An Efficient Stepper Motor Audio Noise Filter

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This paper provides a computationally efficient noise filtering technique for reducing stepper motor noise energy in a Digital Still Camera application without significantly altering the speech signal energy. The filtering technique uses a cascade of bi-quadratic filters to improve the signal to noise ratio by reducing noise energy. The noise filter described in this paper uses an Automatic Gain Controller stage to maintain signal energy when the noise filter is active. Such a filter achieves about 10 dB PSNR improvement at 2 MHz of ARM9 processor load. Furthermore, the paper briefly explores the filter structure for efficient hardware or software realization.

Index Terms—Bi-quadratic filter, digital filter, digital still camera (DSC) audio noise filter, infinite impulse response filter (IIR) filter, noise filter, stepper motor sound noise filter.

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1 Introduction

In Digital Still Camera (DSC), sound is recorded along with captured video frames for a movie capture application. The sound signal is converted to an electrical signal by a microphone and converted to a digital signal by an Analog to Digital Converter (ADC). Often the intent of movie capture is to record speech associated with the video (either verbal comments of the camera operator or the speech of the human subject under the movie capture).

While capturing video, it is possible to adjust the lens focus (zoom in / zoom out). The lens focus is adjusted by a stepper motor. The motor causes audible noise when active, which gets added onto the speech signal (being recorded) that is fed to the microphone. The computation complexity required by spectral domain noise subtraction is not practical in low power handheld devices [7], [1]. This paper describes the filtering approach to improve the signal to noise ratio (SNR) and intelligibility of speech, with less than 2 MHz CPU utilization on an ARM9 processor, when the stepper motor is active.

2 Noise Characteristics

In the case of additive noise, the noise spectrum is added onto the speech spectrum as in (1) [1].

$$x_n[k] = x[k] + n[k] \tag{1}$$

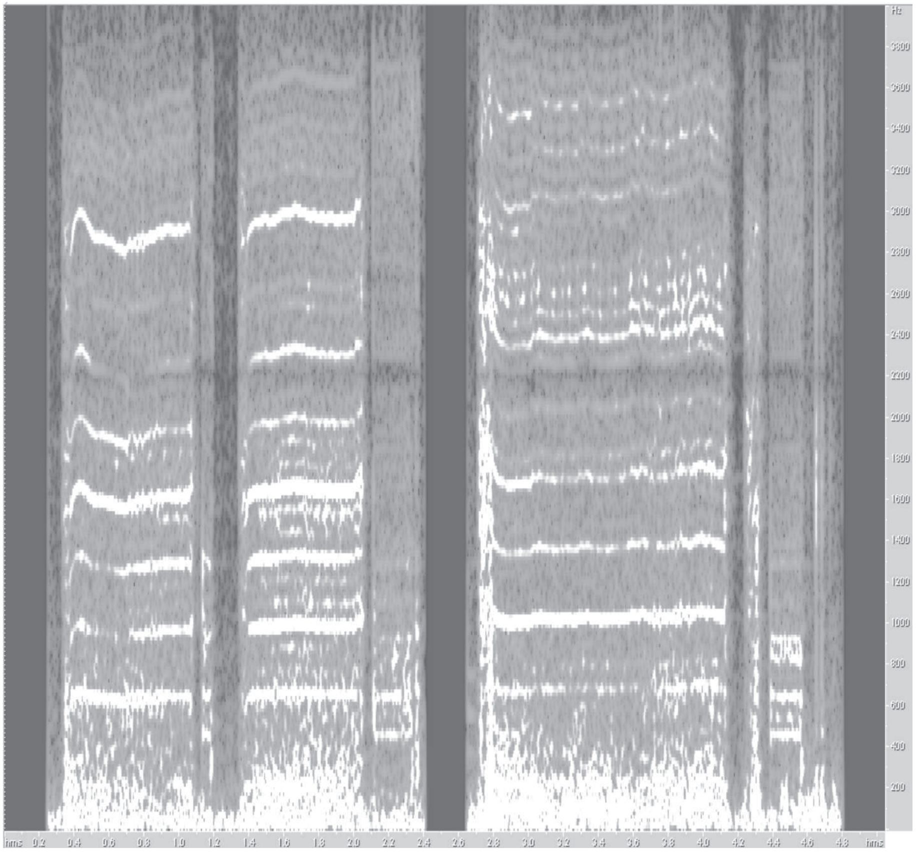
The different types of additive noise that may be encountered in a DSC are:

- **Microphone rumble noise**: Rumble noise is caused by low frequency sound produced by sound sources like wind, microphone feedback, and mechanical sounds. Rumble noise (<100 Hz) typically lies outside the speech spectrum. Thus, the fundamental frequency of rumble can be filtered out by a high pass filter (HPF).
- **Stepper motor noise**: Stepper motor noise is wideband with frequency content existing over the entire speech spectrum. The noise can be considered as segmented stationary noise (i.e. the noise when taken in short time window remain stationary). The stepper motor noise further has the characteristic of having significant power at low frequencies, high frequencies and distributed narrow-band noise as shown by spectrogram in Figure 1 [1]. The noise power is -20 dB and the sampling rate is 8 KHz for noise as shown in Figure 1.

  The effect of the stepper motor noise audibility, added onto the speech signal, depends on characteristics of the following:

  1. Microphone
  2. ADC/Digital to Analog Converter (DAC) filter
  3. Motor noise
  4. Microphone and motor placement
  5. DSC casing (sound absorption properties of the material and cabinet)
  6. Speech signal

Figure 1. Spectrogram of Stepper Motor Noise Used for Zoom Operations



By reducing noise density outside the speech spectrum, SNR can be improved as can be seen from (2) and (3). The speech signal bandwidth is about 50-5000 Hz. The prominent speech section is around 150-3500 Hz. By band-limiting the signal to 100-5000 Hz, noise power can be reduced, thereby increasing SNR and speech intelligibility of the noisy signal.

$$NoisePower = NoisePowerDensity \times Bandwidth \tag{2}$$

$$SNR = \frac{SignalPower}{NoisePower} \tag{3}$$

- **Background noise**: Background noise can be stationary or non-stationary noise. The noise is typically additive in nature as given by (1).

  In the case of stationary noise, the noise characteristic remains the same with respect to time and spectrum [5]. Some examples for stationary noise are idle engine sound, in-vehicle and tire noise in a running vehicle, machinery (white), sea wave, wind and babble.

  The noise characteristic varies with time and/or the spectrum of non-stationary noises [5]. Some examples are the rubbing of a hand against the DSC casing, foot steps, passing vehicles, sirens, horns, coughs and sneezes.

  Band-limiting the ADC output is effective for speech signals embedded in background noise as can be seen from (2) and (3).

3 Noise Filter Design

3.1 ADC Filter

Analog microphone output is converted to digital data by the ADC [10]. ADCs for audio are typically Delta-Sigma Modulators with decimation filters (to convert over-sampled digital data to the desired sample rate), gain controllers (pre-amplifiers) and optional anti-aliasing filters (to attenuate high frequency noise).

In order to prevent aliasing resulting from down sampling in the ADC, the digital data needs to be band-limited to a half-sampling rate as given by (4).

$$f_{Nyquist\_rate} = \frac{f_{sampling\_rate}}{2} \tag{4}$$

The decimation filter in the ADC would act as an LPF with a cut-off at the half-sampling rate. Thus, in the case of an 8 KHz ADC setting, the speech signal is limited to 4 KHz max frequency. However, in the case of 16 KHz sampling rate, the ADC output contains frequency components up to 8 KHz. In order to limit the signal bandwidth to that of the speech signal to reduce noise power (and increase SNR), a low pass filter might be needed.

3.2 BPF

The low frequency noise can be removed by the use of a high pass filter, without affecting signal power. The signal can be limited to the upper speech frequency by use of LPF as already seen. Thus, a BPF filter is suitable for improving SNR irrespective of the presence of an ADC filter. The band-pass filter can be realized by cascading LPF and a high pass filter (HPF) as given by (5).

$$H_{BPF}(z) = H_{LPF}(z) \times H_{HPF}(z) \tag{5}$$

A second stage of HPF can be added if the noise has significant power density at a low frequency (0-100 Hz).

3.3 BPF and Band Stop Filter Cascade

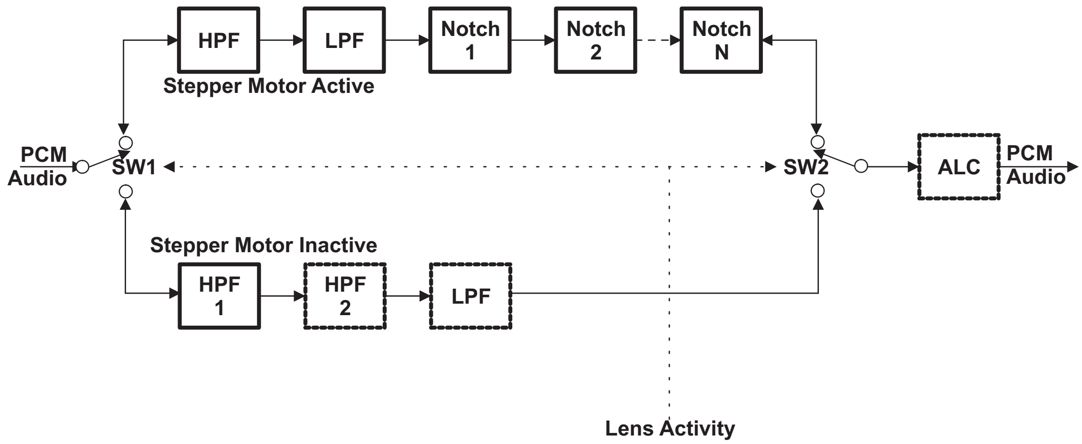
As can be seen from Figure 1, an efficient filtering for stepper motor noise should incorporate HPF, LPF, and notch filters. The high-pass filter is needed for reducing/removing stepper motor noise energy contained in low frequency and microphone rumble. If the noise energy is too high, cascaded two stages of HPF can be used [3]. An LPF with gradual attenuation can be used for reducing noise energy at a high frequency (2200-3800 Hz in Figure 1). Notch filter can be used for removing noise energy in narrow bands (1000-1100, 1300-1450, 1600-1800 Hz in Figure 1).

Figure 2 illustrates the cascading of filter stages. During the stepper motor operation (for zoom in and out), Pulse Code Modulation (PCM) samples are passed through cascaded filter stages. Since the motor is controlled within the DSC, the start time and duration for which the motor is running is known. Thus, the cascaded BPF and notch filters need to be turned on only during the activity of lens adjustment. This would aid in preserving the natural sound of speech, when the motor is inactive. In the case of buffering between the ADC and filter stages, the cascaded filter has to be active for an additional time for the duration of the buffered samples. This is typically required when the filters are implemented in software, since buffering of PCM samples between the ADC and filter is required.

In normal recording without zoom operations (stepper motor active), 1 or 2-stage HPF can be used to eliminate microphone rumble. Additionally, a high-pass filter can be used to reduce or minimize background noise (stationary and non-stationary).

4 Noise Filter Implementation

Figure 2. Cascaded Filter Stage in DSC



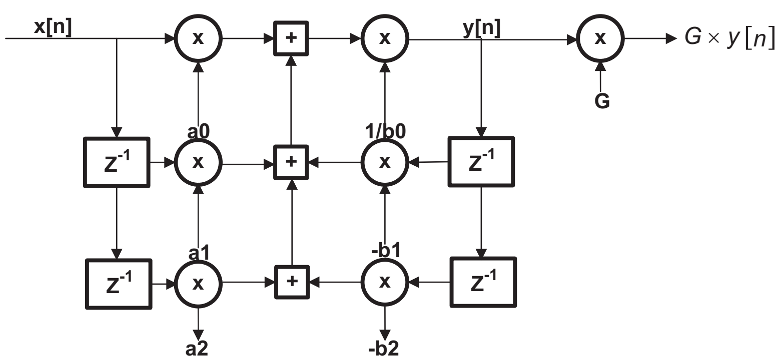
Second order Infinite Impulse Response (IIR) LPF and HPF can be used in cascade to create a noise filter as shown in Figure 2. Finite Impulse Response (FIR) filters would require the order of the filter to be higher to achieve the same frequency response of IIR filters, and hence would require increased computations.

A bi-quadratic filter can be used for creating different frequency responses by programming coefficients a0, a1, a2, b0, b1, and b2 as given by (6).

$$\begin{aligned} b0 \times y[n] = {} & a0 \times x[n] + \\ & a1 \times x[n-1] + a2 \times x[n-2] - \\ & b1 \times y[n-1] - b2 \times y[n-2] \end{aligned} \quad (6)$$

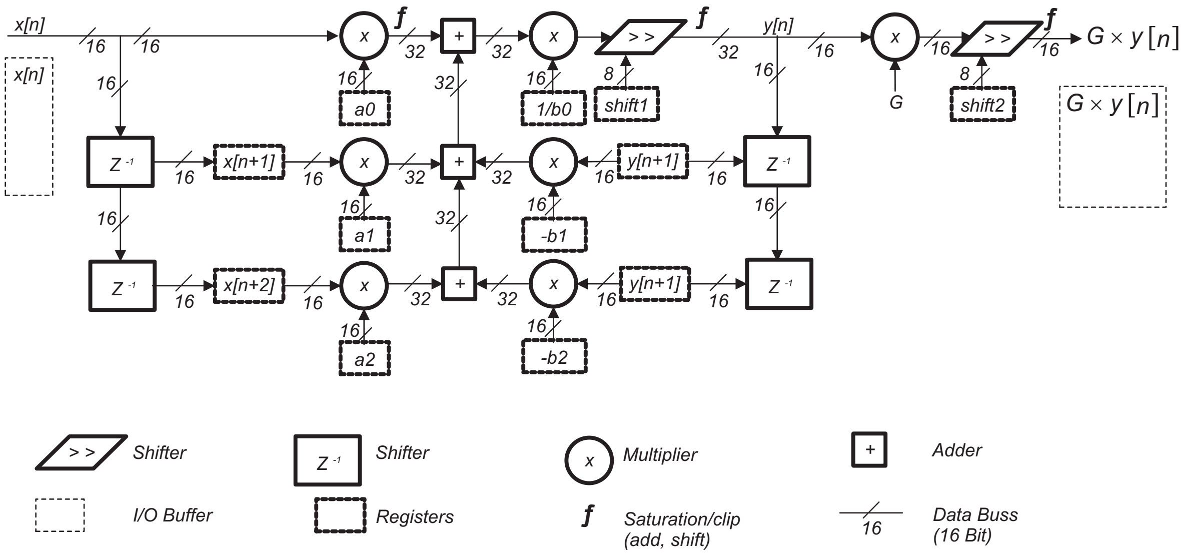
Scaling may follow the bi-quadratic filter to achieve unity gain, if b0 is not unity. The bi-quadratic filter structure in Transpose Direct Form-II is shown in Figure 3. The filter structure can easily be implemented in fixed-point software or hardware with the desired precision requirements [8], [2].

Figure 3. Bi-Quadratic Transpose Direct Form-II Structure



In order to reduce the gate count for a cascaded filter in hardware, loopback can be used using the structure shown in Figure 4. The filter realization structure is flexible enough to take different Q formats for coefficients by the use of shifters. Using a block processing of PCM samples, programmability of coefficients, and context (past output and input samples) save/restore features, it is possible to use the same structure for creating a cascaded structure in Figure 2. Thus, only a single stage needs to be implemented in hardware even for cascaded filter structures. With use of 1 MAC, the computational load can be addressed for sampled speech signals.

Figure 4. Bi-Quadratic Structure Hardware Block Diagram



In the case of filter design, take advantage of equal loudness curves. The ear is most sensitive to sound in 3-4 KHz. The LPF does not yield a sharp cut-off in second order IIR filters. Use gradual attenuation starting around 3 KHz for the filter design.

The HPF eliminates low frequency noises like rumble and wind noise from the signal captured by the DSC microphone. In case the noise attenuation is not sufficient with a single stage HPF, use cascaded two stages of second order HPF.

Narrow band noises (e.g. hum) can be eliminated by the use of a notch filter. The bi-quadratic filter structure can be programmed for notch filter creation.

After the filter stages, an optional Automatic Level Controller (ALC), as shown in Figure 2, can be used to boost the speech signal energy [6]. The ALC is realized with less than 0.5 MHz for 8 KHz sampled signal on the ARM9EJ processor [5].

5 Experimental Results

Figure 5 and Figure 6 illustrate the noise reduction achieved by cascaded Chebysev-II second order LPF shown in Figure 7, and second order IIR HPF shown in Figure 8 [4], [5]. The computation requirement of such a filter is below 1.5 MHz on ARM9EJ processor with 1-cycle memory access. The input signal is speech embedded in motor noise, sneeze, and thud on a microphone sampled at 16 KHz.

Figure 5. Histogram of speech+motor noise+sneeze+thud

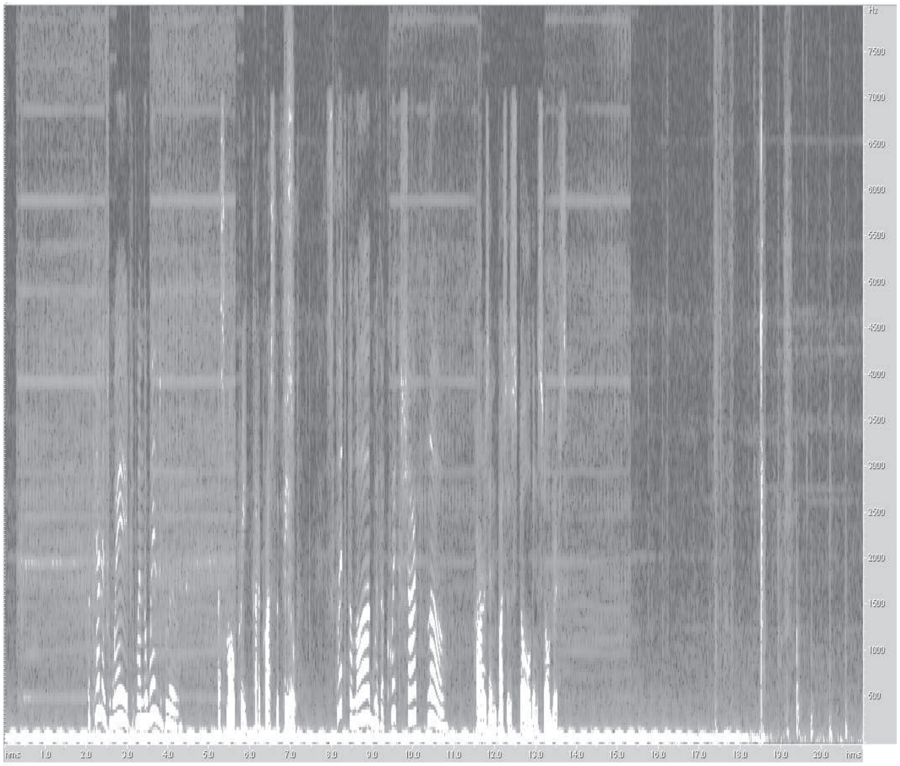


Figure 6. Histogram of Noise Filtered speech+motor noise+sneeze+thud with BPF

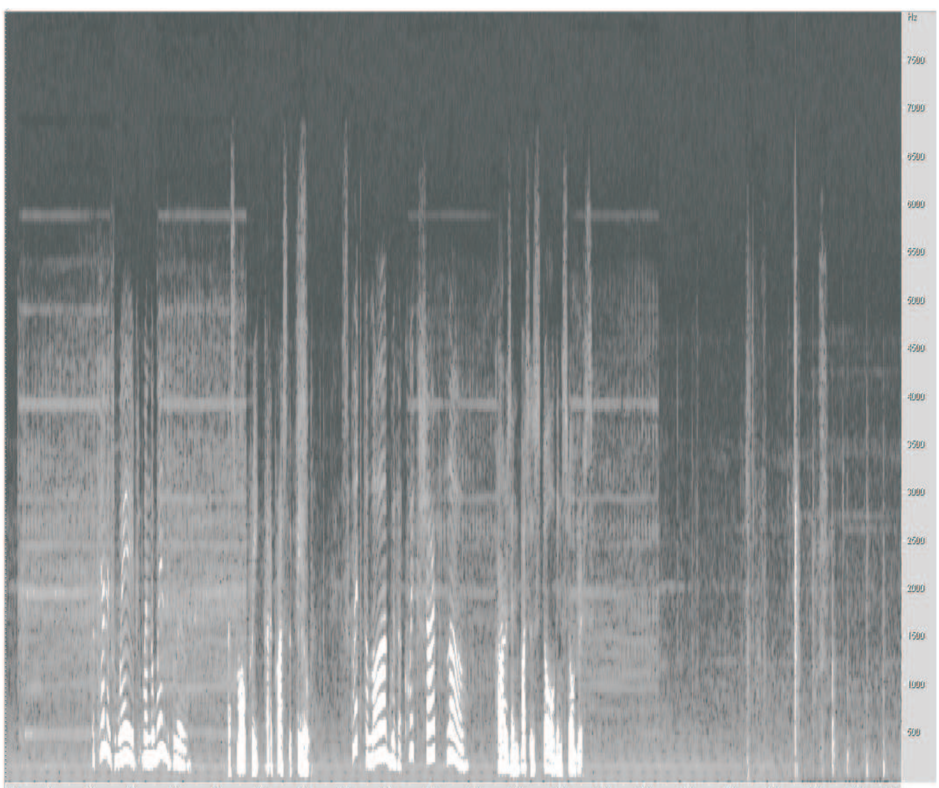


Figure 7 shows the magnitude, phase and group delay response of a low pass bi-quadratic with filter coefficients as follows: a_0 = 0.0793, a_1 = 0.1335, a_2 = 0.0793, b_1 = -1.1064, b_2 = 0.3983. Note that the frequency roll-off in Figure 7 starts about 2 kHz and is down to -26 dB at 5 kHz. The speech intelligibility is maintained, whereas the noise energy is reduced.

Figure 7. LPF Magnitude, Phase, Group Delay Response

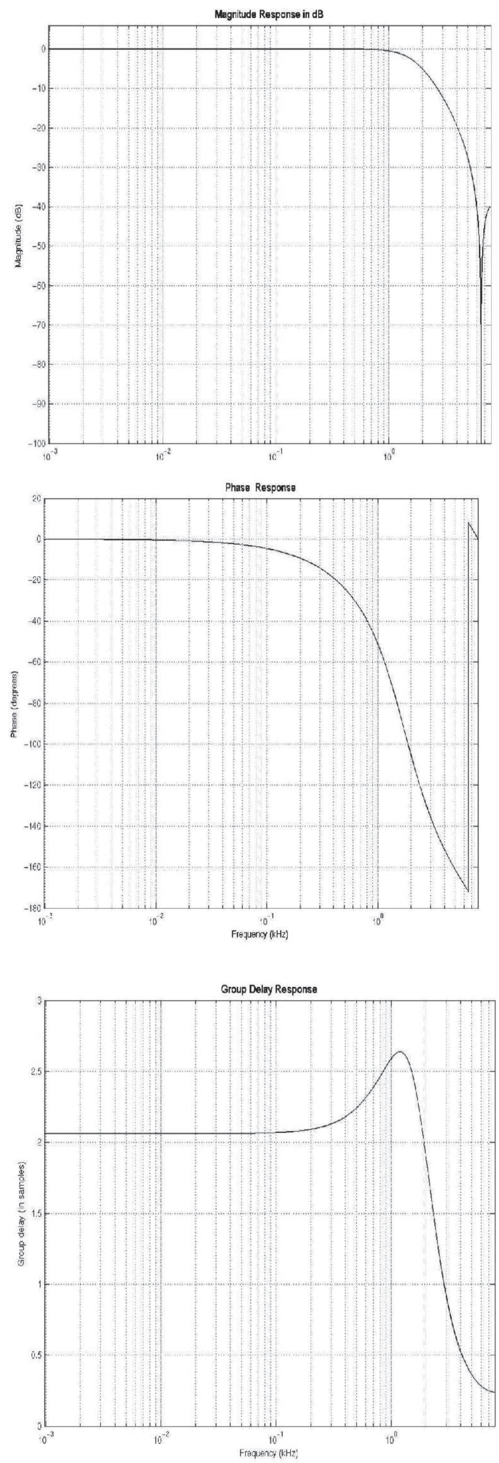


Figure 8 shows the magnitude, phase and group delay response of a high pass bi-quadratic with filter coefficients as follows: a0 = 0.9617, a1 = -1.9233, a2 = 0.9617, b1 = -1.9219, b2 = 0.9248. The frequency roll-off in Figure 8 starts about 120 Hz and is down to -19 dB at 50 Hz. This provides significant low frequency noise attenuation with a single stage. The speech signal energy is preserved.

Figure 8. HPF Magnitude, Phase, Group Delay Response

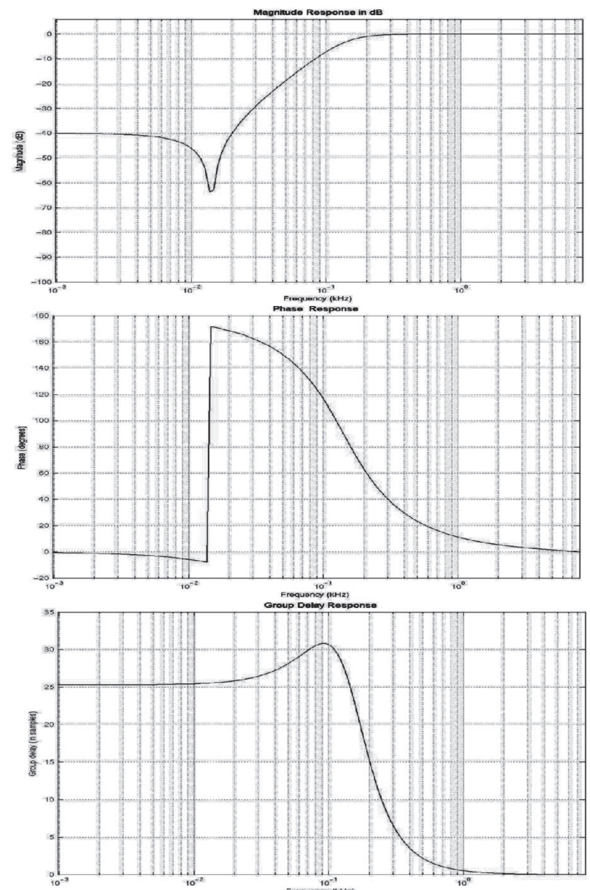


Figure 9 and Figure 10 illustrate noise reduction by use of cascaded second order Butterworth filters (HPF with cut-off frequency at 300 Hz, LPF with cut-off frequency at 1700 Hz, Band stop (notch) filter with cut-off frequencies 1500-1800 Hz, and Band stop (notch) filter with cut-off frequencies 1200-1450 Hz) [4], [5]. The computation requirement of such a filter is below 3 MHz on the ARM9EJ processor with 1-cycle memory access. The input signal is additive motor noise and a speech signal sampled at 8 KHz. The cascade of filters in Figure 11, Figure 12, Figure 13, and Figure 14 would result in the cross-coherence of input and output signals as shown in Figure 15.

Figure 9. Histogram of speech+motor-noise

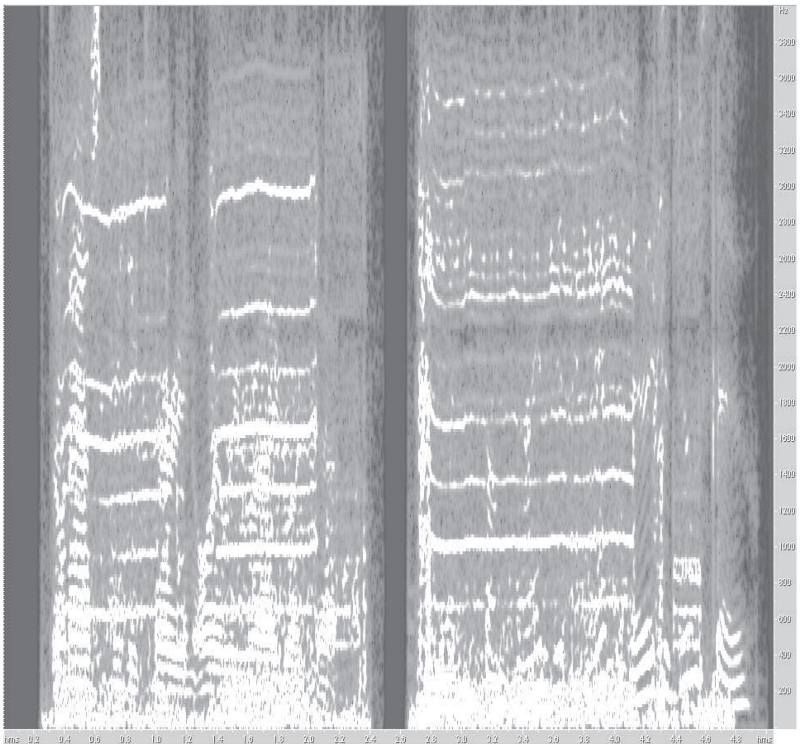
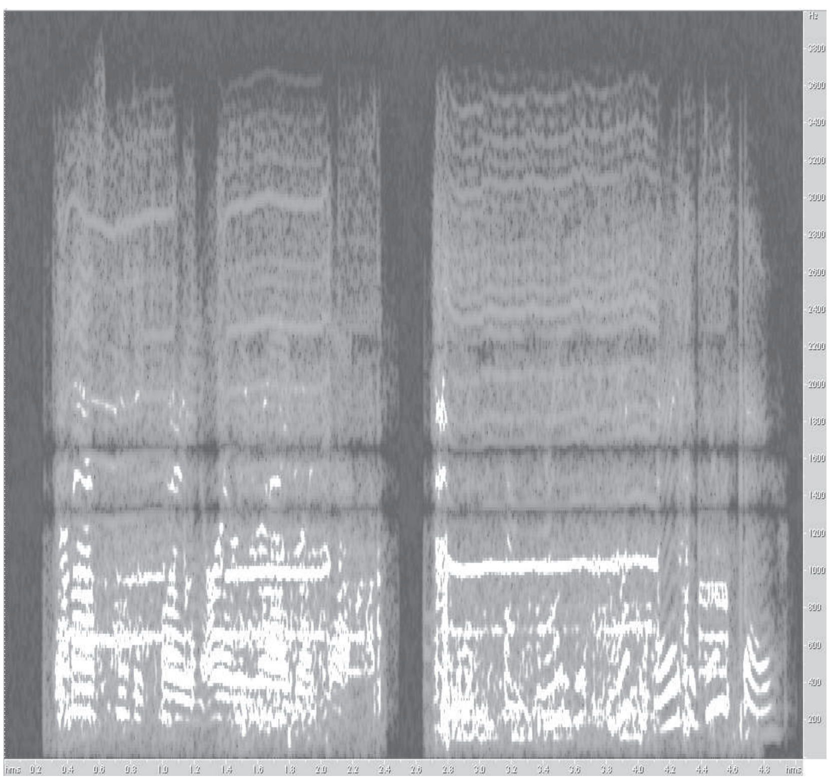


Figure 10. Histogram of Noise Filtered speech+motor-noise With BPF and Notch Filters



The band-pass plus notch filtering enhances the noisy speech intelligibility in the presence of stepper motor noise by preserving the prominent speech band while suppressing everything outside this band. The filtered signal has intelligible speech and significant reduction in noise power (8-12 dB noise power reduction). Speech power reduction due to filtering is in the order of 1.5 to 2.5 dB. SNR improvement is in the order of 10 dB. The background stepper motor noise is substantially masked by the speech, thereby improving intelligibility [11], [1]. This is proved by listening tests. The narrow band band-stop filters have very less impact on speech signal quality (with harmonics), as observed in listening tests. The LPF and HPF with sloping stop-band do very little to affect speech energy present at low frequencies and clarity carried at high frequencies, as can be verified by listening tests and cross-coherence plot.

Figure 11 shows the filter response of LPF with coefficients as follows: a0 = 0.846459, a1 = -1.692918, a2 = 0.846459, b1 = -1.669203, b2 = 0.716633. The cut-off frequency is at 300 Hz, and attenuation is around 20 dB at 100 Hz. The group delay is small and the phase response is close to linear.

Figure 11. Filter Response of LPF

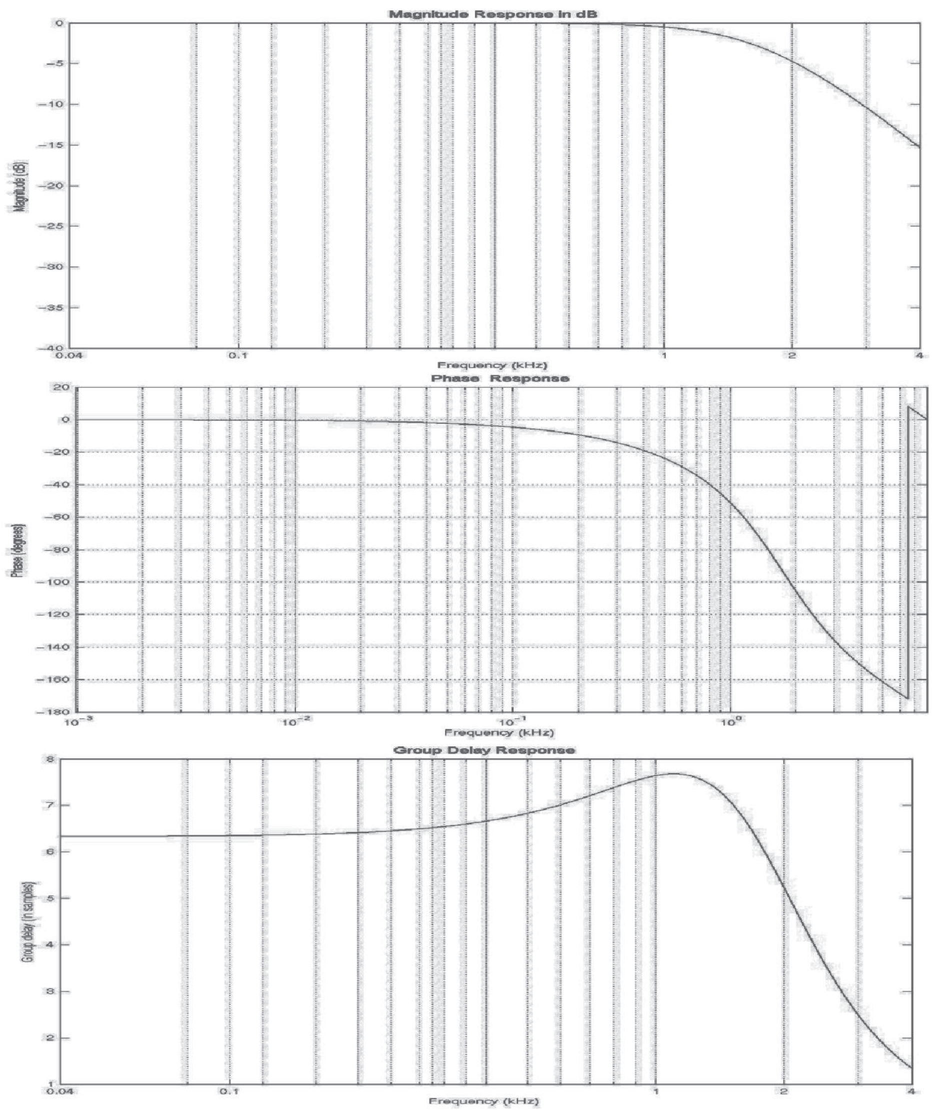


Figure 12 shows the filter response of HPF with coefficients as follows: a0 = 0.227117, a1 = 0.454235, a2 = 0.227117, b1 = -0.276664, b2 = 0.185136. The cut-off frequency is at 1700 Hz, and attenuation is around 10 dB at 3 KHz. The HPF has a slower roll-off compared to LPF in order to maintain the speech signal energy at high frequencies, which is important for intelligibility. The group delay is small. A cascade of two stages of the same HPF filter would provide attenuation of 40 dB at 100 Hz.

Figure 12. Filter Response of HPF

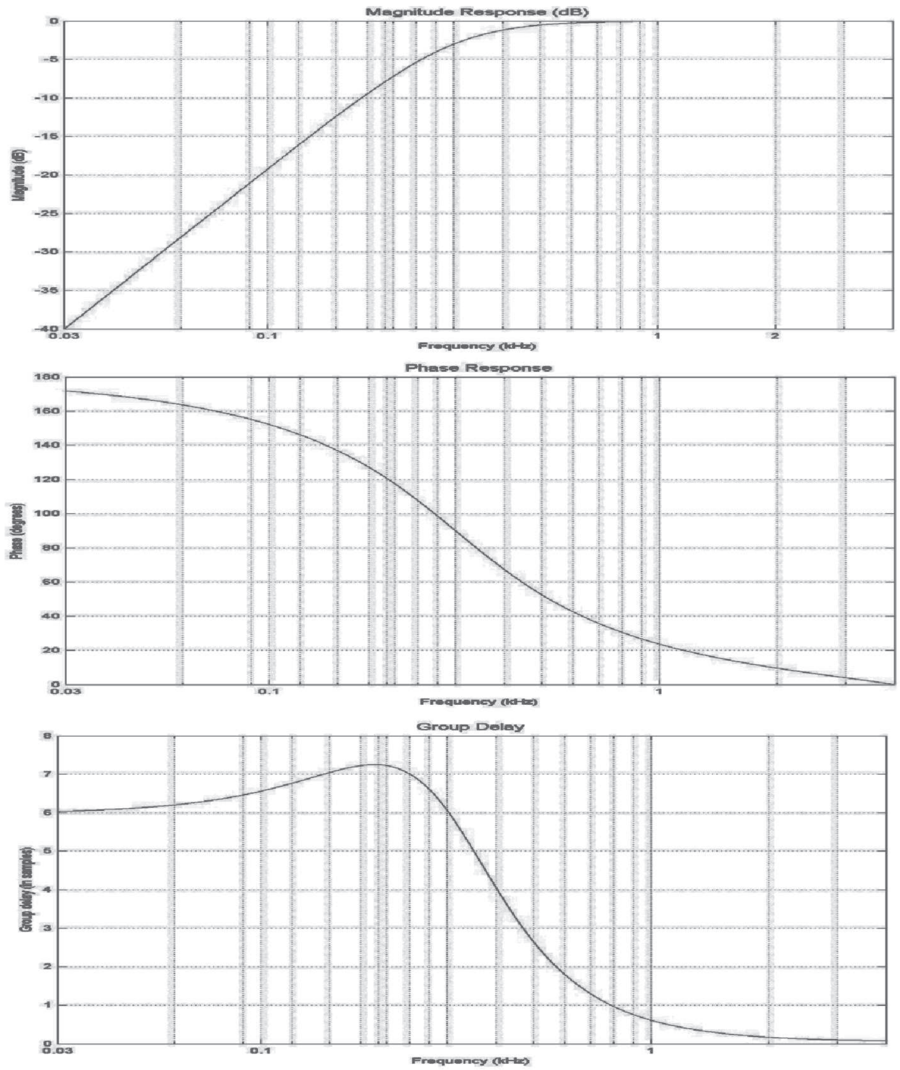


Figure 13 shows the filter response of a notch filter with coefficients as follows: a0 = 0.910339, a1 = -0.925094, a2 = 0.910339, b1 = -0.925094, b2 = 0.820678. The cut-off frequency is at 1200, 1450 Hz with as much as 40 dB attenuation at the center of stop-band. The purpose of band-stop or notch filters is to reduce the noise energy by attenuating the frequencies where noise energy is concentrated. The impact on the speech signal is minimal with respect to intelligibility and signal energy since the speech signal consists of fundamental frequency and harmonics.

Figure 13. Filter Response of Notch Filter

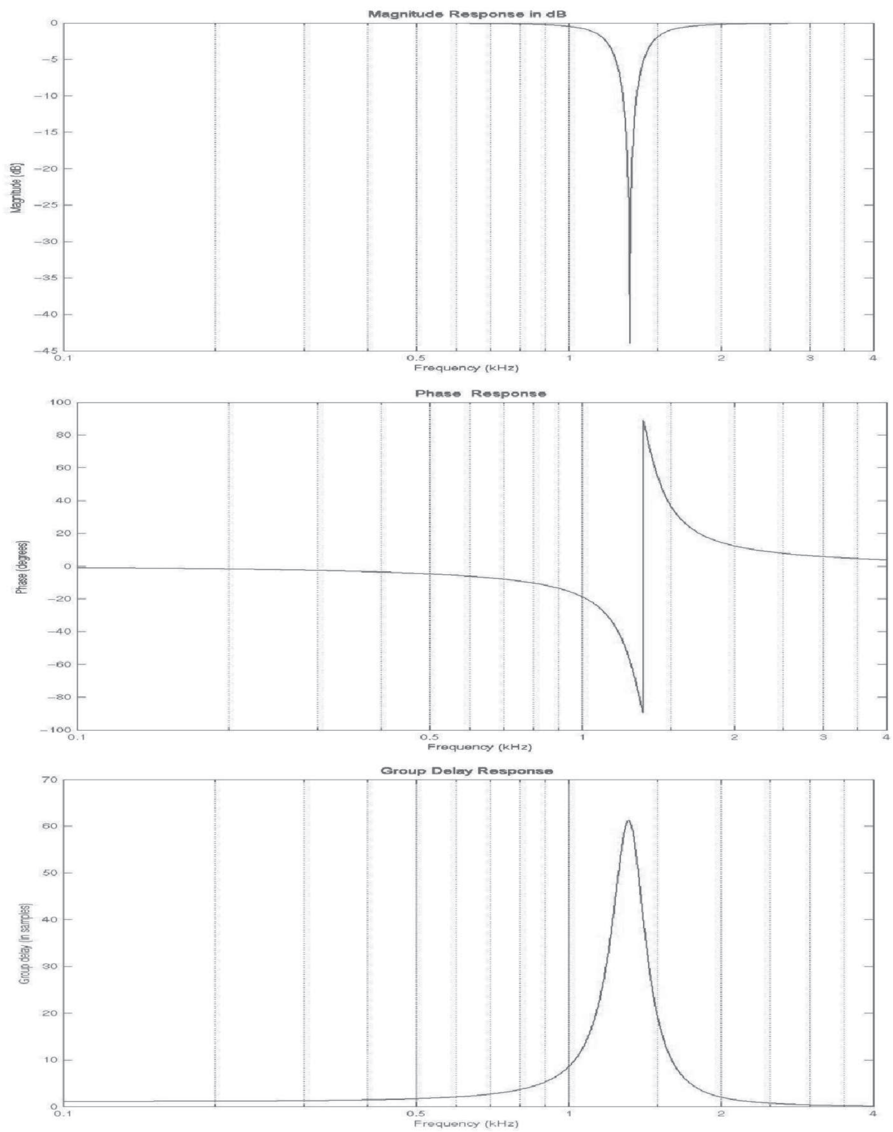


Figure 14 shows the filter response of a notch filter with coefficients as follows: a0 = 0.894168, a1 = -0.488815, a2 = 0.894168, b1 = -0.488815, b2 = 0.788336. The cut-off frequency is at 1500, 1800 Hz with as much as 50 dB attenuation at the center of stop-band.

Figure 14. Filter Response of Notch Filter

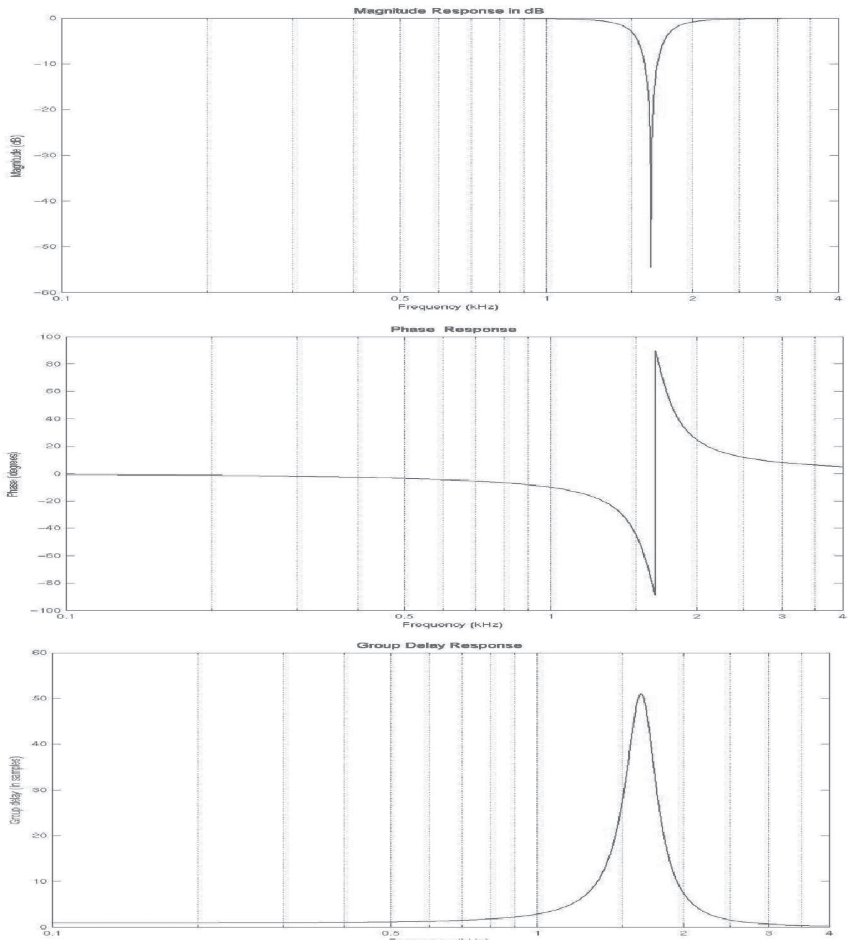
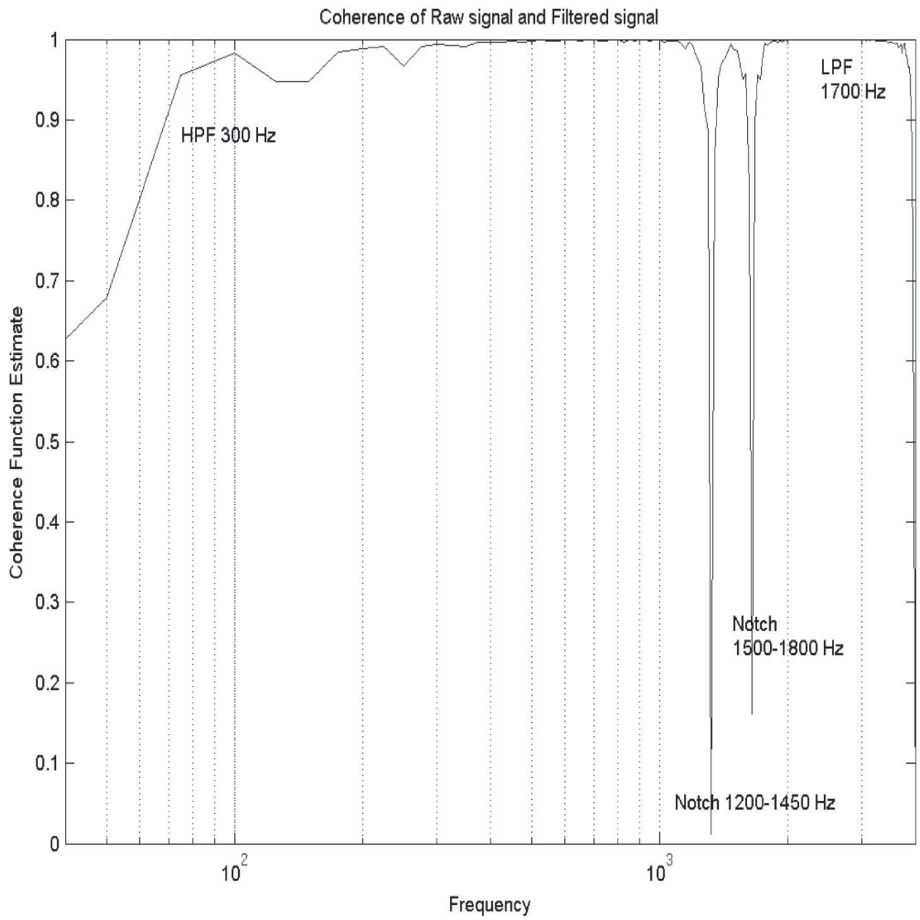


Figure 15. Coherence of Input and Output of Cascaded-Filter



6 Summary

By use of cascaded bi-quadratic filters and ALC, significant improvement can be achieved on speech intelligibility and clarity by apt choice of filter responses. Such cascaded filters require a very low processing load, about 2MHZ on the ARM9EJ processor, when implemented in software. The cascaded filter stages can be realized in hardware using single stage bi-quadratic structure in hardware with gate count as low as 5000.

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