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Audio Digital Equalizer 2/3 Octave Using OMAP-137 as the Signal Processor with Digital

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ABSTRACT

Background: This project was about design and implementation of an audio digital equalizer with digital control using computer as the controller. The equalizer had 2/3 octave band separation resulting into 15 bands across the entire audio spectrum frequency and with left and right channel with total of 30 bands. The band separation was based on the well-known standard defined by IEC 61260/1260 year 1995, ISO 266 year 1997 and ISO 3 year 1973. This specification also incorporated the filter $\pm 3\text{dB}$ bandwidth for each filter band of the equalizer. The hardware to realize the equalizer was a special purpose digital signal processing system from Texas Instruments. The processor for the equalizer was the OMAP137 in the form of a development board. This system had several features which were very essential for digital signal processing such as fast central processing unit to process both integer and floating point numbers, embedded analog to digital converter (ADC), digital to analog converter (DAC) and universal asynchronous receiver/ transmitter (UART) to connect to a computer. The main part of this equalizer was the bandpass filters to manage the frequency response of the digital equalizer based on the second order Butterworth with transfer function of $H(s) = (s^2+1)^{-1}$. The transformation of the filter from the analog s-plane into the digital z-plane used the bilinear Z-transform which had been shown to have stable cut-off characteristics across all frequency with different sampling frequency. The filter coefficients were computed and adjusted using sampling frequency of 48 kHz and 16-bit-ADC. Computation of the filters was implemented in 32-bit integer formats to expedite the results and ensure there is no overflow and then truncated back into 16-bit integer to match the capability of the 16-bit-DAC. All of the 15 channels of the filters were computed in parallel and checked for the characteristics of the filters as an equalizer as opposed to single bandpass filter. DSP system was also checked for the capability to perform all 15 channels of the equalizer and see if the resulting system works as desired. This design approached was for both channels and then tested together to ensure real-time performance for all filters with the sampling rate of 48 kHz. Visual control of the equalizer was via UART port of the system connected to a computer mimicking similar to the view of a graphic equalizer with few added buttons such as bypass, duplicate left-right channel, and undo. The setting can be downloaded into the DSP system and stored permanently so when the equalizer was turned on, it would be at the condition of the last setting. The output of the DSP system was observed and compared using controlled input and then observed to ensure the performance was similar to the calculated value. The final test was to observe the performance for each bandpass filter at each different audio channel with music using auditory observation. The overall system performance was also evaluated using common music and the audio output evaluated and compared with common equalizer and the result was very similar.

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INTRODUCTION

Equalizer is a very important part of a sound system. This component is responsible for changing the sound characteristic of the sound surrounding by controlling the frequency output of the system. With good equalizer and proper settings, the quality of the sound can be improved such as limitation of other audio components and the condition/ limitation of the room.

Based on the type of setting the parameters of the equalizer, there are three types of equalizer which are shelving equalizer, parametric equalizer, and graphic equalizer. Graphic equalizer usually has a slider button to

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change the amplitude of the filter at certain frequency. By sliding the button at different frequency, the sliders will form a graphic with frequency as the horizontal axis. A graphic equalizer basically is collection of bandpass filters with different cut-off frequency and bandwidth.

Typical graphic equalizers are divided into three types based on the frequency separation in octave. An equalizer with 1 octave separation has bandpass filters with 1 octave bandwidth, therefore for the entire audio spectrum frequency, there are eight bandpass filters. Professional equalizer usually has 2/3 octave bandwidth with a total of 15 bandpass filters across the audio spectrum frequency and a 1/3 octave equalizer has 31 bandpass filters.

Typical equalizers found are equalizers with analog electronic components that implement the bandpass active filters. With many analog electronic components, there are many limitations such as lack of repeatability of analog components, wear and tear of the moving components of the equalizer, noise due to heating and noise due to electrostatic of components.

In this work, a digital equalizer is proposed and built using digital components while at the same time retain the look of the analog equalizer. This is a 2/3 octave equalizer stereo (there will be left and right channel) with gain of ± 6 dB for each bandpass filter and also a general/ total amplification/ reduction of ± 6 dB. The equalizer will have the look and feel of an analog graphic equalizer with 15 filters for each channel.

Implementation of the graphic equalizer:

Implementation of the graphic equalizer is to use a digital signal processing board with embedded analog-to-digital converter (ADC) and digital-to-analog converter (DAC). The system already has two ADC and two DAC for the left and right channel of audio signal conversions. All the converters used in this system have 16-bit resolution and this resolution is used for all the work here.

The system board used here is the Texas Instrument OMAP-137 evaluation board with the ADC and DAC included in the board. Figure 1a shows the photograph of the board while Figure 1b shows the functional block diagram of the system. The input channels of the system are compatible with audio signals for the line-in and also for line-out as shown in the figure. The converters can be programmed using predetermined sampling frequency of 48 kHz or 96 kHz which is the standard for professional audio system. For this work the sampling frequency of 48 kHz is chosen (Spectrum Digital, 2009; Texas Instrument).

Channel separation for each bandpass filter is calculated using standard for 2/3 octave audio equalizer which is IEC 61260/1260:1995, ISO 266:1997, ISO 3:1973 (International Electrotechnical Commission, 1995). The channel center frequency, lower and upper cutoff frequencies are tabulated on Table 1 for convenience and notice that although the bandwidth of the each filter increases with the increase of the center frequency in Hz, but the relative width with respect to the value of the center frequency stays the same.

Table 1: Frequency for every channel of 2/3 octave graphic audio equalizer.

Center frequency (Hz)	Lower cutoff frequency (Hz)	Upper cutoff frequency (Hz)	Center frequency (Hz)	Lower cutoff frequency (Hz)	Upper cutoff frequency (Hz)
25	19.95	31.62	1000	794.33	1258.93
40	31.62	50.12	1600	1258.93	1995.26
63	50.12	79.43	2500	1995.26	3162.28
100	79.43	125.89	4000	3162.28	5011.87
160	125.89	199.53	6300	5011.87	7943.28
250	199.53	316.23	10000	7943.28	12589.25
400	316.23	501.19	16000	12589.25	19952.62
630	501.19	794.33			

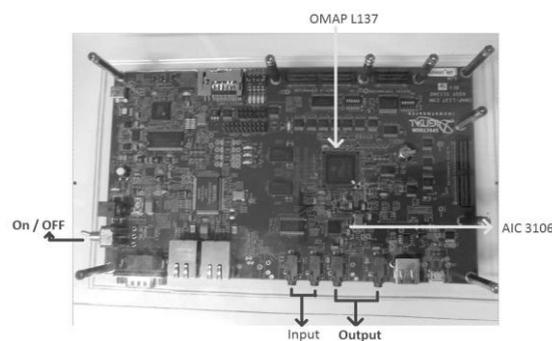


Fig.1(a): Photograph of the board used in this work.

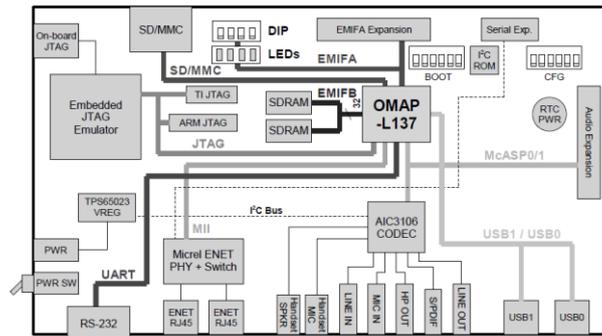


Fig. 1(b): Functional block diagram of the board used in this work.

The digital filter designed for this work is based on the Butterworth filter first order. The transfer function of the filter is:

$$H(s) = \frac{1}{s+1} \tag{1}$$

The transformation of the first order Butterworth filter from the s-plane into the z-plane of band pass filters uses the bilinear z-transform which is widely known and after the transformation, the transfer function becomes in the z-domain as shown in (2) (Kahrs and Brandenburg, 2002; Zölzer, 2002; Spanias, *et al.*, 2006):

$$H(z) = \frac{b_0 + b_2 z^{-2}}{1 + a_1 z^{-1} + a_2 z^{-2}} \tag{2}$$

Based on the result of (2) and using the direct form I structure to implement the digital filter as shown in Figure 2 for the filter implementation will result into (3)

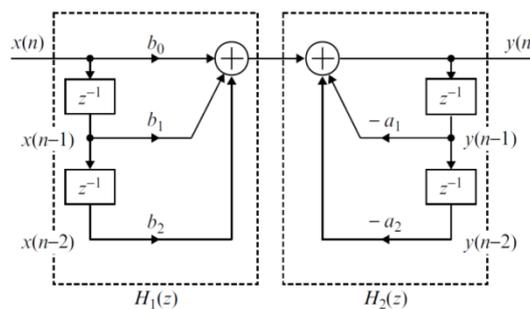


Fig. 2: Block diagram of Direct Form I Structure.

And in the time domain, the function becomes:

$$y(n) = b_0 x(n) + b_2 x(n - 2) - a_1 y(n - 1) - a_2 y(n - 2) \tag{3}$$

Table 2: Filter coefficient after calculation of at different center frequency.

Center Frequency	b ₀	b ₂	a ₁	a ₂	Center Frequency	b ₀	b ₂	a ₁	a ₂
25	-1.99846	0.99847	0.0007632	-0.00076	1000	-1.92435	0.94096	0.0295195	-0.02951
40	-1.99755	0.99758	0.0012094	-0.00120	1600	-1.86703	0.90798	0.0460109	-0.04601
63	-1.99610	0.99617	0.0019147	-0.00191	2500	-1.75810	0.85782	0.0710897	-0.07108
100	-1.99377	0.99394	0.0030316	-0.00303	4000	-1.54525	0.78309	0.1084565	-0.10845
160	-1.98998	0.99041	0.0047966	-0.00479	6300	-1.12860	0.67469	0.1626542	-0.16265
250	-1.98377	0.98484	0.0075802	-0.00758	10000	-0.35899	0.52229	0.2388550	-0.23888
400	-1.97339	0.97608	0.0119614	-0.01196	16000	0.78600	0.31314	0.3434306	-0.34343
630	-1.95566	0.96235	0.0188271	-0.01882					

Table 2 shows the values of the coefficients after the calculations for the implementation of equation. Implementation of the band pass filters will be in parallel, therefore for each band pass filter is calculated the output is added into a variable so that all fifteen filters are calculated and added. After the calculation finished, the value of the bits which is now 31 bit long, shifted to the right 15 bits to result into 16-bit long value. This calculation is performed for the left and right channel of the equalizer producing a digital value of 16-bit. The value of this calculation is fed into the DAC and then in turn fed into the amplifier (Chassaing and Reay, 2008; Chassaing, 2002; Kehtarnavaz, 2005).

Equalizer built in this system will use all the 15 band pass filters all in parallel and therefore each of the filters can have its own individual gain from -6 dB to +6dB. The attenuation of each band pass filters is very similar to a professional equalizer. The diagram block of the system with the slider control and display is shown in Figure 3a. The system uses the standard input and output of audio system and can use the typical RCA jack connections for the input and also output. In this case the connection uses 3.5 mm audio jack with left and right input/ output. The display control is serial connection (full handshake RS-232) between the OMAP evaluation board with the computer (Spectrum Digital, 2009; Texas Instrument).

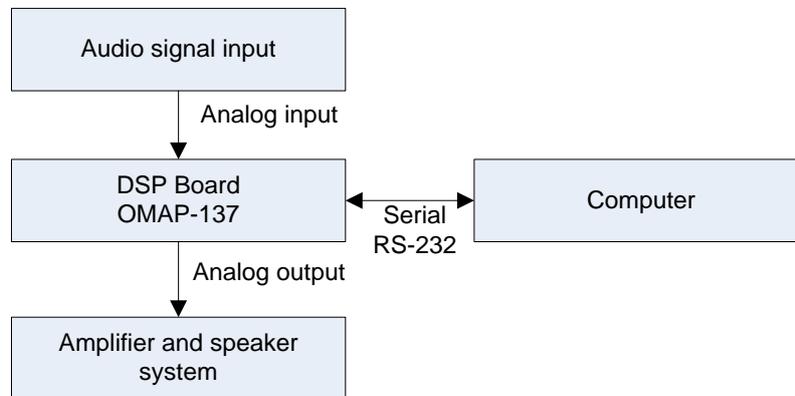


Fig. 3(a): Block diagram of digital equalizer with digital control.

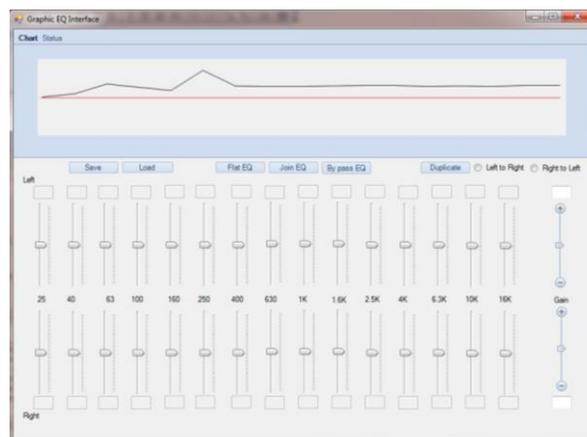


Fig. 3(b): Digital control interface of the digital equalizer.

Figure 3b is the digital control interface of the digital equalizer. This figure shows all the 15 channels on the top row for the LEFT channel and the bottom row for the RIGHT channel. Each of the slider channel capable of attenuate or amplify the signal from -6dB to +6dB and on the top part of the display is the approximate amplitude response of the equalizer for each channel, the LEFT and RIGHT channel. Oh the far right side of the slider is overall gain (\pm 6dB) of the equalizer in case a slight gain adjustment is needed.

Unlike the analog counterpart, this equalizer has additional six radio buttons on the top row which is explained in Table 3.

Table 3: Button functions found on the digital equalizer interface.

Button	Function
Save ^{*)}	To save the adjustment of the equalizer into the digital equalizer system memory
Load ^{*)}	To load the adjustment of the equalizer from the digital equalizer system memory that was saved earlier
Flat EQ	To obtain flat frequency response of the equalizer
Bypass EQ	To bypass the digital equalizer
Join EQ ^{*)}	To join the channel for adjustment so changing one frequency of the left channel will also change the frequency response of the right channel and vice versa
Duplicate ^{*)}	To duplicate the channel depending on the radio button selected, can be duplicated for the left or right channel
	Not found on analog equalizer

Testing and evaluation of the system:

Testing of the system and evaluation of the performance of the system is conducted using pure sine-wave generator and evaluate the results using an oscilloscope and voltmeter. Figure 4 shows the block diagram of the evaluation system.

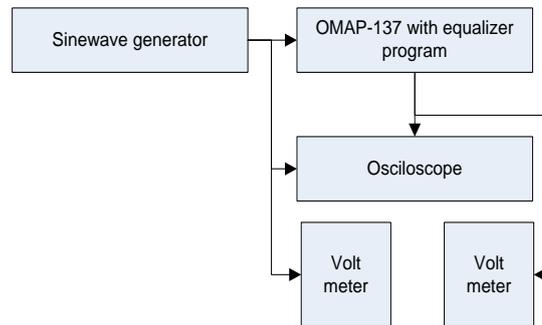


Fig. 4(a): Equipment to evaluate the digital equalizer against controlled input.

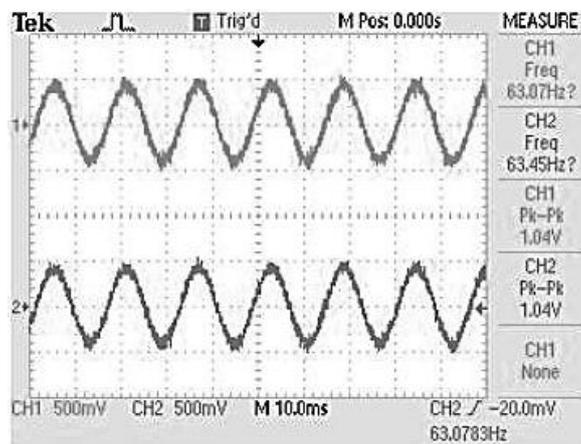


Fig. 4(b): Result of filter at 63 Hz with input from generator (top) and output from equalizer (bottom).

After testing the results against all frequencies with the two channels (left and right channel), the result of the filter shows similar results as in the 63 Hz frequency, which the output is the same as the input signals. When the frequency deviates from the center frequency, the signal experiences attenuation correspond to the same value as the calculated attenuation.

With attenuation using the controller to attenuate the value of the signals, the output shows no difference between the calculated value and the actual value. Observation is conducted using the same method and equipment as shown in Figure 4a (Malepati, 2010).

After the evaluation for every audio channel as mentioned in Table 2, the equalizer is subjected to the real environment which is using audio input and observes the value using human auditory perception. Based on the perception of several people, the attenuated value corresponds similar to the attenuated or enhanced value of the frequency.

Conclusion:

The digital equalizer system has been developed and built using OMAP-137 with frequency sampling of 48 kHz. With controlled input of sine wave, the equalizer provides output similar to the predicted value using calculations. This valued is shown using equipment to test the output results using oscilloscope and voltmeter.

The digital equalizer is also subjected to regular audio input and the output of the equalizer is observed using auditory perception. The result of the signal after any modification agrees with the perception of human hearing. These two results (controlled input and audio input) show that the digital equalizer works satisfactorily.

ACKKOWLDEGEMENTS

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