Sound Signal Analysis Using FIR Filters for Musical Fountain Operation

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***Abstract*- In digital signal processing (DSP), FIR digital filter is very important device to deal with particular frequencies of a certain signal to be appropriate for some applications such as communications, sound equalizers, etc. In this paper, FIR filters are adopted to decompose the original sound signal into four signals. Each one is created by one FIR filter and each filter represents a narrow band of frequencies. The filter output is used to drive a certain variable speed drive (VSD) to control the speed of a water pump and light intensity of a colored lamp. This filter output signal is applied to the analog control voltage terminals of the VSD unit to control the frequency and magnitude of the voltage supplied to the lamp and pump. Thus, the heads of the water jets and the light intensity are controlled according to the analog control signals which are created by the FIR filters (The VSD is used to map the filter output into light intensity and water head by controlling the supplied voltage of them). The goal of this study is to design and simulate four sound harmonics bands produced by FIR filters to drive four VSDs which are simulated using V/F ratio constant method for musical fountain operation.**

Index Terms*- *finite impulse response(FIR), digital filter, fast Fourier transform (FFT), musical fountain, variable speed drive (VSD).

I. INTRODUCTION

Fountain history returns to thousands of years ago. Ancient people used fountains as a water source and for cooling. Over the years, fountains have become used to decorate the cities, for entertainment purpose as well as a source of water. Recently, fountains have been developed into multimedia shows with light, music and special effects. Many of them are used in a lot of urban places because they give an aesthetic to the place, create a beautiful environment, and the economic importance by attracting many viewers.

To continue the delights of viewers, the water fountains should create unpredicted patterns. They should be able to stimulate the hearing sense as well as the visual one to produce more integrated fountains. To accomplish this aim, the present fountains developers use digital controllers to control their pumps and lights. In the present work, the control system of the musical fountain is proposed based on the sound harmonics which are produced by the four FIR digital filter to drive four pumps and four lamps via four VSDs.

There are several studies have been presented in recent years such as: in 1999, Miguel, et al., [1], designed and implemented 20 band FIR filter. In this study they used TI TNS320C31 processor and ADC and DAC with 44.1k Hz sampling frequency for high audio quality. In 2004, Shakerin S., [2] designed a water fountain that displays three letters U-O-P. Each letter is displayed for 0.5sec by using BS2 microcontroller according to a certain binary code which is given to the nine output ports for every 0.5sec. In 2009, Yoo M. and Lee I., [3] introduced intelligent musical fountain authoring system, the system produced a musical fountain scenarios by analyzing musical information (onsets and beats). Then using Bayesian network to operate nozzles according to the probabilistic relationships. Due to the Bayesian network is based on probability, the generated scenarios are different in each case. Thus the nozzles will be different in operation according to the current probabilities at each onset time or beat time recognized. In 2014, Surumbarkhuzhali and RachelinSujae, [4], implemented real time audio equalizer. The effect of control is by volume and channel movement when press the push buttons on ADSP-BF533.In this study, they designed low pass, high pass, band pass and stop band chebyshev IIR filters. In 2015, Mohsin and et al, [5], designed and simulated FIR filters with window method using MATLAB program. In 2015, Feng Jiang, [6], introduced a fancy fountain control system by using PLC and AC drive. Audio signal is applied to PLC as a discrete signal then, this signal compared with a fixed value in PLC. The PLC output given to the inverter to control the pump.

This paper organized as following: section II gives theoretical background about the water jets, FIR filter with window technique and the fast Fourier transform (FFT). Section III will review the MATLAB/SIMULINK of a proposal system of a musical fountain and the simulation results and section IV gives the conclusion.

II.THEORETICAL BACKGROUND

A. Heights Control of Water Jets

There are two variables to define the pump output. The first variable is the head H (m) and the second is the flow quantity Q (m3/h). The human eyes can not feel the water flow quantity instantaneously but the variation of the water head is sensed by the eyes, therefore the water jets head will be taken as the attracting variable. This variable can be controlled either by controlling the pump speed (using VSD) or by throttling (closing an upstream control valve) [7].

In the current work, the first way is adopted to control the water jets heights by varying the input frequency and the input voltage of the AC motor via VSDs. There are several advantages for using variable speed AC drive to control the pump such as maintaining a constant torque, accurate pump speed control that matches the process requirements, energy saving, soft starting and the facility to connect the VSD to a controller for automatic process control [7].

In the pump, the affinity laws that link the pump characteristics (head H, flow Q, and power P) operating at varying speeds n1 (previous speed) and n2 (current speed) are shown in the following equations [8]:

$$\frac{Q_1}{Q_2} = \frac{n_1}{n_2} \tag{1}$$

$$\frac{H_1}{H_2} = \left(\frac{n_1}{n_2}\right)^2 \tag{2}$$

$$\frac{P_1}{P_2} = \left(\frac{n_1}{n_2}\right)^3 \tag{3}$$

Figure 1 illustrates the variation curves of Q, H and P for centrifugal pump depending on the pump rotation speed n [8].

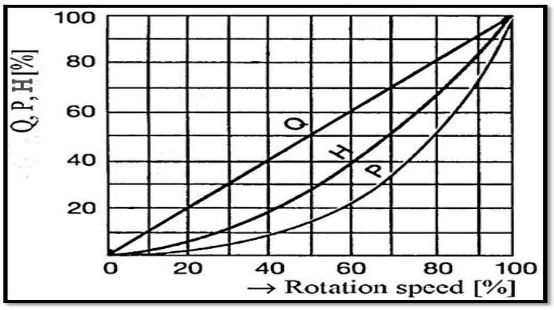


Fig.1 Flow, head and power for centrifugal pump versus pump rotation speed.

Practically, there are two terms of speed in the induction motor, shaft speed and synchronous speed. The rotor or shaft speed is close to the synchronous speed (revolving magnetic field speed) but not equal to it. Therefore controlling the synchronous speed results in controlling the shaft speed. The synchronous speed Ns is function of the applied frequency (F) as shown in the following relation:

$$\mathrm{Ns} = 120\ \mathrm{F/P} \tag{4}$$

Where P is the number of motor poles.

In V/F ratio constant mode of operation of the variable speed AC drive, the relationship between the coming voltage control signal Vc and the value of the VSD output voltage frequency (F) is :

$$\mathrm{F} = \mathrm{K} * \mathrm{Vc} \tag{5}$$

Therefore from eqauations 4 and 5, the motor rotor speed Nr can be approximated to :

$$\mathrm{Nr} = \mathrm{K} * \mathrm{Vc} \tag{6}$$

B. Finite impulse response (FIR) filters with window technique

FIR filters have impulse response with finite duration. In this filters, the relationship between the input signal and output signal is the convolution sum and it is described by the equation (7) [9].

$$y(n) = \sum_{i=0}^{N} \mathrm{bi}\ x\ (n-i) \tag{7}$$

where b_i are filter coefficients and N represents filter order.

By taking the Z-transform of Equation (7), we get equation (8) [9].

$$Y(z) = \sum_{i=0}^{N} b_i\ X(z) z^{-i} \tag{8}$$

$$H(z) = \frac{Y(z)}{X(z)} = b_0 + b_1 z^{-1} + \ldots\ldots + b_N z^{-N} \tag{9}$$

The structure of FIR filter is shown in Fig. 2.

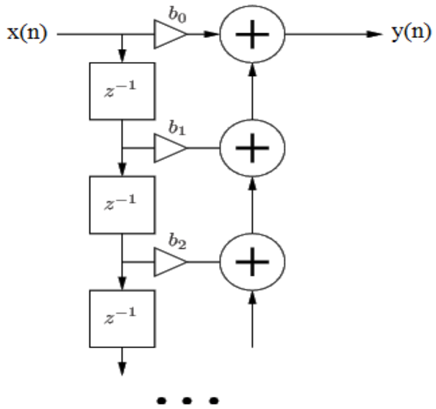


Fig.2 The structure of FIR filter

The non causal FIR filters coefficients are given as following [9]

For low pass

$$h(n) = \begin{cases} \dfrac{\Omega_c}{\pi} & n = 0 \\ \dfrac{\sin(\Omega_c n)}{n\pi} & for\ n \neq 0 \quad -M \leq n \leq M \end{cases} \tag{10}$$

For high pass

$$h(n) = \begin{cases} \dfrac{\pi - \Omega_c}{\pi} & n = 0 \\ -\dfrac{\sin(\Omega_c n)}{n\pi} & for\ n \neq 0 \quad -M \leq n \leq M \end{cases} \tag{11}$$

For band pass

$$h(n) = \begin{cases} \dfrac{\Omega_h - \Omega_l}{\pi} & n = 0 \\ \dfrac{\sin(\Omega_h n)}{n\pi} - \dfrac{\sin(\Omega_l n)}{n\pi} & for\ n \neq 0 \quad -M \leq n \leq M \end{cases} \tag{12}$$

where Ω_h is normalized cutoff frequency, Ω_h is normalized upper frequency, Ω_h is normalized lower frequency. $\Omega = 2\pi f T_s$, f is frequency in (Hz), f_s is sampling frequency in (Hz) and $T_s = 1/f_s$.

For causal filter n=n-M and the filter coefficients $b_n = \mathrm{h(n)} = \mathrm{h(n - M)}$ where n = 0,1,2,…….,2M.

The window method is used to design the FIR filters. When apply the window to the filter impulse response, we get

$$h_w(n) = h(n).w(n) \tag{13}$$

where w(n) the window function [9]

There are several windows such as rectangular, Bartlett, Hanning, Hamming and Blackman. The convolution h(n) and w(n) has the effect on the smoothing of the filter frequency response [10]. when the order increase, the window become narrower and the width of the area between pass-band and stop-band edges [10]. The rectangular window not have a significant impact on the side lobs which are generated at both ends of the filter compare with all the other windows [10]. The windows describe in the following Equations [9].

The rectangular window

$$w_{rec}(n) = 1 \qquad -M \le n \le M \tag{14}$$

The Bartlett window

$$w_{bar}(n) = 1 - \frac{|n|}{M} \qquad -M \le n \le M \tag{15}$$

The Hanning window

$$w_{han}(n) = 0.5 + 0.5cos\left(\frac{n\,\pi}{M}\right) \qquad -M \le n \le M \tag{16}$$

The Hamming window

$$w_{ham}(n) = 0.54 + 0.46cos\left(\frac{n\,\pi}{M}\right) \qquad -M \le n \le M \tag{17}$$

The Blackman window

$$w_{ham}(n) = 0.42 + 0.5cos\left(\frac{n\,\pi}{M}\right) + 0.08cos\left(\frac{2\,\pi\,n}{M}\right) \qquad -M \le n \le M \tag{18}$$

$$\frac{\Delta f}{fs} = \frac{B_W}{k} \tag{19}$$

where k=order of the window function, B_w=window bandwidth in bins, $\Delta f =$ frequency difference.

Window must be with length 2M+1.The window is symmetrical and can gradually down the FIR filter coefficients to zeros at both ends for the range of -M ≤n≤ M [9].

C. Fast Fourier Transform (FFT)

The fast Fourier transform transfers the time signals into the frequency domain. In the time domain, the vertical axis represents the amplitude and the horizontal axis represents the time, but in the frequency domain, the vertical axis represents the power value and the horizontal axis represents the frequency.

If the input signal is analog signal, it is necessary to use the sampling theorem to convert it into the digital. If the sampling rate of the signal is f_s, the frequencies of the signal will be showed in less than $f_s/2$ in the frequency spectrum.

III. SIMULATION RESULTS

The complete proposed system of a musical water fountain which is simulated in MATLAB/SIMULINK is shown in Fig. 3.

From Figure 3, the sound input block is used to bring the desired sound file which is stored in the computer to the MATLAB/SIMULINK. After the sound input blocks, each MATLAB function blocks stands for a one FIR filter code. Also, it is possible to select the discrete FIR filter block in Simulink library as shown above instead of the programming code. When the filters outputs are produced, each one is represented by a signal with amplitude from 0V to 2.25V by taking the absolute value of the filter output and then its multiplied by 2. This signal is linked as an analog control input to the variable speed AC drive block.

The AC drive controls the speed of the pump in proportion to voltage control signal that is obtained from the FIR filter. The minimum value of the control input corresponding to a zero speed and water head. The maximum value of the control signal corresponding to the rated motor speed and maximum water head. AC drive drives the pump based on the V/F ratio still constant at any variation in the pump speed. RMS block is used to find the r.m.s value of the output voltage. The MATLAB function1 block is used to find the motor speed by writing the speed equation of the induction motor inside this block.

In this paper, four FIR filters are designed. The first with a 800Hz cutoff frequency is a low pass filter. The second with frequencies from 1300 Hz to 2600 Hz is a band-pass filter. The third with frequencies from 3100 Hz to 6000 Hz is a band pass filter. The fourth with 6500 Hz cutoff frequency is a high pass filter. These filters are designed in these selected ranges of

frequencies to obtain less overlap between them and less filter order. Figure 4 illustrates the filters coefficients or the impulse response of each filter with hamming window. The length of each filter equals to 51.

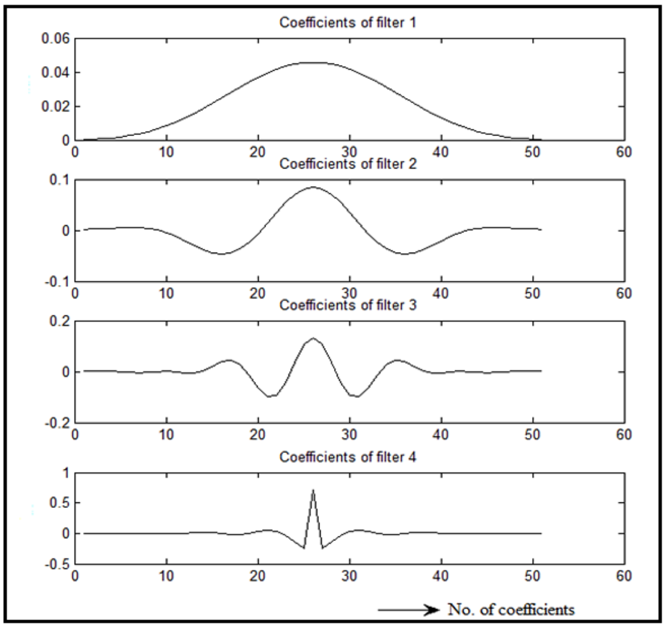


Fig.4 Filters coefficients with Hamming window

The system is simulated using MATLAB program on PC have the following specifications:

Processor: Intel(R) Core(TM) I5-5200U CPU @ 2.20GHz

Capacity of Hard: 500GB

Installed memory (RAM): 4GB

System type: 64-bit operation system, x64 based processor

The magnitude response of the FIR filters with hamming window is shown in Fig. 5. from this Figure, it can be seen that the x-axis is the normalized frequency in interval [0 1] which equals the frequency from 0 to $f_s/2$ (i.e from 0Hz to 22050Hz).

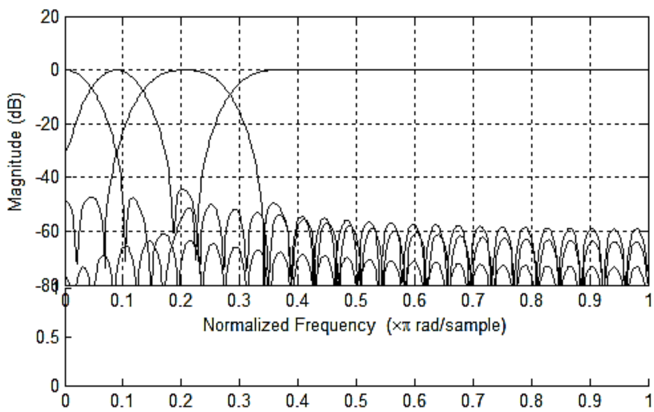


Fig.5 Magnitude response of FIR filters.

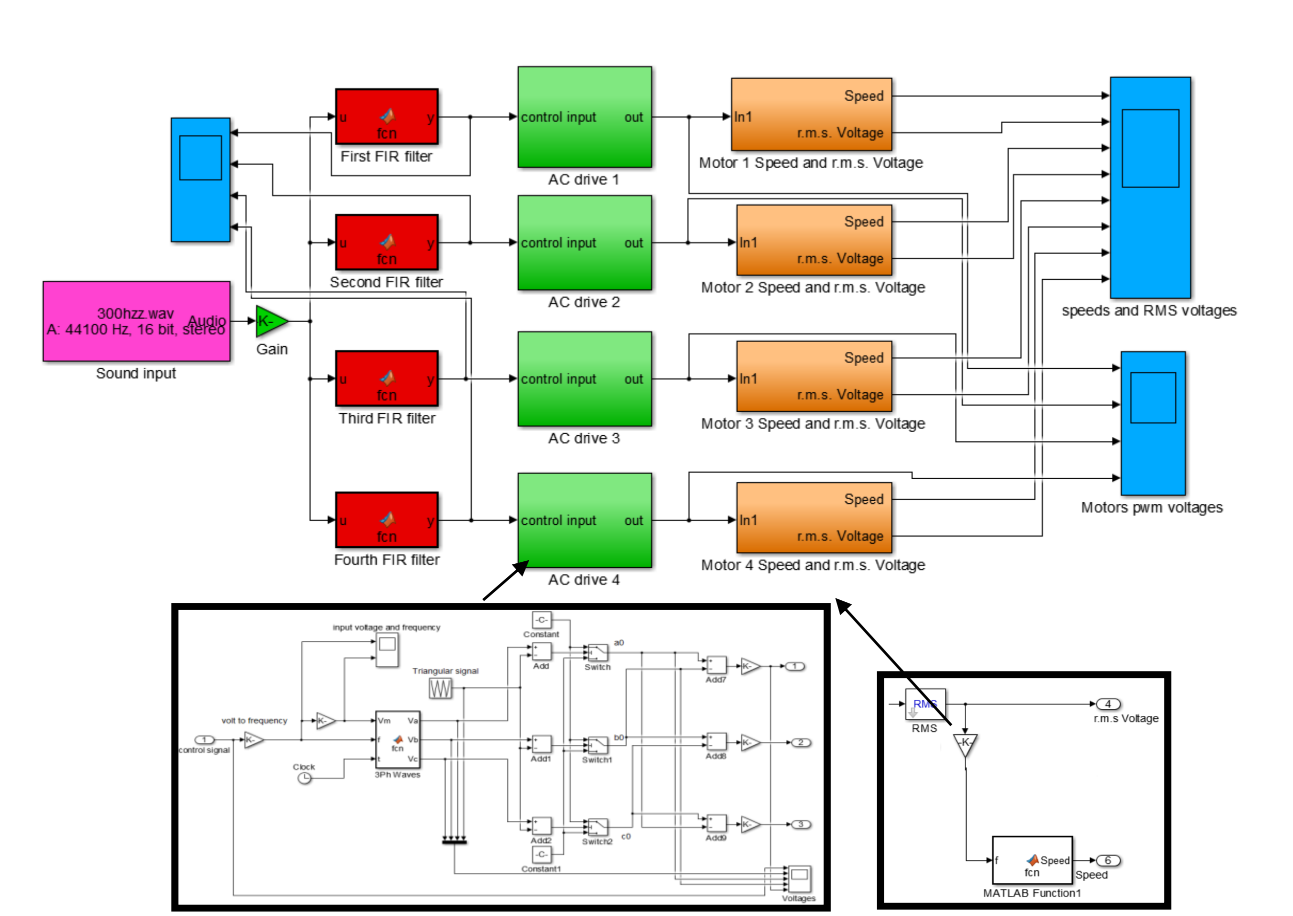
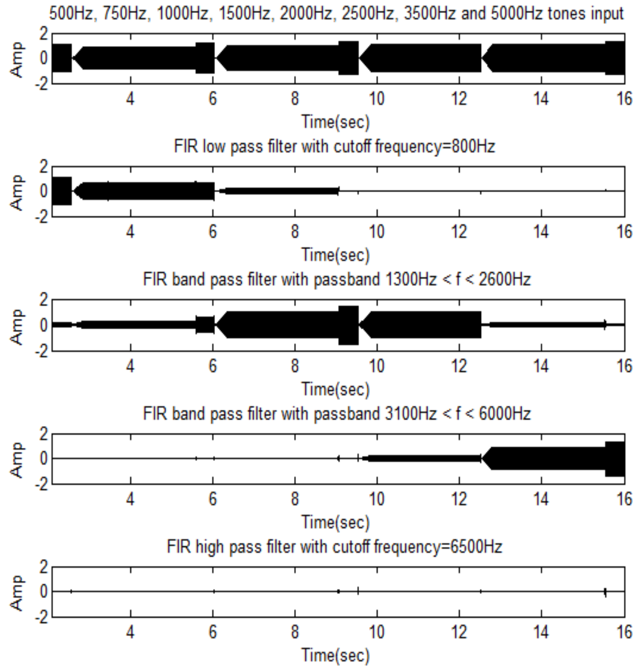


Fig.3 MATLAB/SIMULINK model of a complete musical fountain system

Figure 6 shows the filters outputs when a 500Hz, 750Hz, 1000Hz, 1500Hz, 2000Hz, 2500Hz, 3500Hz and 5000Hz tones are applied as an inputs. The outputs of a 10000Hz tone input is shown in Fig. 7. Figure 8 illustrates the filters outputs with the music input.



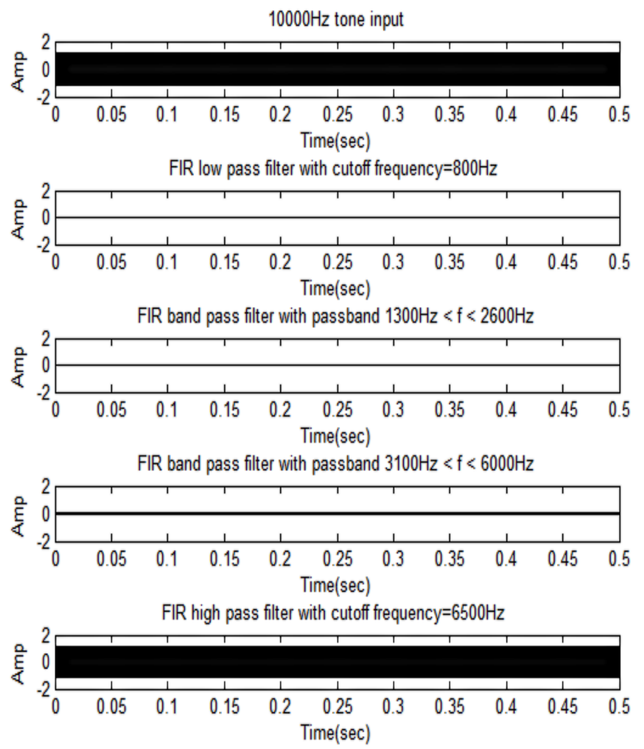


Fig.7 FIR filters output with 10000Hz tone input

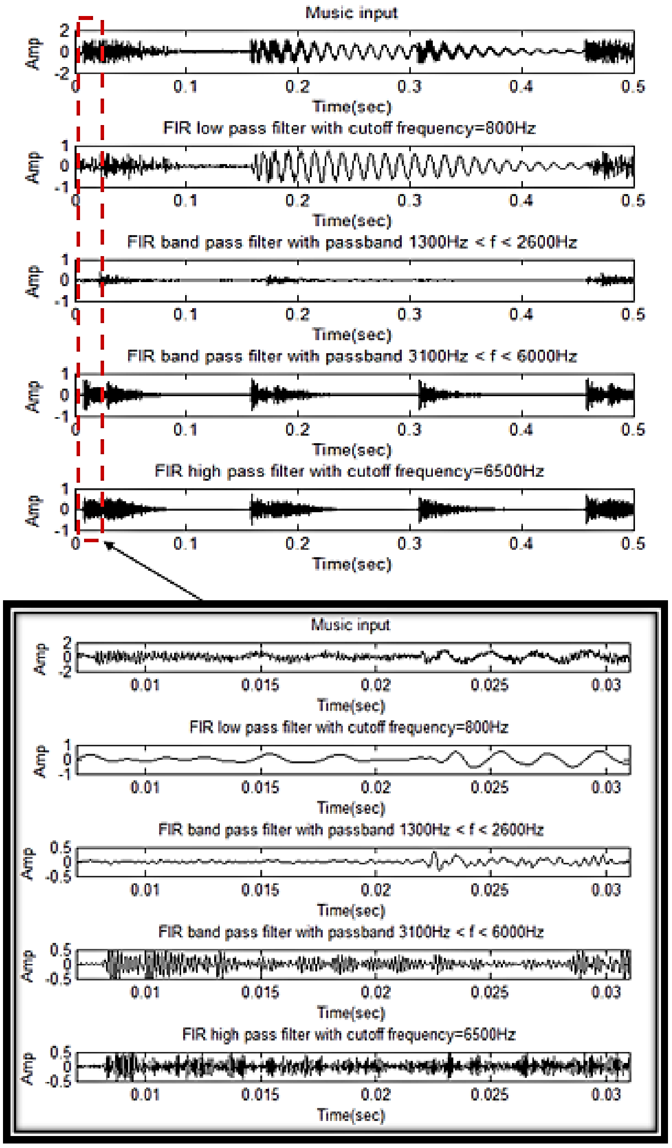


Figure 8 FIR filters outputs with music input

In Figure 6, it can be noted that each filter is allowed to pass only a tone which has the frequency belong to its pass-band frequency region. The output of the fourth filter is zero because there is no tone that carries frequency equals or above 6500Hz .

From Figure 7 it can be seen that only the fourth filter is allowed to pass a 10000Hz tone input due to this filter has been designed to pass any frequency above 6500Hz, all the other filters have almost zero output.

Figure 8 indicates that all filters with different amplitudes and different frequencies according to the filters specifications and the frequencies in musical sound.

Figure 9 shows the frequency domain for the input and filters outputs by taking the FFT when applied the same input in the Fig.6. The frequency domain for a 10000Hz tone input and all filters outputs by taking the FFT are presented in Fig. 10. Figure 11 illustrates the frequency domain for filters outputs and input when the same music in Figure 8 is applied as an input.

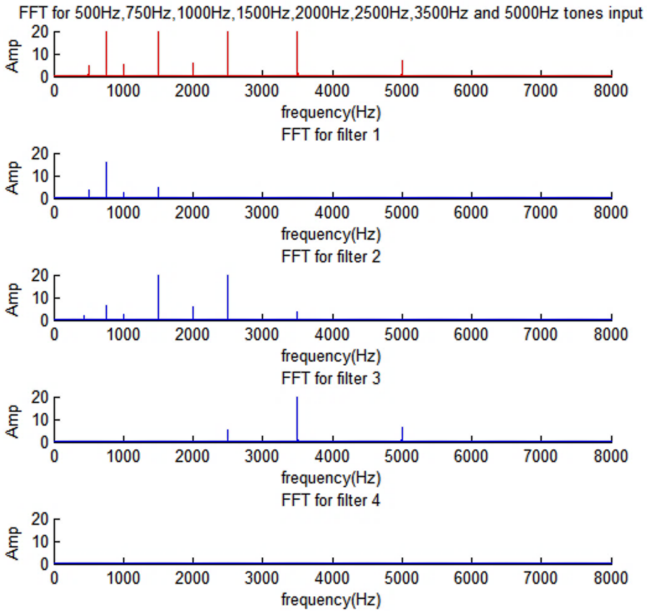


Fig.9 Frequency domain of FIR filters outputs with 500Hz, 750Hz, 1000Hz, 1500Hz, 2000Hz, 2500Hz, 3500Hz and 5000Hz tones as input

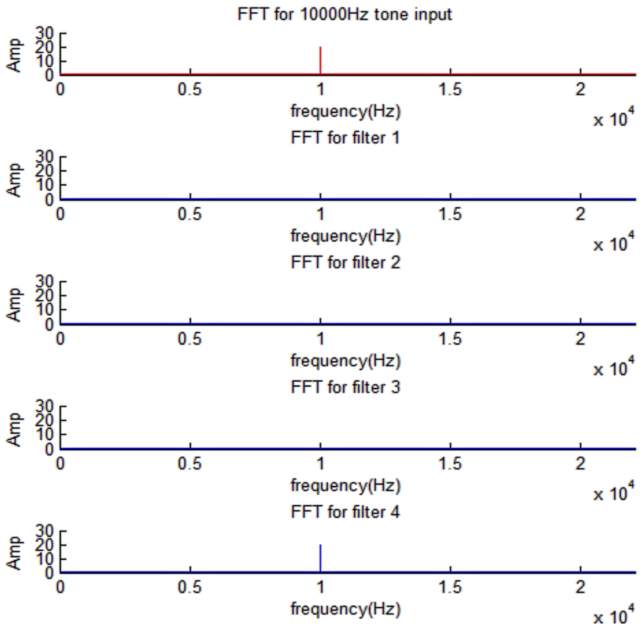


Fig.10 Frequency domain of FIR filters outputs with 10000Hz tone input

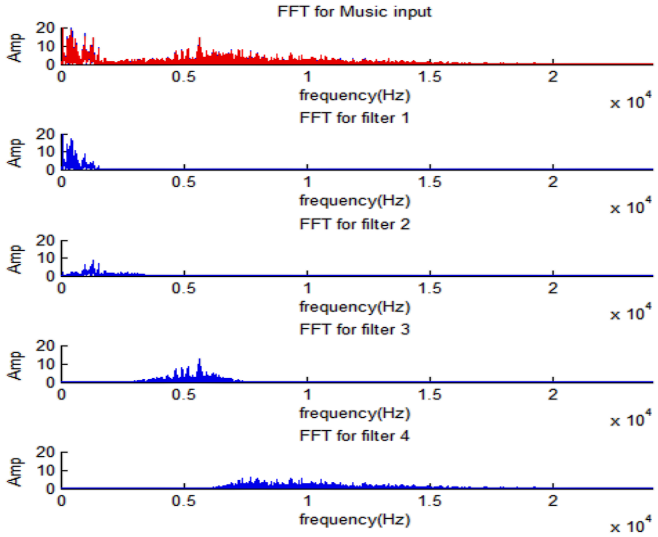


Fig.11 Frequency domain of FIR filters outputs with music input

In Figure 9 it can be noted that the input and filters outputs in frequency domain, the signals are illustrated as a number of spikes, each one have a certain power and frequency. Each filter is allowed to any input spike which has a frequency belong to its pass-band frequency region to pass through it. Due to the fourth filter is designed to pass any frequency above 6500Hz, only this filter is allowed to pass a 10000Hz input spike through it as shown in Figure 10. Figure 11 shows the frequency domain for all filter when applied music input, from this Figure, it can be seen that the low frequencies components are passed through first filter, medium frequencies components are passed through the (second and third) filters and high frequencies components are passed through the fourth filter.

Figure 12 shows the speed and voltage of each motor (pump) when apply 500Hz, 750Hz, 1000Hz, 1500Hz, 2000Hz, 2500Hz, 3500Hz and 5000Hz tones as inputs. The speed and voltage of each motor when a 10000Hz tone is applied as input is shown in Figure 13. Figure 14 illustrates the speed and voltage of each motor with the same musical sound input in Fig. 8.

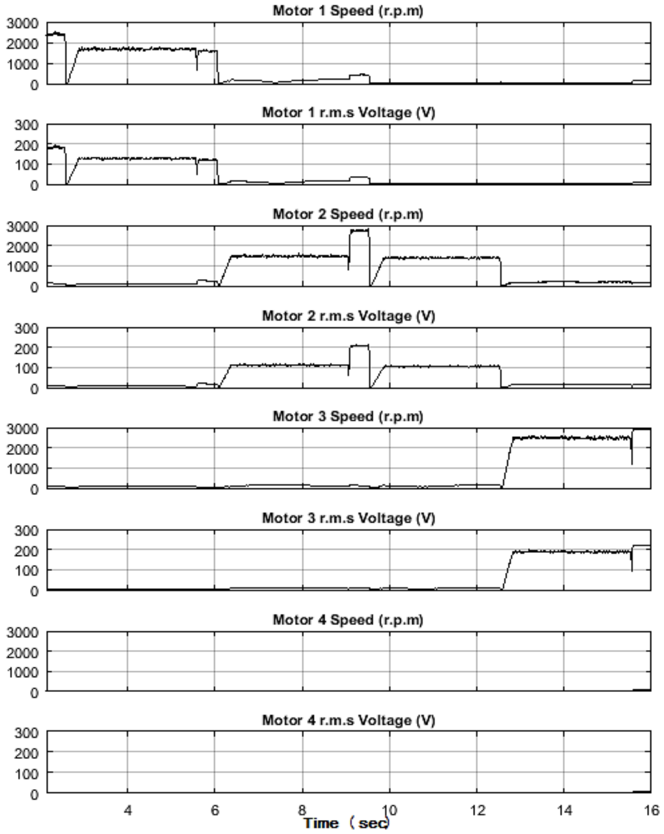


Fig.12 Motors speeds and r.m.s voltages of 500Hz, 750Hz, 1000Hz, 1500Hz, 2000Hz, 2500Hz, 3500Hz and 5000Hz tones as input

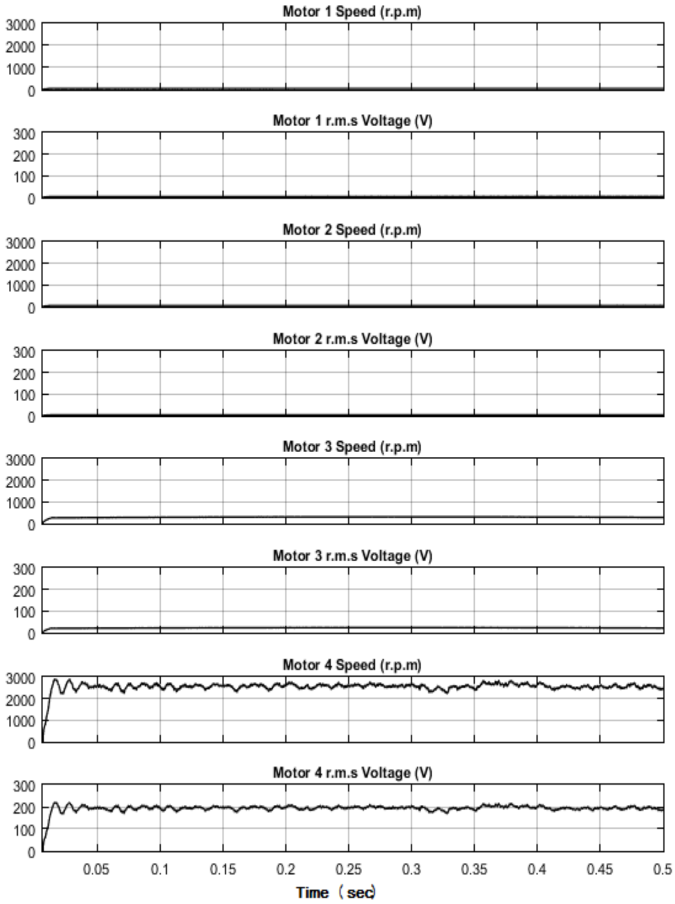


Fig.13 Motors speeds and r.m.s voltages of 10000Hz tone as input

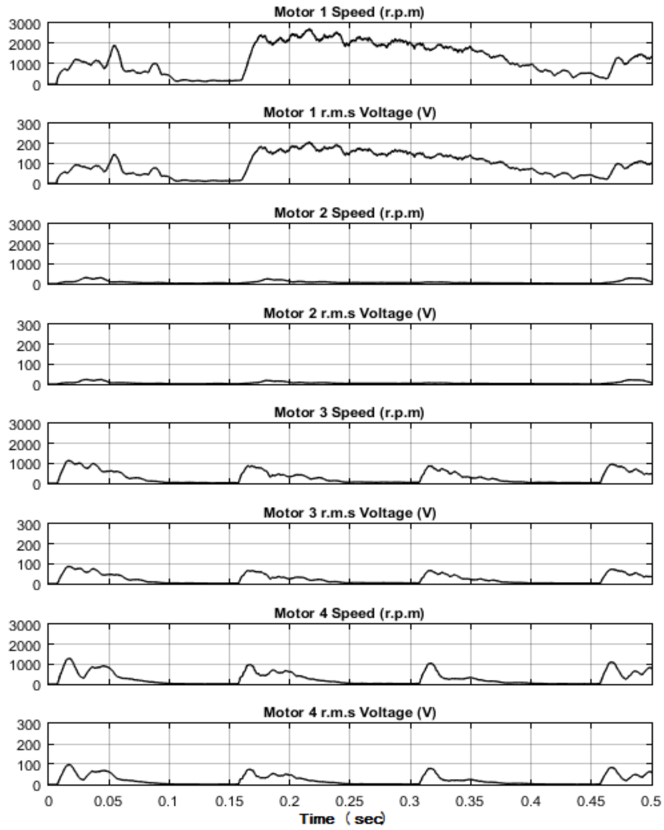


Fig.14 Motors speeds and r.m.s voltages of the music input

From Figure12, it can be seen that each motor have speed and r.m.s voltage directly proportional with the amplitude of the filter output that shown in Fig.6. In Figure 13 the fourth motor has a high r.m.s voltage and speed values while all the other motors have almost zero voltage and speed because the fourth filter has been designed as a high-pass filter with a 6500Hz cutoff frequency and this filter output is linked to the fourth AC drive which controls the speed of the fourth motor. In Figure 14 all the motors have variable values of voltage and speed according to the variation of the filters outputs in Fig.8 which are related to these motors.

Figure 15 illustrates the ramp input to represent the control voltage signal in the range from 0 to 2.25V. This input is used only to test the simulation of the pump that is driven by a VSD.

The relationship curve between the pump speed and the control voltage signal that is shown in Figure 15 is shown in Fig 16. The speed is directly proportional to the control voltage in the V/F constant method according to the equation (6).

Figure 17 shows the parabola curve between the pump speed and the water head from the nozzles of this pump. This curve is very compatible with the equation (2) which is illustrated in Fig.1.

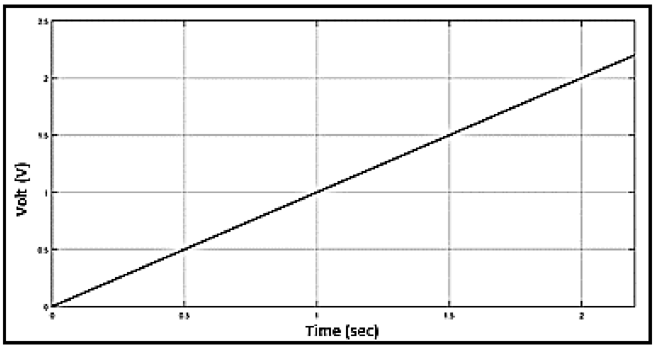


Fig.15 Control voltage signal

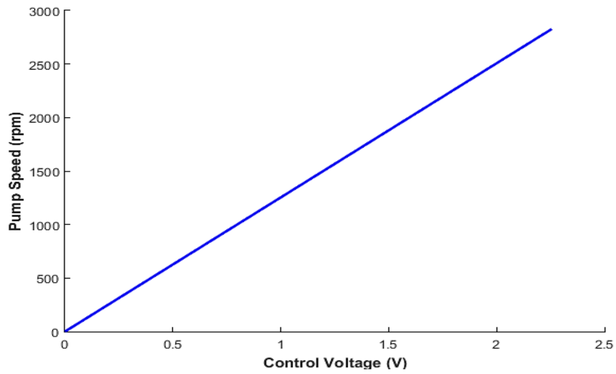


Fig.16 Pump speed versus voltage control signal

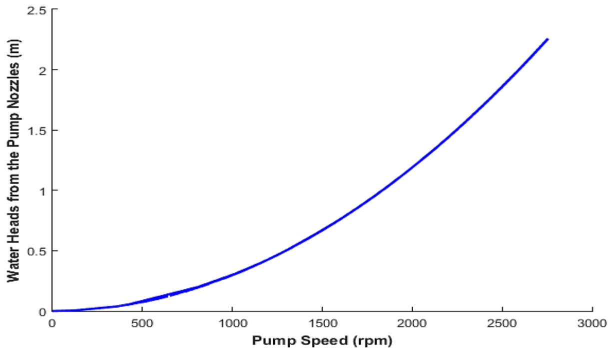


Fig.13 Water head versus pump speed

IV.CONCLUTION

In this paper, the proposed control system for musical water fountain has been simulated. In this system, four FIR filters are designed for low, medium and high frequencies to separate the original sound signal into four signals. Each FIR filter represents a band of frequencies that creates output signal with a special frequency to control the altitudes of the water jets and light intensity of the colored lamps using VSD.

The MATLAB Simulink of the VSD has been performed based on the method of V/F ratio remains constant throughout the analog voltage control range. Then, the simulation of the VSD has been used to simulate the overall system.

The simulation results of the filtering output signals and the speed of each motor are very satisfactory and compatible with the desired objective.

Due to each lamp is connected in parallel with one pump and they are driven by one AC drive unit, the light intensity of the lamp is directly proportional with the motor speed and their supply voltage are the same VSD output (minimum motor speed and light intensity are achieved at minimum VSD output voltage and vice versa) .

This system is not specified to work with a certain sound signal(s).

Practically, to implement this system, it is necessary to use a high speed controller such as DSP chip, four VSDs, four pumps and four lamps to construct a musical water fountain that is operated based on the sound frequencies and the power of these frequencies. This proposed system has been applied in practice, but with the IIR filters. It works in the best status and shows the unpredicted water jets patterns.

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