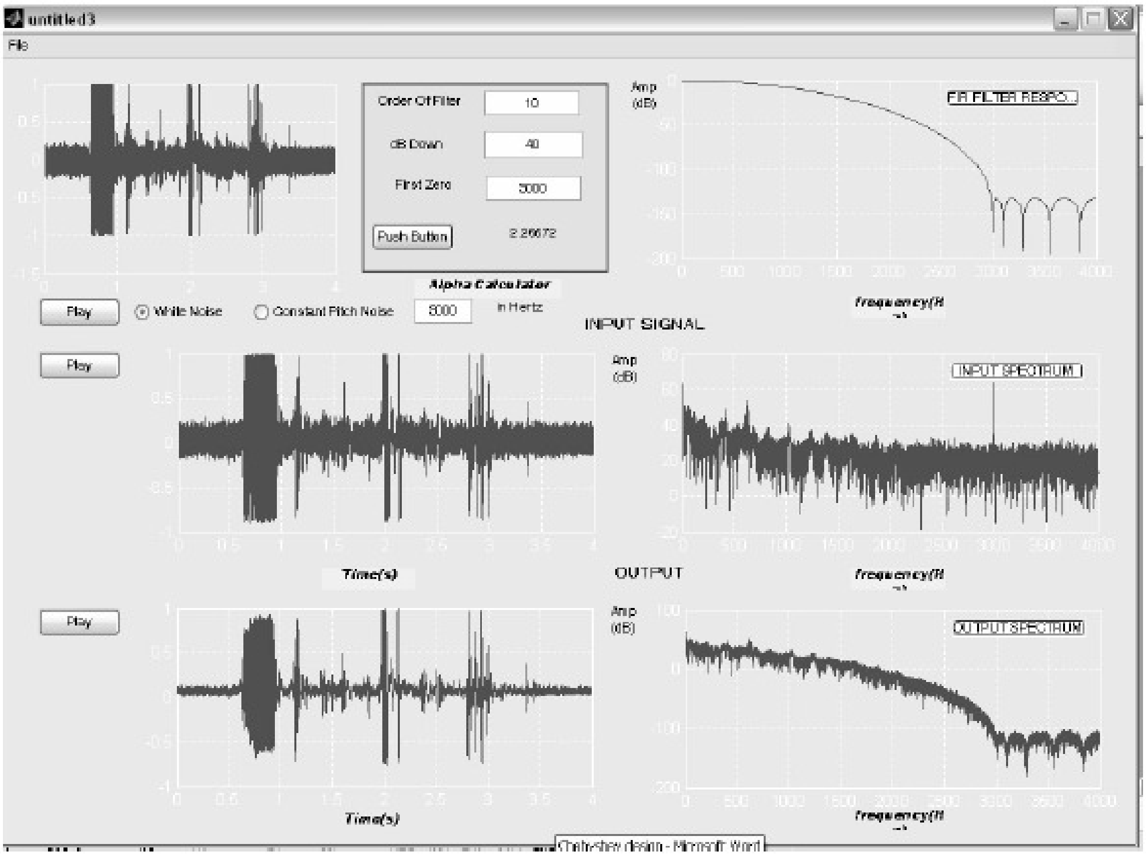


Figure 14 : Partial Elimination of Burst Noise

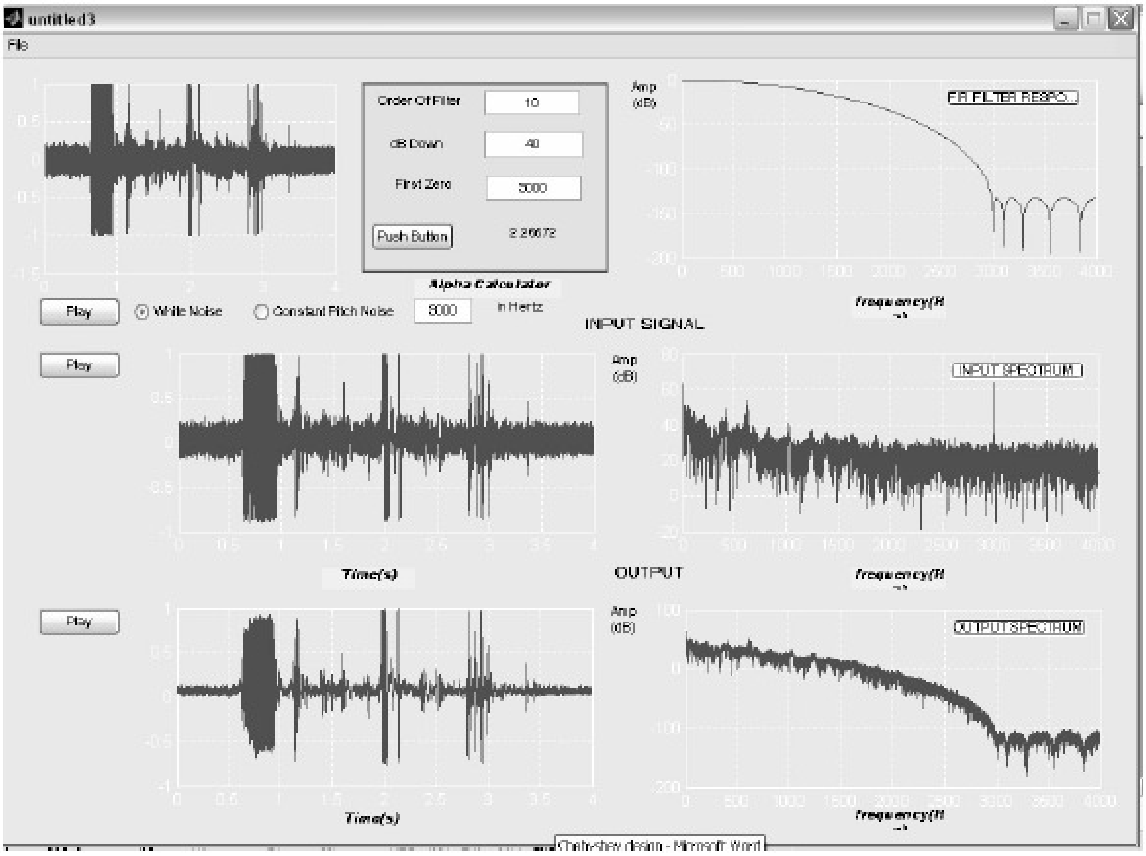


Figure 15 : Complete Elimination of Burst Noise

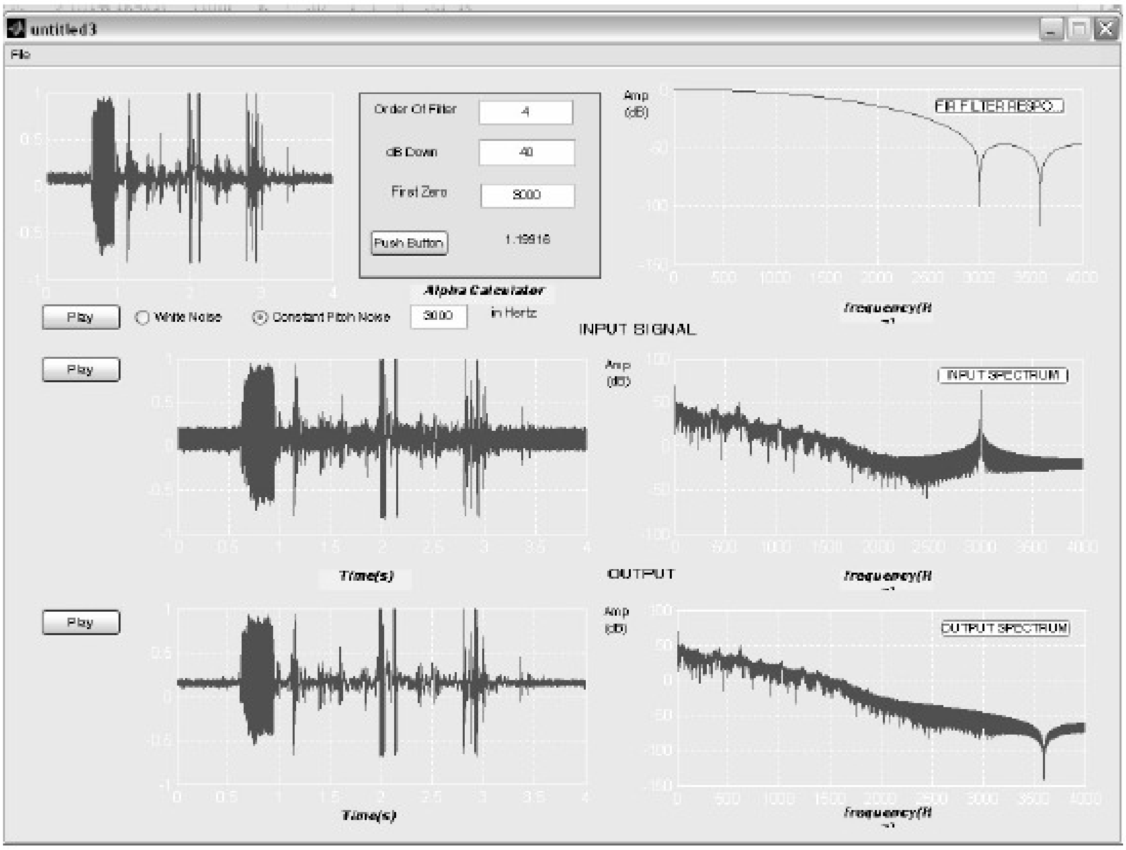


Figure 15 : Elimination of Continuous Pitch Noise

Some screen Shots of the GUI are shown in Figure (13), (14) and (15). Figure (13) shows the output of the filter when the value of alpha is not chosen correctly or the zero is not placed at the correct place; i.e., where the burst noise is occurring. Although the effect of burst noise is reduced in this case but complete elimination has not occurred. In Figure (14) it is clearly visible that the burst noise is eliminated completely. Figure (15) shows elimination of continuous pitch noise from the speech signal.

## V. CONCLUSIONS

The application successfully removes the burst noise and attenuates the higher frequency components of the white noise. The filter designed here can be realized practically and can be used with applications for speech filtering. Digital filters can easily realize performance characteristics far beyond what are practically implementable with analog filters. Because of its finite impulse response characteristics it can be easily adopted for mathematical calculations. The scope of this application can be extended for various other filtering applications. The potential areas of extension are listed below

- The filter concept can be modified to give a high pass filter or band pass filter.
- Instances of constant pitch noise occurring in the voice signal can be easily removed by multiple filtering using the same concept.
- It can be used as an anti-aliasing filter.
- It can be used for 'destriping' images by subsequent low pass and high pass filtering.
- It can be used for limiting the frequency band of the luminance signal in a video recorder.

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