International Journal of Computing and Digital Systems
ISSN (2210-142X)
Int. J. Com. Dig. Sys. 9, No.2 (Mar-2020)
http://dx.doi.org/10.12785/ijcds/090217

Design and Implementation of a Musical Water Fountain Based on Sound Harmonics Using IIR Filters

Ali M. Jasim[1,2], Jawad R. Mahmood[2] and Ramzy S. Ali [2]

[1] *Communication Engineering Department, Iraq University College, Basrah, Iraq*
[2] *Electrical Engineering Department, University of Basrah, Basrah, Iraq*

Received 18 Jun. 2019, Revised 03 Feb. 2020, Accepted 23 Feb. 2020, Published 01 Mar. 2020

Abstract: Infinite Impulse Response (IIR) digital filters are used in many types of Digital Signal Processing (DSP) applications. One of these application fields is the sound which occupies an effective area of our life and takes number of functions like communication, entertainment, etc. These filters are used to separate the mother signal into its comprising partners groups. In this paper, the design and implementation of musical sound driven water fountain has been covered. The scenario is to decompose the adopted sound signal into four frequency bands (where each band occupies narrow range of frequencies and represents the output signal of its producing IIR filter) and allow each band to drive one submersible pump via an AC Variable Speed Drive (AC VSD) unit. The Arduino based IIR filter's output is applied to the analog voltage control terminal of the AC drive unit to control the magnitude and frequency of the voltage supplied to the pump to control the head of the water and at the same time the light intensity of colored light lamps. The hardware implementation has been done by four groups of components, where each one consists of one Arduino microcontroller (to function as an IIR filter), one AC drive unit to map the filter output into variable voltage variable frequency voltage signal, one submersible pump to produce variable head water burst (the head follows the filter output) and one colored pump to display the filter output behavior in visual mode.

Keywords: Infinite Impulse Response Filter, Digital Filter, Sound Harmonics, Musical Fountain, AC Drive

1. INTRODUCTION

Human beings have been dealing with water fountains for ages. Earlier, they were used as water source points. Currently, they are no longer used as water source points but they are used for more urban purposes. They can be used for evaporative cooling in open areas, can be used to draw people to specific zones in gardens through producing spectacular and unusual water jets. In general, one can say, the water fountains are necessary for modern life from points of view of economic and health because of their contributions in making the environment more beauty and also comfortable for citizens and visitors.

To keep the delights of visitors, the fountains should produce non ordinary patterns. They should produce dynamic and unpredicted patterns. They also should be able to excite the hearing sense in addition to the visual one to create more integrated water fountains. To achieve this goal, the modern fountains developers adopt digital controllers to drive their pumps and / or nozzles. In this context, in 2004, Said S., [1] designed a water fountain that display three letters U-O-P. Each letter is displayed for 0.5sec by using BS2 microcontroller. In 2009, Min-Joon, Yoo and In Kwon Lee, [2] introduced intelligent musical fountain authoring system, this system produces fountain scenarios by analyzing musical information (onsets and beats) and using Bayesian network to operate nozzles at each recognized time according to the probabilistic relationships. In 2014, Copindean R., Munteanu R. and Dragan F., [3] presented PLC driven water fountain in which the water jets and colored lights are controlled with musical background or generated sequence of control of jets or lights. . In 2015, Feng J., [4] introduced control system of fancy fountain. His system relies on the control functions of PLC to control the inverter. The sound signal supplied by CD or DVD player is compared with a fixed value set up in the PLC and the comparison value is adopted to control the inverter and thus the pump and lights. In 2016, Visconti P., Costantini P. and Cavalera G., [5], proposed a system to drive a musical fountain and it makes the (light and water pump) scenarios synchronized with the musical files. Each file stored in the SD memory and it is related to a specific scenario.

In the present work, the proposed technology focuses on generating unpredictable water jets and colored lighting scenarios automatically based on the sound frequencies

E-mail: e.alim.j.92@gmail.com, alali.jhana@yahoo.com, rsalwaily@gmail.com

and intensity. This fountain system can work perfectly with any sound track and does not depend on a specific track. Therefore, the system gives a great aesthetic in the presentation of scenarios of water jets and light intensity consistent with sound frequencies, which is beneficial to the fountain place and attract large numbers of tourists and visitors and thus it brings economic benefit in addition to aesthetic and self-entertainment.

The proposed fountain has been driven by the outputs of the four frequency bands which are created by designing four IIR digital filters. The filters have worked to pass or attenuate the sound input signal according to the design of them to control the water heads of the submersible pumps and lights intensity of colored light lamps via four VSDs.

2. THEORETICAL BACKGROUND

A. Water Jet Height control

As it is well known, the pump output is defined by two terms. These are the flow quantity Q (m^3/h) and the head H (m). The human being eyes can not sense the flow quantity instantaneously but it can sense the variation of the water jet head, therefore the water jet height will be adopted as the interesting variable. To control this variable or to allow it to follow a certain pattern, one has to adopt either throttle control or pump speed drive control. The first solution is achieved by controlling the orifice of water nozzle and the second one uses AC drive unit to control the speed of the water pump.

In this paper the second solution has been adopted because of its power saving feature. Here the AC drive unit is used to map the analog control signal coming from the IIR filter into water pump head by controlling the frequency and root mean square (rms) value of the voltage applied to the water pump. The water pump output is function of its speed which is function of the applied voltage frequency. So controlling the frequency of the applied voltage leads to control the pump speed and in turn controlling the water head. The pump speed is controlled by controlling the speed of the stator windings revolving magnetic field according to the following scenario:

Practically speaking, the induction motor speed variation from no load to full load is small. Also the shaft speed is close to the speed of the revolving magnetic field but not equal to it. So controlling the field speed results in controlling the motor shaft speed. The revolving field speed Ns is function of the applied voltage frequency (F) according to the following relation:

$$\mathrm{Ns} = 120\ \mathrm{F/P} \tag{1}$$

Where P is the number of poles.

In constant V/F ratio (K_{v2f}) mode of operation of the AC drive unit, the relationship between the coming control signal Vc (which takes a values from 0 DC volt to 10 DC volt in the used HYUNDAI brand of inverters) and the AC drive output voltage frequency value (F) is :

$$\mathrm{F} = \mathrm{K} * \mathrm{Vc} \tag{2}$$

So from 1 and 2, the motor shaft speed Nr can be approximated to :

$$\mathrm{Nr} = \mathrm{K} * \mathrm{Vc} \tag{3}$$

B. Infinite Impulse Response (IIR) Filter

IIR filter or system is a type of the digital filter. The relationship between the input and output signal is the convolution sum and its described in the Equation (4) [6].

$$y(n) = \sum_{i=0}^{M} b_i x(n-i) - \sum_{j=1}^{N} a_j\, y(n-j) \tag{4}$$

where x(n) is the input signal, y(n) is the output signal, b_i and a_j are the filter coefficients.

IIR filter contains forward and feedback paths. The transfer function of IIR filter is given in Equation (5) [6].

$$H(z) = \frac{Y(z)}{X(z)} = \frac{b_0 + b_1 z^{-1} + \cdots + b_M z^{-M}}{1 + a_1 z^{-1} + \cdots + a_N z^{-N}} \tag{5}$$

Figure 1 illustrates the structure of the IIR filter.

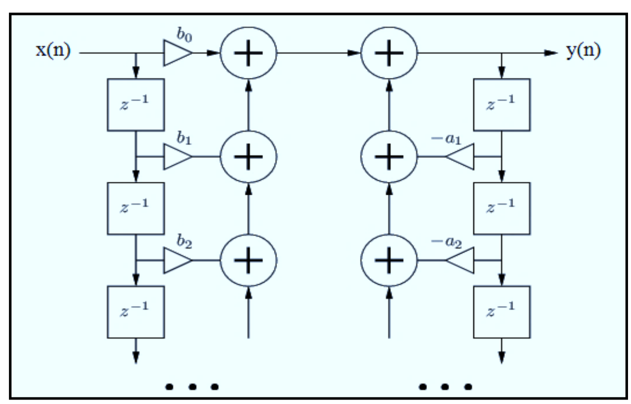

Figure 1. Structure of IIR filter

There are several methods used to design the IIR filter such as bilinear transformation (BLT), pole-zero placement and impulse invariant design [6]. In this paper, the bilinear transformation method has been adopted due to its ease and the digital filter that is produced by the bilinear transformation has the same frequency response characteristics of the analog filter.

The square magnitude response of the low-pass filter is shown in Fig.2.

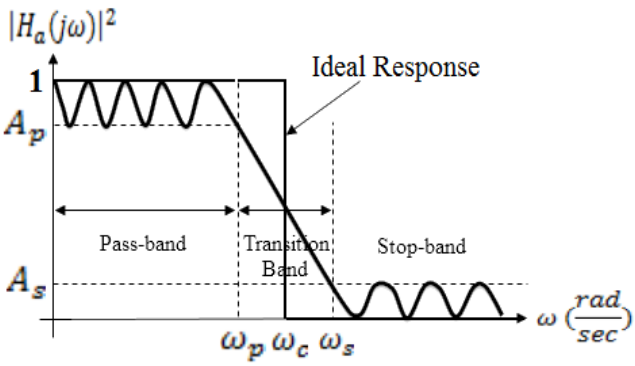

Figure 2. The square magnitude response of the low-pass filter

Where ω_p =pass-band frequency edge, ω_s =stop-band frequency edge

Pass-band region extends from 0 to ω_p. Transition region extends from ω_p to ω_s. Stop-band region located in the interval $[\omega_s, \infty)$. From the Fig.2 above, it is possible to conclude:

$$1 \geq |H_a(j\omega)|^2 \geq A_p \qquad 0 \leq \omega \leq \omega_p \qquad (6)$$

$$0 \leq |H_a(j\omega)|^2 \leq A_s \qquad \omega \geq \omega_s \qquad (7)$$

The square of magnitude response of order N Butterworth analog low-pass filter is given by:

$$|H_a(j\omega)|^2 = \frac{1}{1+\left(\frac{\omega}{\omega_c}\right)^{2N}} \qquad (8)$$

By taking the square of the magnitude response at ω_p and ω_s .

$$\left|H_a(j\omega_p)\right|^2 = \frac{1}{1+\left(\frac{\omega_p}{\omega_c}\right)^{2N}} \qquad (9)$$

$$|H_a(j\omega_s)|^2 = \frac{1}{1+\left(\frac{\omega_s}{\omega_c}\right)^{2N}} \qquad (10)$$

From the equations above, filter order can be derived as follows:

$$\left(10^{\frac{A_p}{20}}\right)^2 \leq \frac{1}{1+\left(\frac{\omega_p}{\omega_c}\right)^{2N}} \qquad (11)$$

$$\left(10^{\frac{A_s}{20}}\right)^2 \geq \frac{1}{1+\left(\frac{\omega_s}{\omega_c}\right)^{2N}} \qquad (12)$$

By simplifying the equations above, we get the order of low-pass filter prototype as given in equation (13) [6].

$$N \geq \frac{\log_{10}\left(\frac{10^{0.1A_s}-1}{\epsilon^2}\right)}{[2\log_{10}(V_s)]} \qquad (13)$$

Where $V_s = \frac{\omega_s}{\omega_p}$ and $\epsilon^2 = 10^{0.1A_p} - 1$. ϵ is the band-pass ripple parameter. V_s for any filter can be computed from the analog filter frequency specification by using equations below [6].

$$V_s = \frac{\omega_p}{\omega_s} \qquad \text{for highpass filter} \qquad (14)$$

$$V_s = \frac{\omega_{sh}-\omega_{sl}}{\omega_{ph}-\omega_{pl}} \qquad \text{for bandpass filter} \qquad (15)$$

$$V_s = \frac{\omega_{ph}-\omega_{pl}}{\omega_{sh}-\omega_{sl}} \qquad \text{for bandstop filter} \qquad (16)$$

The magnitude square of the analog low-pass Butterworth filter with various order is shown in Fig. 3. The shape of the filter near to the ideal response by increasing the filter order.

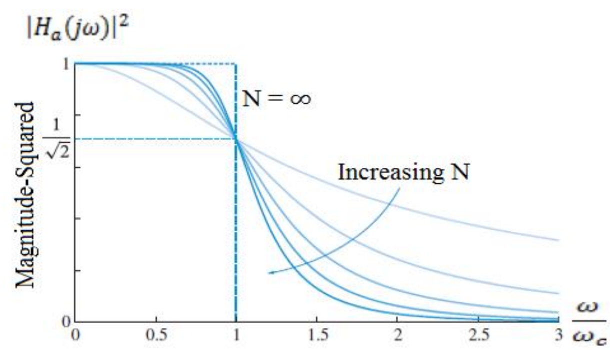

Figure 3. Square magnitude response of the analog low-pass Butterworth filter with various order

The center frequency and the bandwidth of the band-pass filter are given in equation (17) and equation(18) respectively [6].

$$\omega_0 = \sqrt{\omega_{pl}\omega_{ph}} \qquad (17)$$

$$W = \omega_{ph} - \omega_{pl} \qquad (18)$$

Where ω_{pl} =lowest passband cutoff frequency, ω_{ph} =highest passband cutoff frequency, ω_{sl} = lowest stopband cutoff frequency, ω_{sh} =highest stopband cutoff frequeny, ω_0=center frequency and W= filter bandwidth.

The 3dB Butterworth transfer function of analog low-pass prototype with $\epsilon = 1$ can be obtained from Table (1) [6]. Also the transfer function can be found not rely on the table as following:

$$S_i = \omega_c e^{j\left(\frac{\pi(1+2i+N)}{2N}\right)} \quad (19)$$

Where S_i = transfer function poles, i=0, 1, ... , N-1.
The transfer function is given by equation (20)

$$H(S) = \frac{1}{\prod_{i=0}^{N-1}(S - S_i)} \quad (20)$$

TABLE (1) BUTTERWORTH TRANSFER FUNCTION OF THE LOW-PASS PROTOTYPE WITH ϵ=1

N	$H_p(S)$
1	$\frac{1}{S+1}$
2	$\frac{1}{S^2+1.4142S+1}$
3	$\frac{1}{S^3+2S^2+2S+1}$
4	$\frac{1}{S^4+2.6131S^3+3.4142S^2+2.6131S+1}$
5	$\frac{1}{S^5+3.2361S^4+5.2361S^3+5.2361S^2+3.2361S+1}$
6	$\frac{1}{S^6+3.8637S^5+7.4641S^4+9.1416S^3+7.4641S^2+3.8637S+1}$

Using the prototype of the low-pass filter $H_p(S)$ to perform the prototype transformation [6].

$H(S) = H_p(S)\big|_{S=\frac{s}{\omega a}}$ from Low-pass to Low-pass

$H(S) = H_p(S)\big|_{S=\frac{\omega a}{s}}$ from Low-pass to High-pass

$H(S) = H_p(S)\big|_{S=\frac{s^2+\omega_0{}^2}{sW}}$ from Low-pass to Band-pass

The bilinear transformation is given by Equation (21):

$$s = \frac{2}{T_s}\left(\frac{1-z^{-1}}{1+z^{-1}}\right) \quad (21)$$

where, T_s is sampling time (sec) and equal to 1/sampling frequency(Hz)

The non-linear relationship between digital frequency ω_d in the z-plane located on the unit circle and analog frequency $j\omega_a$ in the s-plane, produced by the bilinear transformation.

Substituting $S = j\omega_a$ and $z = e^{j\omega_d T}$ into the equation (21)

$$j\omega_a = \frac{2}{T_s}\frac{1-e^{-j\omega_d T_s}}{1+e^{-j\omega_d T_s}} = \frac{2}{T_s}\frac{e^{\frac{j\omega_d T_s}{2}} - e^{\frac{-j\omega_d T_s}{2}}}{e^{\frac{j\omega_d T_s}{2}} + e^{\frac{-j\omega_d T_s}{2}}}$$

$$= j\frac{2}{T_s}\tan\frac{\omega_d T_s}{2}$$

Pre-warped frequency can be calculated by using equation (22)

$$\omega_a = \frac{2}{T_s}\tan\frac{\omega_d T_s}{2} \quad (22)$$

Design of IIR filter with bilinear transformation requires four stages as shown in Figure 4 [6].

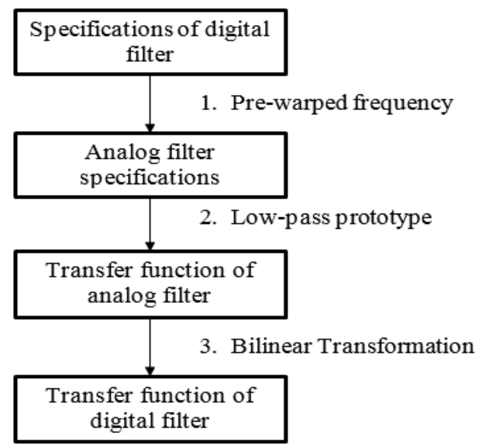

Figure 4. Procedures to design IIR filter with bilinear transformation

3. SYSTEM IMPLEMENTATION

A. Proposed Musical Fountain System

Several devices are linked together to form the musical fountain system. The proposed system consists of a sound source, a sound conditioning circuit, a main programmable control unit, configurable drive unit (AC drives) and infrastructure components. Figure 5 shows the block diagram of electrical connection of the musical fountain system

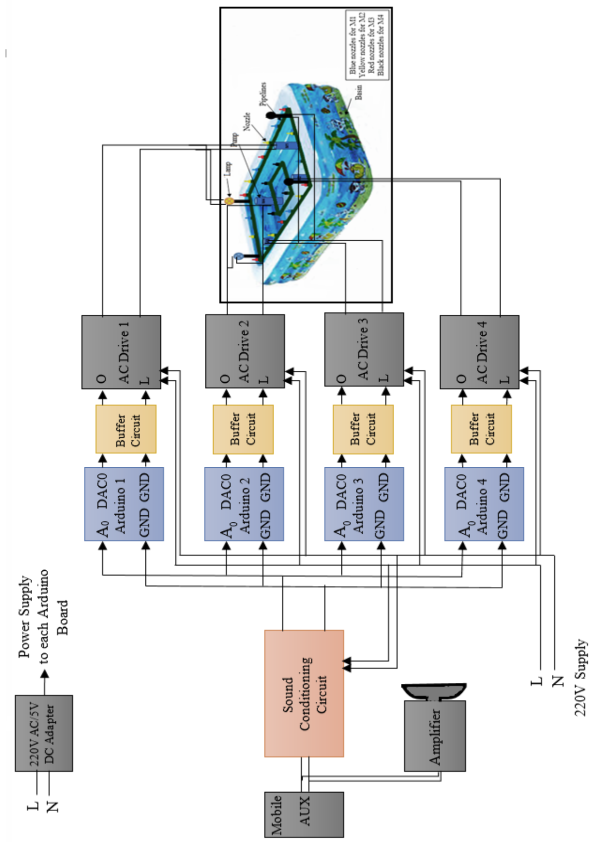

Figure 5. Electrical connection of musical fountain system

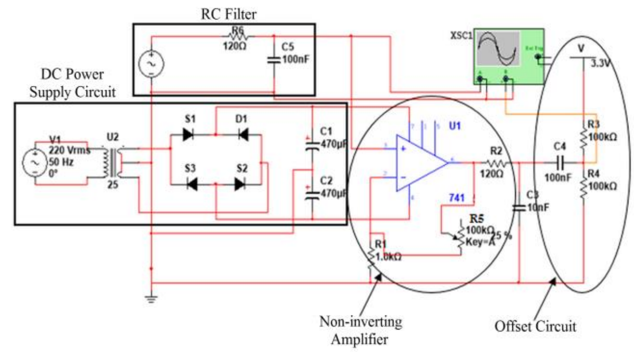

Figure 6. Sound conditioning circuit

i. Sound source

The sound signal can be taken from variety of sound sources such as Mobile, PC, MP3, etc. In the present work, the sound signal is obtained from the sound files stored in Mobile.

ii. Sound Conditioning Circuit

The sound conditioning circuit consists of an RC filter, DC power supply circuit, an amplifier and an offset circuit as shown in Fig. 6. The RC filter has been used to reduce the noise and to limit the sampling frequency. A non-inverting amplifier is constructed using 741 IC which is used to amplify the sound signal. The DC power supply circuit consists of 220/9V center tap transformer and full wave rectifier circuit to give +Vcc and –Vcc to the 741 IC. The offset circuit has been used to make the amplifier output within the defined range of the analog input port of the Arduino.

iii. Programmable Main Control Unit

The programmable main control unit consists of four Arduino type microcontrollers. Each one stands for one IIR filter. The four microcontroller operate in parallel like mode to provide the four harmonics band to control four AC drives. The Arduino that has been used in this work is Arduino Due. This type of Arduino based on the Atmel SAM3X8E ARM Cortex-M3 CPU. Arduino Due is the best Arduino type for audio applications compare with the other Arduino types [7]. Arduino Due have several specifications: 84MHz clock speed, 96kB SRAM, 512k flash memory, 16*12 bits analog input (ADC), 2 analog output (DAC) also with 12 bits resolution (4096 steps), while the other Arduino types have 10 bit resolution (1024 steps) [7]. The input voltage range from 0V to 3.3V and the output voltage range from 0.55V to 2.75V.

iv. Configurable Drive Unit

The configurable drive unit consists of a four variable speed AC drive. AC drive, variable speed drive or variable frequency drive (VFD) is a device used to control the speed of induction motor by changing the motor input voltage and frequency. AC drive consists of five stages. These are AC/DC converter, DC link, inverter stage, control power supply circuits and control circuits as shown in Fig. 7. The AC /DC converter converts the AC supply voltage to DC voltage. The DC link maintains smooth constant DC voltage to the inverter stage. The inversion stage converts the DC voltage to a variable voltage variable frequency output. The control power supply circuits provide power to the control circuits. The control circuits functions are to drive the inversion stage power switches in such away that provide controllable output voltage and frequency, they also provides protection functions in addition to allowing the user to define the mode of operation he or she desires [8].

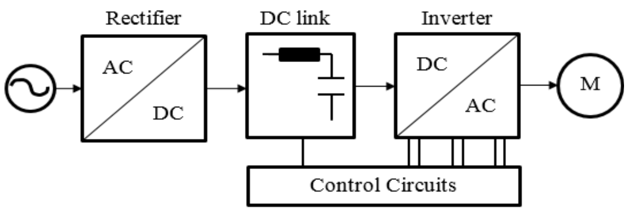

Figure 7. Block diagram of the VFD.

This device is used to map the analog voltage control signal coming from the each Arduino into water pump head and light intensity. When the Arduino output is changed between its minimum to maximum values, each AC drive will control the speed of the submersible pump and light intensity by varying the input voltage and frequency between 0V/0Hz to 220V/50Hz.

v. Buffer circuit

A buffer circuit is used to isolate purpose and to reduce the noise between each Arduino output and the AC drive. The unity gain buffer circuit is shown in Fig. 8.

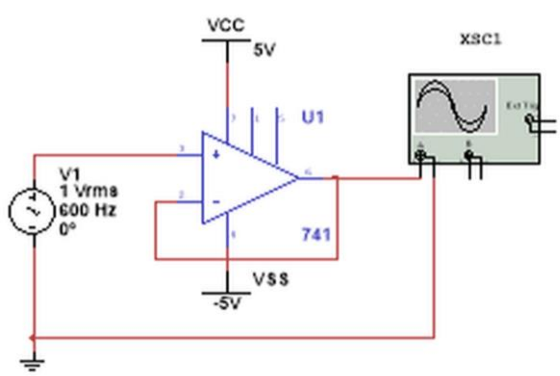

Figure 8. Buffer circuit

vi. Infrastructure Components

Figure 9 shows the infrastructure components of the proposed musical fountain. Infrastructure components must be selected and arranged to suit the required target. These components include four submersible pumps (each one has a set of nozzles), four colored light lamps (each one with a special color) and the basin. The infrastructure components are associated with electrical full control circuit to complete the musical fountain system.

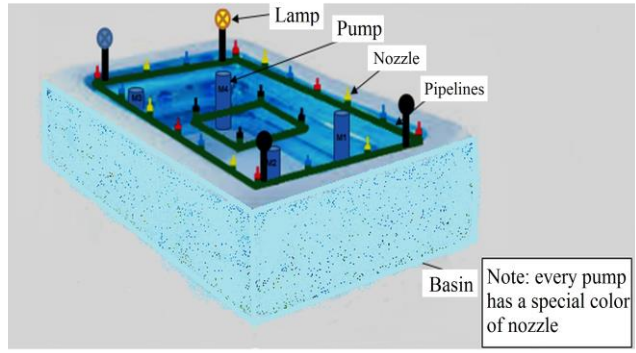

Figure 9. The Structure Components

***1.** Submersible Pump*

In this work, four submersible water pump have been used to generate the required water heads. Each one with the following Specifications: 220/240V 50/60Hz input supply, 0.5 horse power = 0.37KW, 2900 r.p.m rated speed, flow=1.5 m^3/h and head=15m. Figure 10 shows the submersible water pump.

Figure 10. Submersible Pump

***2.** Light Lamp*

Colored lights give the fountain display more pleasure and beauty. A single light with special color is connected in parallel with each submersible pump. When the fountain is operating, the light intensity is synchronized with the audible sound frequencies. The used lamp specifications are 220V supply, 80W power and 30^o beam angle. Figure 11 shows the light lamp.

Figure 11. The lamp-based light

B. System Software

As mentioned above, four Butterworth IIR filter have been designed and implemented here, each one with 4th order. The first filter is designed as a band-pass filter with 489.89Hz center frequency. The second filter is a band-pass filter with 1838.477Hz center frequency. The third filter is also designed as a band-pass filter with 4312.77Hz center frequency. The fourth filter is a high-pass filter with cutoff frequency equals to 6500Hz. Figure 12 shows the flow chart to implement 4th order IIR filter. Points below show the IIR filter algorithm:

1) Acquire the stopband and passband edge frequencies.
2) Calculate the filter order.
3) Compute the transfer function of the analog system.
4) From analog transfer function, find the IIR digital filter transfer function
5) Calculate and draw the IIR filter frequency response.

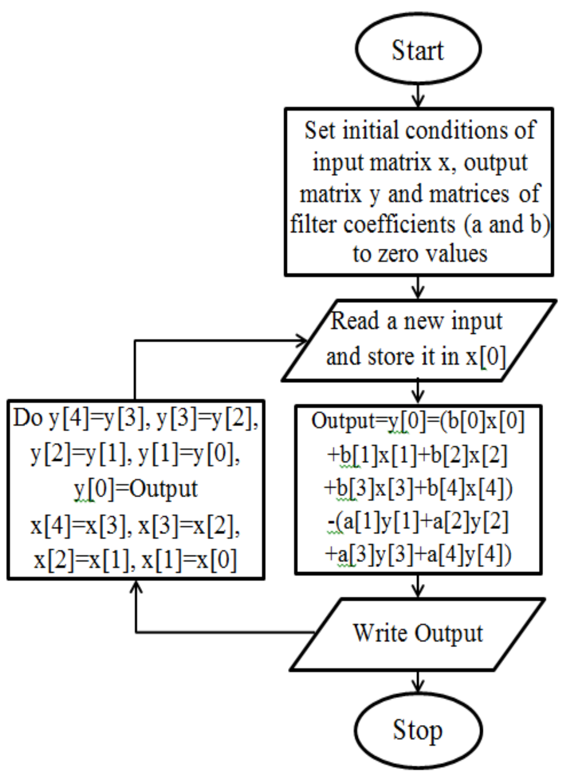

Figure 12. Flow chart of IIR filter implementation

To implement the IIR filter, it is necessary to use two circular buffer. The circular buffer is similar to FIFO (first in first out) queue. The first buffer for the input sound signal and the second buffer for the output signal. Each buffer has been used to store and update the signal sample by sample until the filter order .The buffer length must be equaled to the filter order+1.

The coefficients of each filter can be found by applying the equations in section 2 (B). Also the filter coefficients can be obtained by using MATLAB program as follows: the coefficients of the analog filter is obtained using low-pass to band-pass function ([bb,aa]=lp2pb(nn,dd,w0,bw)) for the band-pass filter and ([bb,aa]=lp2hb(nn,dd,w0,bw)) for the high-pass filter. where bb and aa are the coefficients of the analog transfer function of desired filter, nn and dd are the coefficients of the analog transfer function of low-pass filter and they can be obtained from table 1, w_0 is the center frequency of the desired filter and bw= bandwidth of the desired filter. Then the coefficients of transfer function of the analog filter can be converted to the digital using ([b,a]=bilinear(bb,aa,fs)) where fs=sampling frequency, a and b are the coefficients of the desired digital filter.

4. RESULTS

A. Simulation Results

A complete Simulink model of a fountain system which is simulated using MATLAB program is shown in Fig 13. From the figure, it is clear that the system consist of sound source, four IIR filters, four AC drives units and other blocks to represent the speed and r.m.s voltage of each submersible induction motor.

The sound input block is used to apply the sound signal to the four IIR filter blocks by bringing an existing sound file in the computer to MATLAB SIMULINK. Then, a four IIR filters have been used to decompose the original sound signal (supplied by the sound input block) into four signals, each one carries its own frequency according to its filter specifications. These filters can be simulated using discrete filter block that is existed in the simulink library or by building the IIR filter using adders, delay units and gains. Also the filters can be programed in MATLAB language in each MATLAB function block. The filtering output signal of each filter which obtained during the design of the filters with 44100Hz sampling frequency is shown below where the numbers multiplied by x(n) are the feedforward filter coefficients [represent the numerator of the discrete block] and the numbers multiplied by y(n) are the feedback filter coefficients [represent the denominator of the discrete block] (see section 3 B to know how the coefficients can be obtained).

y1(n)=0.001207407685083x(n)-0.000000000000001x(n-1) 0.002414815370162x(n-2)-0.000000000000004x(n-3)+0.001207407685084x(n-4)+3.889835578715863y1(n-1) -5.684324474515597y1(n-2)+3.698630167076128y1(n-3)-0.904163840901860y1(n-4)

y2(n)=0.007566658839219x(n)-0.000000000000001x(n-1)-0.015133317678438x(n-2)-0.000000000000001x(n-3)+0.007566658839220x(n-4)+3.611385659509821y2(n-1) -5.012208839482432y2(n-2)+3.166283304565194y2(n-3)-0.769566147769958y2(n-4)

y3(n)=0.032771335536785x(n)+0.000000000000001x(n-1)-0.065542671073572x(n-2)+0.000000000000002x(n-3)+0.032771335536784x(n-4)+2.790836025496597y3(n-1)

$$-3.406276841084701y3(n-2)+2.070425799389155y3(n-3)-0.557732577757384y3(n-4)$$

$$y4(n)=0.282257225635328x(n)-1.129028902541315x(n-1)+1.693543353811973x(n-2)-1.129028902541317x(n-3)+0.282257225635330x(n-4)+1.611784177898776y4(n-1)-1.318360498283419y4(n-2)+0.505991136262026y4(n-3)-0.079979797721033y4(n-4)$$

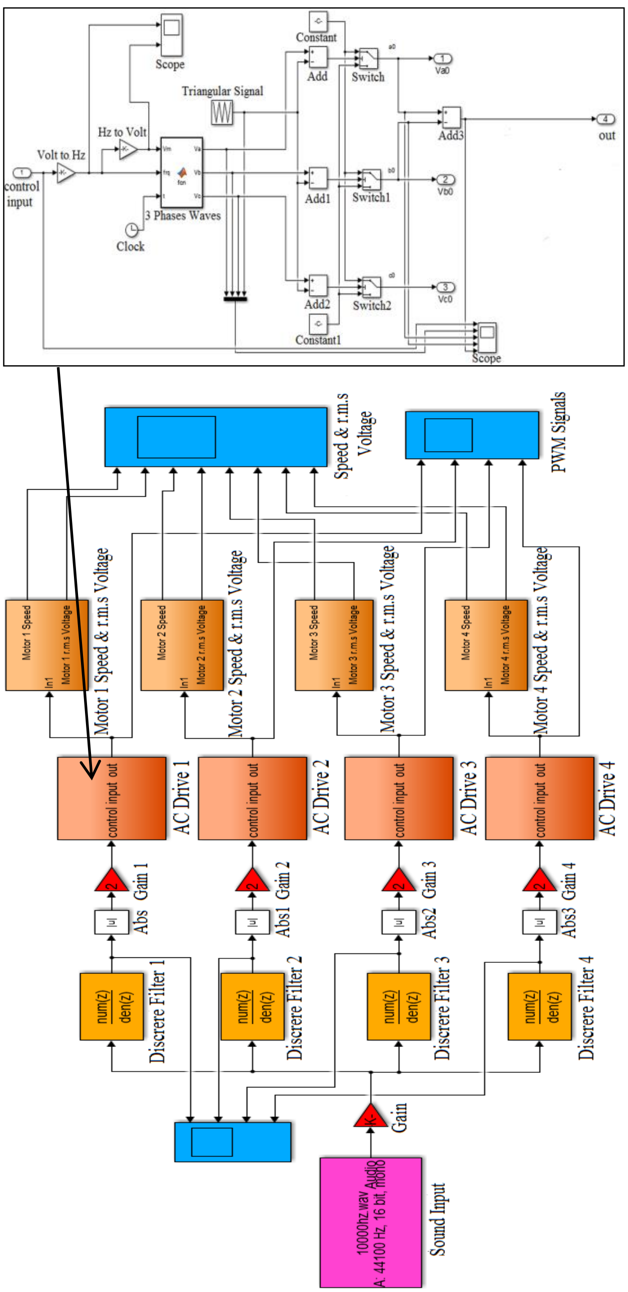

Figure 13. MATLAB/SIMULINK of a complete simulink model of the proposed fountain system

The AC drive blocks used to control the speeds of submersible induction motors (submersible pumps) and the light lamps intensity according to the voltage control signal from the IIR filters. The speed of the pump is changed when the magnitude of the control signal changes. Each filter output is connected to a one AC drive unit which is linked to a one pump and one light. AC drive block is simulated to give the output modulated signal from (0V and 0Hz) to (220V and 50Hz) when the filter output is changed from (0V to 2.25V). To obtain the filter output in this range, the absolute value is taken for the origin filter output and then it is multiplied by 2. Then from the simulation of AC drive block shown in Figure 13, the control signal is multiplied by a specific value to map it to the frequency. Then, the frequency is mapped to the voltage range that possesses a specific rms value. The voltage and frequency are adopted to produce three phase signals. Each one is compared with a triangular signal has a suitable magnitude and a 4kHz frequency to overcome the over-modulating to generate pulse width modulation (PWM) signals will be applied to the power electronics components of inverter to produce the AC output signals with controlled frequency.

To get the pump's r.m.s voltage, r.m.s block is used to convert the output modulated signal of the AC drive to the root mean square voltage. In order to represent the pump's speed, MATLAB function block is used and the equation of speed is written in it.

Figure 14 presents the magnitude response of the four IIR filters.

Figure 15 shows the filters outputs when a musical track is applied as an input. In Figure 15, it can be noted that all the filters have an outputs with different amplitudes and frequencies according to the filters specifications and frequencies in musical sound track.

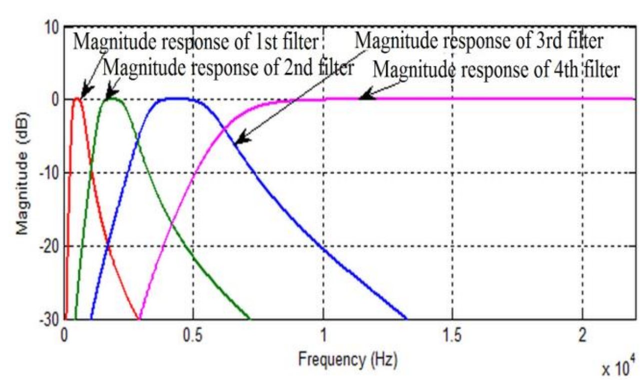

Figure 14. Magnitude response of IIR filters

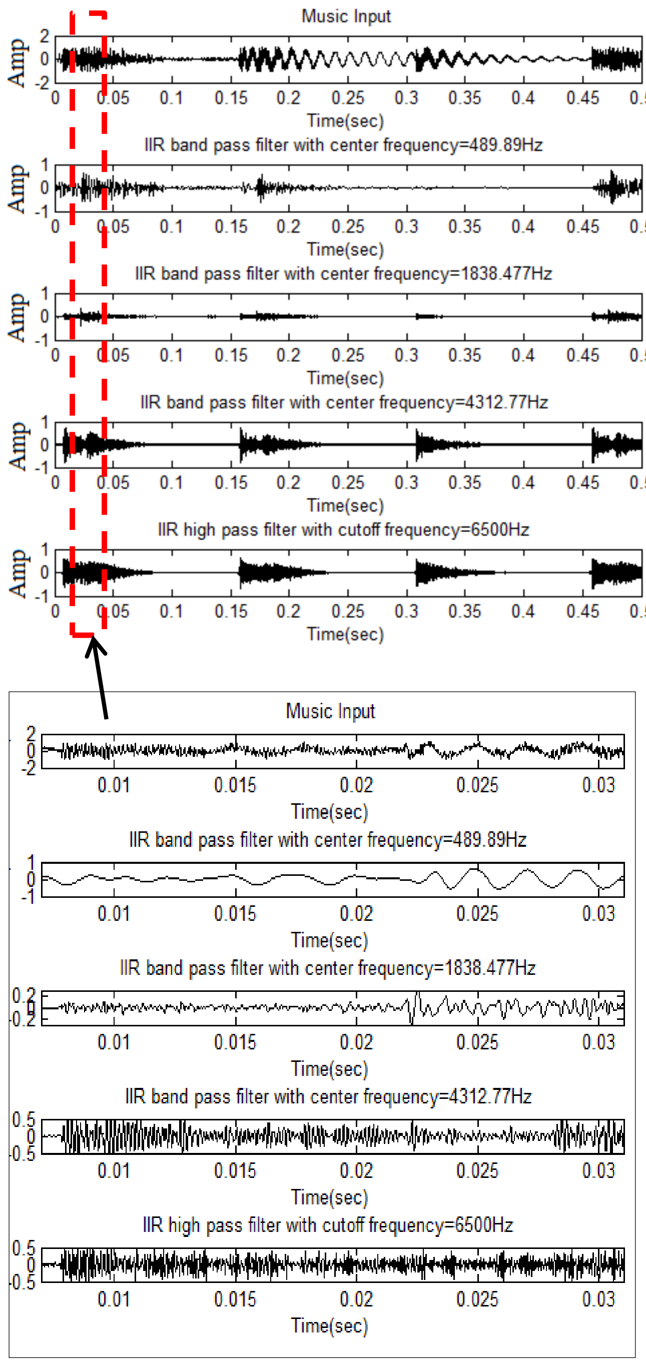

Figure 15. Filters outputs of music input

The outputs of the filters when a 500Hz, 750Hz, 1000Hz, 1500Hz, 2000Hz, 2500Hz, 3500Hz and 5000Hz tones are applied as inputs are illustrated in Fig. 16. From the figure 16, in can be noted that each filter is allowed to pass only a tone that has a frequency belong to the filter pass-band region. It is possible to observe, the output of the fourth filter is almost zero because there is no tone that have a frequency equals or above 6500Hz in the input.

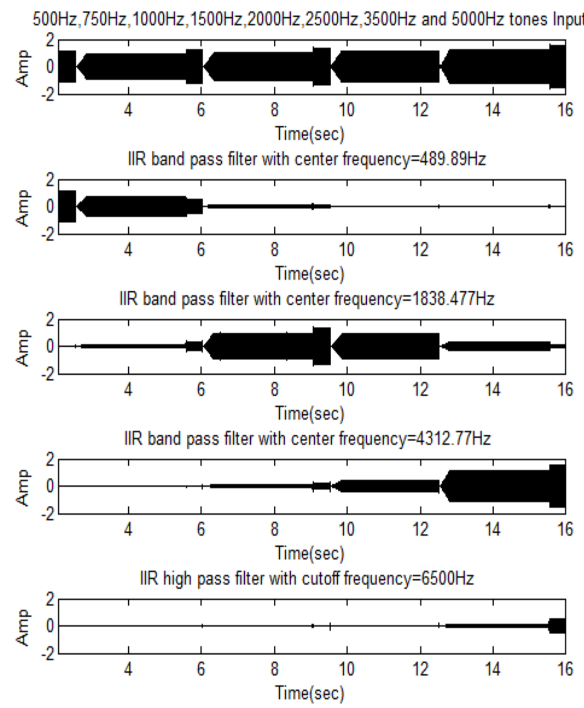

Figure 16. Filters outputs when applied 500Hz, 750Hz, 1000Hz, 1500Hz, 2000Hz, 2500Hz, 3500Hz and 5000Hz tones

The sound frequencies of the signals in Figure 15 and Figure 16 can be obtained and observed using Fast Fourier Transform (FFT) technique as shown in Figure 17 (mapping of Figure 15 into frequency domain) and Figure 18 (mapping Figure 16 into frequency domain).

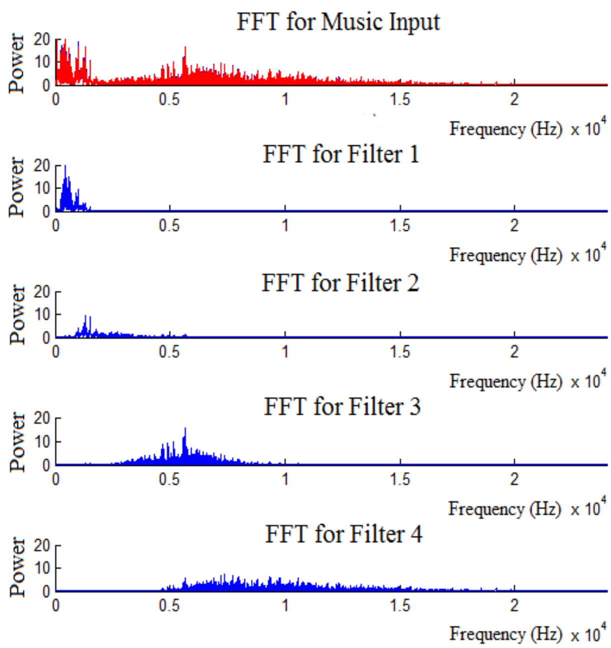

Figure 17 Frequency components of the signals shown in Fig.15.

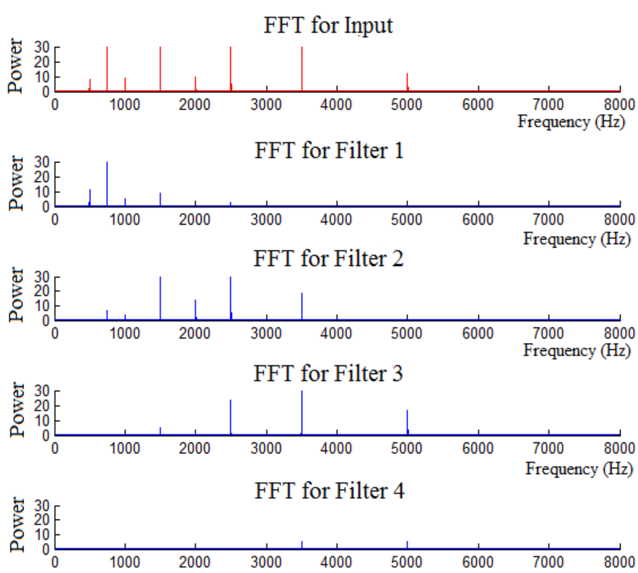

Figure 18 Frequency components for the filters outputs with an input that is shown in Fig.16.

Figure 19 shows the speeds and r.m.s voltages of each submersible induction motor when a 500Hz, 750Hz, 1000Hz, 1500Hz, 2000Hz, 2500Hz, 3500Hz and 5000Hz tones are applied as inputs. The speeds and r.m.s voltages of the four submersible induction motors when the input is a track of musical sound are shown in Fig. 20.

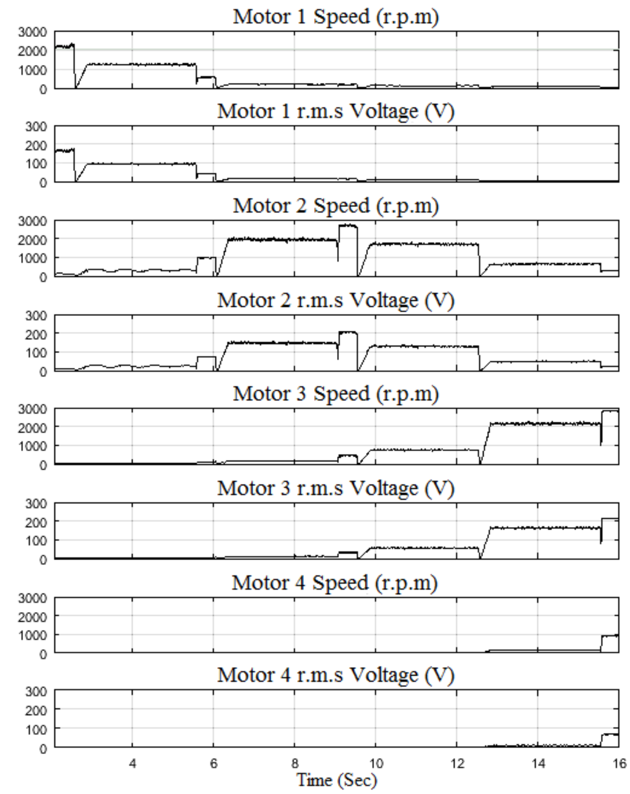

Figure 19. Speed and r.m.s voltage of each motor for a 500Hz, 750Hz, 1000Hz, 1500Hz, 2000Hz, 2500Hz, 3500Hz and 5000Hz tones input

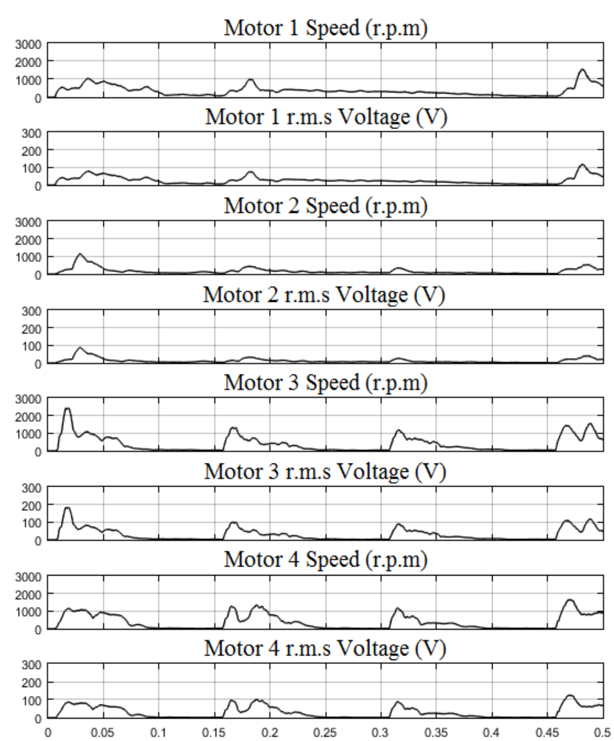

Figure 20. Speed and r.m.s voltage of each motor when the input is a track of musical sound

Figure 19 and Figure 20 illustrate that all submersible induction motor (submersible pump) have worked in the different speeds and they have different input voltages according to the power of the input tones frequencies.

The relationship curves between the control voltage signal and the output (voltage or frequency) are the straight lines as shown in Fig. 21.

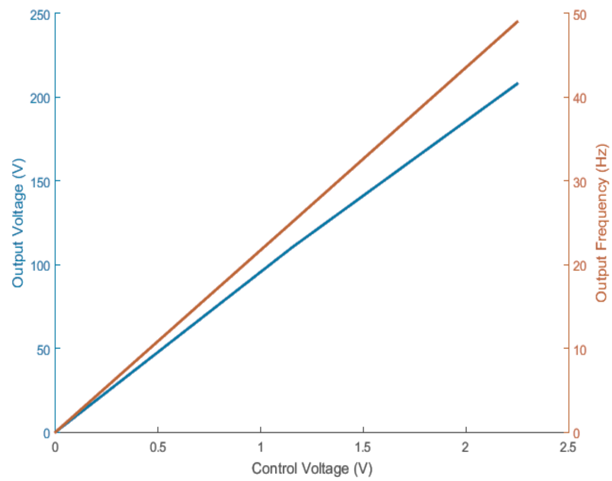

Figure 21. Output voltage and frequency versus control voltage

The pump speed is directly proportional to the control voltage signal and the relationship between them is the straight line as shown in Fig. 22.

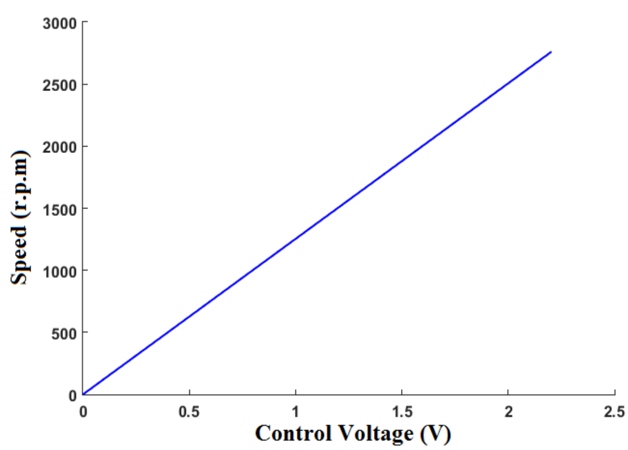

Figure 22. Pump speed versus control voltage

The relationship between the speed of any pump (note that: in this work, each pump have a six nozzles) and the water head through these nozzles is shown in Fig.23. From this Figure, it can be noted that the water head is directly proportional to the square of the pump speed.

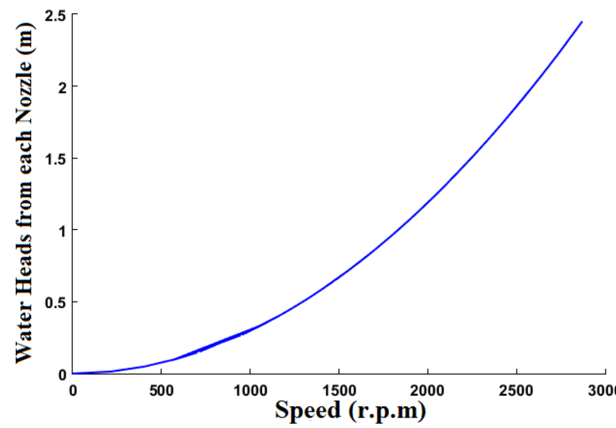

Figure 23. Water head versus pump speed

B. Practical Results

A complete practical circuit of the full control circuit of the musical fountain is shown in Fig. 24.

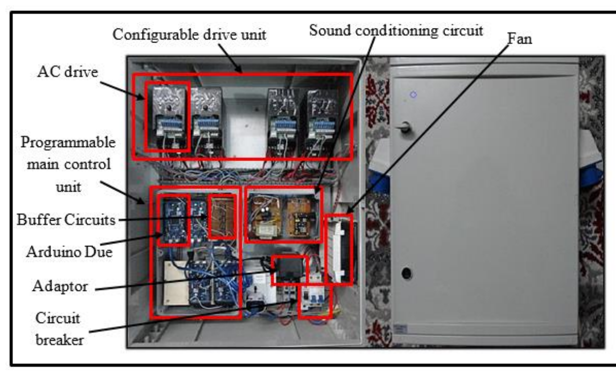

Figure 24. A complete practical circuit of musical fountain

Figure 24 shows the practical circuit of the proposed musical fountain. This circuit consists of four Arduino, four variable speed AC drive units, sound conditioning circuit, buffer circuits and other components. The outputs of this circuit are connected to the four pumps and four colored light lamps, each pump and each lamp work in a certain band of frequencies that is defined through the design of the filter.

The Arduino DAC represents the filter output and it is connected to the analog input terminal of the AC drive unit. The outputs of the AC drives are the outputs of the overall

Figure 25 shows the filters outputs of a tone of 1500 Hz. Figure 26 shows the filters outputs with a tone of 10000 Hz as an input. The filter outputs when a music is applied as an input is shown in Fig.27.

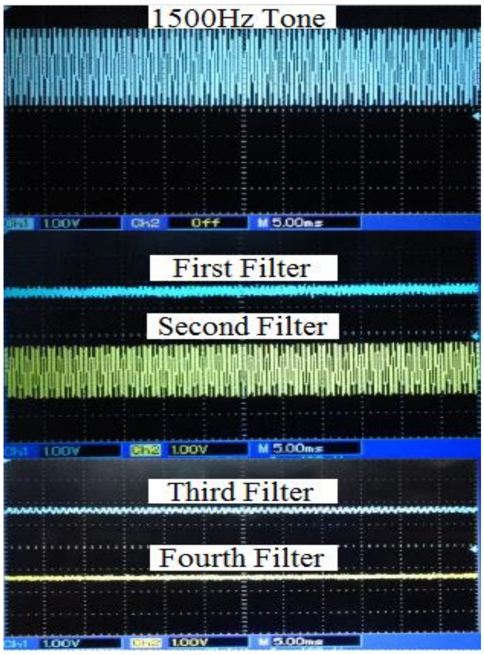

Figure 25. Filters outputs for 1500Hz tone input

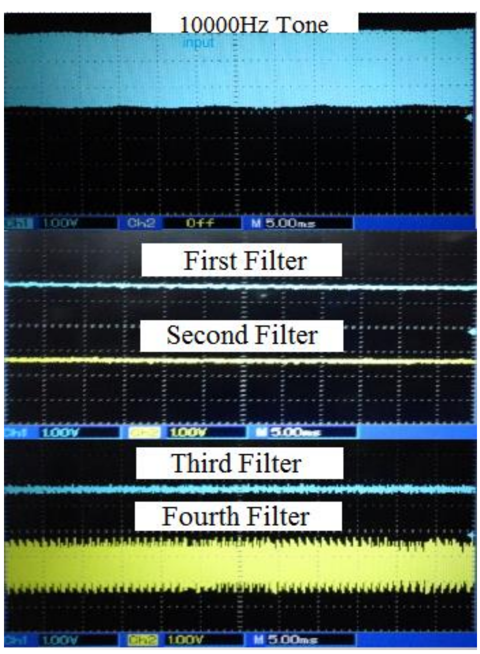

Figure 26. Filters outputs for 10000Hz tone input

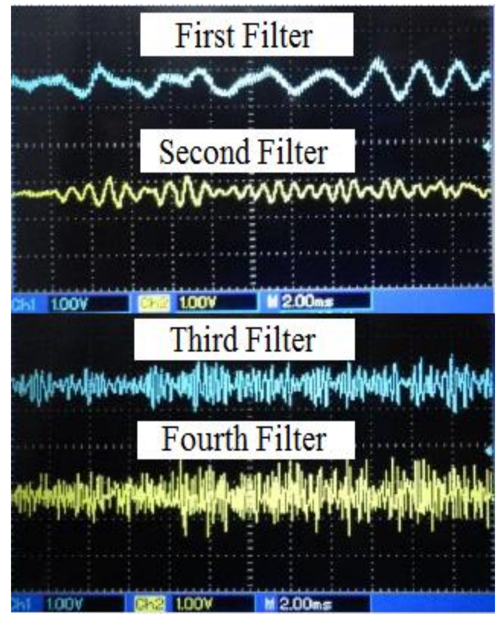

Figure 27. Filters outputs for music input

In Figure 25, only the second filter allows to pass a 1500Hz tone input and all the other filters reject this input due to the second filter has been designed as a band-pass filter with a pass-band region from 1300Hz to 2600Hz.

In Figure 26, the 10000Hz tone is passed through the fourth filter only and all the other filters does not pass this input because of the fourth filter has been designed as a high-pass filter with a 6500Hz cutoff frequency.

The filters outputs in Figure 27 have different amplitudes according to the filters specifications to pass or attenuate the amplitude of the musical input sound frequencies.

Figure 28 shows the voltage and frequency of the first submersible motor versus the tones frequencies. The speed and water head of the first submersible motor versus the tones frequencies are shown in Fig. 29. Figure 30 illustrates the relation between frequency and voltage of the second submersible motor with the input tones frequencies. The speed and water head of this motor versus the tones frequencies are shown in Fig. 31. Figure 32 and Figure 33 show the (voltage and frequency) and (speed and water head) of the third submersible motor respectively versus the tones frequencies. Figure 34 illustrates the relation between frequency and voltage of the fourth submersible motor with the input tones frequencies. The speed and water head of this motor versus the tones frequencies are shown in Fig. 35.

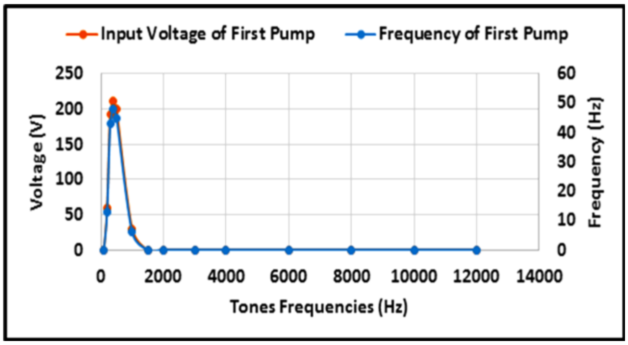

Figure 28. Voltage and frequency of the first motor versus tones frequencies

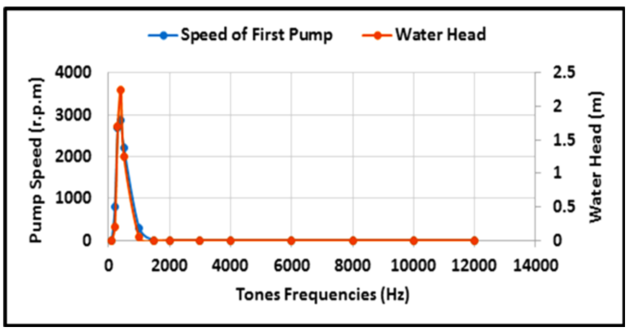

Figure 29. Speed and water head of the first motor versus tones frequencies

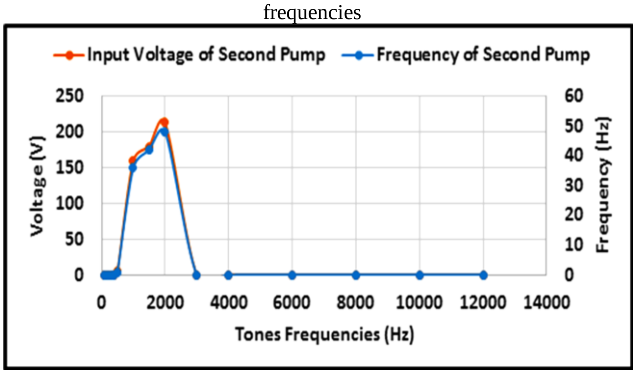

Figure 30. Voltage and frequency of the second motor versus tones frequencies

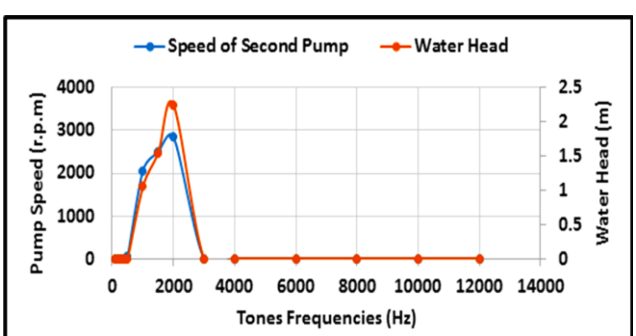

Figure 31. Speed and water head of the second motor versus tones frequencies

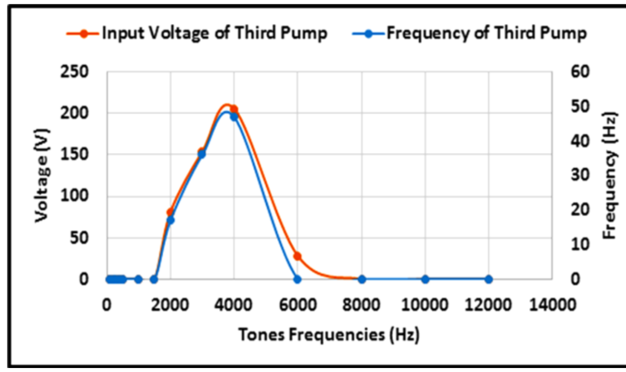

Figure 32. Voltage and frequency of the third motor versus tones frequencies

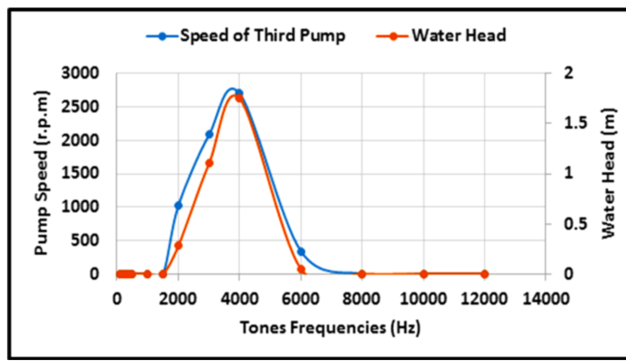

Figure 33. Speed and water head of the third motor versus tones frequencies

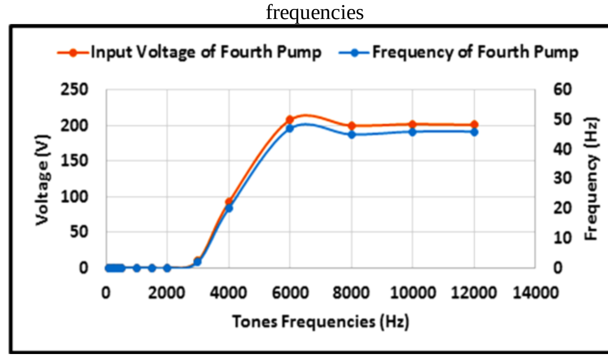

Figure 34. Voltage and frequency of the fourth motor versus tones frequencies

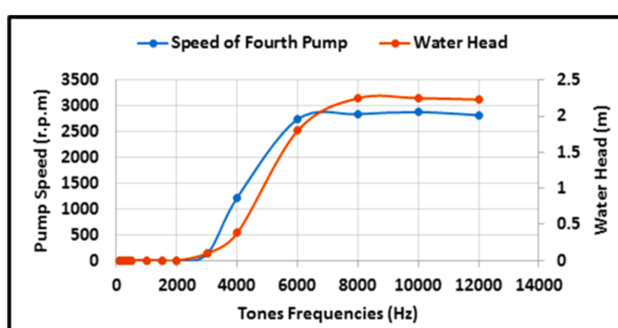

Figure 35. Speed and water head of the fourth motor versus tones frequencies

From Figure 28, it can be noted that the first submersible motor have a high voltage and frequency roughly when the tones frequencies nearly between 300Hz and 800Hz due to this motor depends on the first filter and this filter has been designed to have a frequency band from 300Hz to 800Hz. Also, the first submersible motor have a high speed and water head when tones frequencies nearly between 300Hz and 800Hz as shown in the Figure 29 for the same reason.

The voltage and frequency of the second submersible motor have approached to the rated values when the tones frequencies between 1300Hz and 2600Hz as shown in Fig.30 and the speed and water head of this submersible motor is also high in this range due to this submersible motor is related to the second filter which have pass band region from 1300Hz to 2600Hz as shown in Fig. 31.

From Figure 32 it can be seen that the third submersible motor have a high voltage and frequency roughly when the tones frequencies approximately between 3100Hz and 6000Hz due to this motor depends on the third filter which is designed to have a frequency band from 3100Hz to 6500Hz. Figure 33 shows also the third submersible motor have a high speed and water head when tones frequencies nearly between 3100Hz and 6000Hz for the same reason.

The voltage and frequency of the fourth submersible motor have approached to the rated values when the tones frequencies above a 6000Hz as shown in Fig. 34 due to the fourth filter has been designed to have a bandwidth equal or above 6500Hz. Also, From Figure 35, it can be noted the speed and water head of the fourth submersible motor take a high values in the range of tones frequencies above 6000Hz for the same reason.

Figure 36 shows the straight line relationship between the control input voltage and the pump frequency which has been extracted practically from the submersible pump when a voltage control signal with a range from 0.55V to 2.75V is applied as an input. Also the r.m.s voltage of the pump versus this voltage control signal shown in Fig.37

Practically, the relationship of the pump speed and the voltage control signal with a range from 0.55V to 2.75V is applied as an input as shown in Fig. 38. Figure 39 shows the relationship curve between the pump speed and the water head through each nozzle when the pump have a six nozzles to drive the water up in the air.

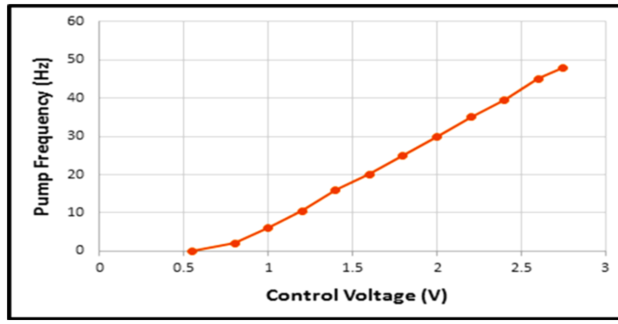

Figure 36. Pump input frequency versus voltage control signal

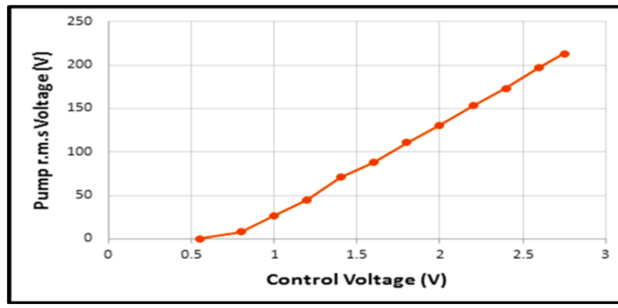

Figure 37. Pump input r.m.s voltage versus voltage control signal

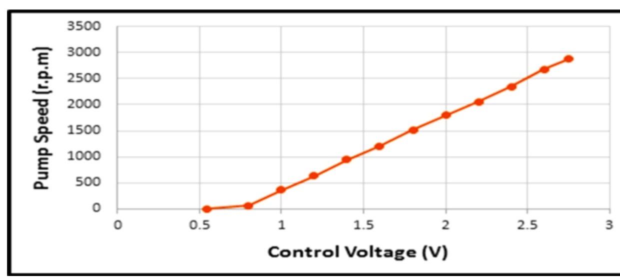

Figure 38. Pump speed versus control voltage

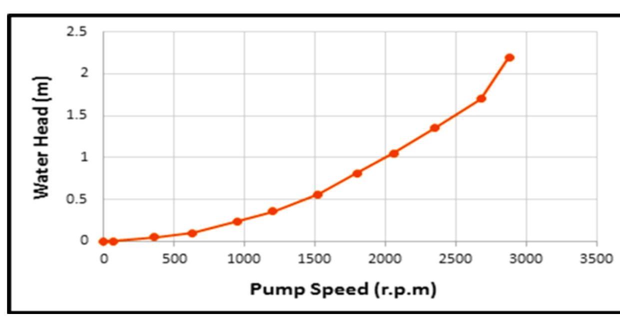

Figure 39. Water head versus pump speed

C. Comparison between Practical Results and Theoretical Results

The practical results are satisfactory and largely approached to the simulation results and the little difference between them is the result of some practical problems occurring in processing of discrete signal such as processing time, memory or sampling speed.

5. CONCLUSION

In the current work, the detailed design of a musical water fountain that shows unpredicted water jets patterns based on the sound harmonic bands has been simulated and implemented. According to the bands output which are created using IIR filters, this fountain exhibits different altitudes of the water jet and different colored light intensity by synchronizing the submersible pumps and colored light lamps with the sound frequencies and the power of these frequencies.

The simulation results are very close to the practical ones and the little difference between the results is caused by the problems faced while processing the digital signal such as sampling time or processing time.

Practically, the fountain is tested with different sound files (it is not specified to work with a certain sound file(s)) and it has worked according to the planned target.

In this fountain, four IIR digital filters (each one with a special band of frequencies) have been used to decompose the mother sound signal into four signals, each one has been used to control the water head of the submersible pump and the light intensity of one lamp with a particular color. IIR filters have been used due to these filters are needed to a less order to be sharp cutoff, do not need more processing time and more memory in software in comparison with the other type of the digital filters.

REFRENCES

[1] Said S, "Microcontrolled water fountain:a multidisciplinary project", Int. J. Engng Ed. Vol. 20, No. 4, pp. 654-659, 2004

[2] Min-Joon Yoo and In-Kwon Lee, "Automatic musical fountain scenario generation using musical information analysis", Proceedings of the International Computer Music Conference (ICMC 2009), Montreal, Canada, August 16-21, 2009.

[3] copindean, R. A Munteanu and F. Dragan, "Artesian fountain with PLC control", Acta Electrotehnica, Vol. 55 Issue,2014.

[4] Jiang F., "Control System of Fancy fountain based on PLC", International Core Journal of Engineering, Vol.1, No.5, pp73-74, 2015.

[5] Visconti P., Costantini P. and Cavalera G., " Smart Electronic System for Dancing Fountains Control Capable to Create Water and Lighting Scenarios Synchronized with Music Track", Asian Research Publishing Network (ARPN) Journal of Engineering and Applied Sciences, Vol. 11, No. 9, pp5669-5675, 2016.

[6] Li Tan, "Digital Signal Processing Fundamentals and Applications", DeVry University Decatur, Georgia, 2008.

[7] Mike Cook, " arduino music and audio projects", first edition, 2015

[8] Malcolm B.," Practical Variable Speed Drives and Power Electronics", Newnes in an imprint of Elsevier, first edition, 2003.

Ali M. Jasim Received the Msc degree in Electrical Engineering/ Power and Machines from the University of Basrah in 2017.
Ali's areas of interest for research include control systems, power electronics, electrical power, renewable energy and programming

Jawad R. Mahmood Received the phD degree from university of basrah.basrah. Jawad Radi currently works at the Department of Electrical Engineering, University of Basrah. jawad does research in electrical power Engineering, power electronics, power converters, inverters, renewable energy technigies and PLC programming.

Ramzy S. Ali Received the phD degree from university of saint-petersburg state polytechnic, Russia. Ramzy Ali currently works at the Department of Electrical Engineering, University of Basrah. Ramzy does research in Engineering Education, Control Systems Engineering and Electrical Engineering.