Universiti Malaysia PAHANG
Engineering • Technology • Creativity

Faculty of Electrical & Electronics Engineering
BEE4413: Digital Signal Processing

LABORATORY 1

Name : _____

Student I.D : _____ **Group #:** _____

Section : _____ **Date :** _____

Title : **Sampling and Filter Design Using EMONA myDSP and NI myDAQ**

Learning Outcomes

At the end of the experiment, student should be able:
- To familiarize with input and output of EMONA myDSP and Soft Front Panel (SFP) of EMONA myDSP.
- To know the function of NI myDAQ Input/Output pins and Instrument Launcher panel.
- To implement signal sampling using NI myDAQ and EMONA myDSP.
- To design real-time FIR and IIR Filter according to the filter specifications using NI myDAQ and EMONA myDSP.

Equipment Required
- PC with LabVIEW 2012 (or higher).
- NI myDAQ - 1 unit
- EMONA myDSP board – 1 unit
- USB cable – 1 unit
- MIC Cable and Connector – 1 each
- NI Minus Screw Driver – 1 unit
- MP3 player – 1 unit
- Probe – 1 pair
- Stereo Earphone Cable – 1 pair
- Jumper wire - at least 2

The component checklist is stated in the equipment box. Please ensure to return the equipment box with all components mentioned above inside the box. Please report to lecturer or lab technician for any missing components.

Procedure

Part A – MyDAQ & myDSP Setup for Experiment 1 (Sampling)

1.   Turn off the NI myDAQ unit by unplugging its USB cable from computer.
        WARNING: DO NOT HOTSWAP THE myDSP BOARD. myDAQ
        MUST ALWAYS BE OFF DURING INSERTION & REMOVAL.

2.   Plug the myDSP board into the NI myDAQ unit. Skip this process if the myDSP board is already connected to NI myDAQ.

3.   Power-up the NI myDAQ by connecting to the PC using the USB cable and wait for the Instrument Launcher to be automatically displayed as shown in Figure 1. If the Instrument Launcher is not automatically pop-up, browse the launcher in the NI directory in computer.

Figure 1: NI myDAQ ELVISmx Instrument Launcher

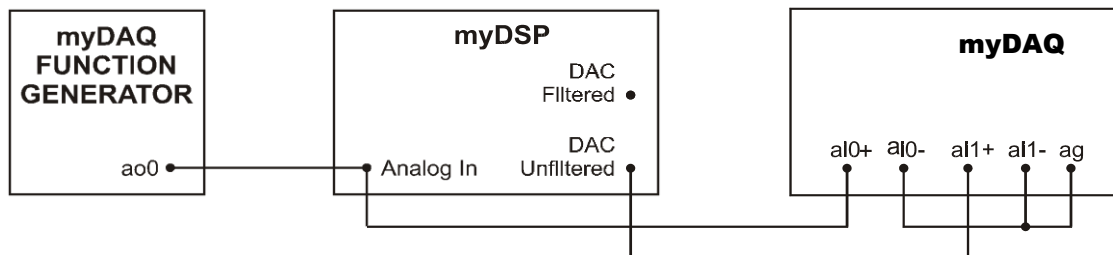4.   Patch the connections for myDSP as illustrated in Figure 2.

Figure 2: The Basic Connections of myDSP, myDAQ & myDAQ Function Generator

Table 1: myDSP board and myDAQ Connection settings

No.	Configuration Parameters	Settings
1	Analog IN	ao0, ai0
3	DAC Unfiltered	ai1+
4	Ground	ai0+ and ai1- set to agnd
5	Input Filter Switch	NO position

Table 2: myDSP board Configurations

No.	Configuration Parameters	Functions
1	Analog IN	Input signal to ADC
2	DAC Filtered	DAC output with filter reconstruction
3	DAC Unfiltered	DAC stepped output
4	Sample Clock	Sample output from DSP as clock signal
5	Input Filter Select Switch	Insert Anti-Aliasing Filter to input signal

5.   Open the Function Generator (FGEN) and Scope Instrument panel (Scope). Set Scope settings to 'Channel 0 is enable for AI0' and 'Channel 1 is enable for AI1', then run the Instrument. 2 flat traces can be seen on the scope.

6.   Now, configure the Function Generator settings at their default settings: Sine, 100Hz,1 Vpp, and then run the instrument. 2 Sinusoidal signals can be seen on the scope. Next, vary or sweep the Function Generator frequency from 100 Hz to 10 kHz. Observe on signal formation when the frequency of input signal is varied.

7.   Launch the myDSP Soft Front Panel (SFP) as shown in Figure 3.

Figure 3: EMONA myDSP Soft Front Panel (SFP); Filter Design to Gain Mode

8.   Here, the EMONA DSP SFP can be configured into 2 configurations; Filter Design to Gain Mode and Manual Gain Entry Mode. The myDSP block diagram is shown in Figure 4.

Figure 4: EMONA myDSP Block Diagram

9.   MyDSP Parameter Configurations;

9.1 Filter Design to Gains mode
Filter type: Lowpass, Highpass, Bandpass, Bandstop. The default filter type is Lowpass.
Design method: Butterworth, Elliptic, Chebyshev and Inverse
Chebyshev.
Sampling Frequency: enter the clock frequency of the unit delay
elements.
Filter specifications: pass and stop frequencies (Hz) for the Filter
type.
Ripple Specifications: passband ripple (dB) and stopband
attenuation (dB).

9.2 Transfer Function display & format
The tabs allow the display of a number of representations of the Transfer Function for
the filter design. Expanded Form of Transfer Function is more preferable as this
relates most closely to the polynomial and coefficients. The default format is
expandable form.

9.3 Filter Order
The default value of filter order or length is 4. The maximum filter order is 10.

9.4 Graphic displays
The filter design is displayed by frequency (magnitude) response, phase response and z-
plane plot of poles & zeroes. These displays are instantly updated to match the filter
design. The exact location of each pole and each zero is available at the bottom center
indicators 'zeroes' (denoted by 'o') and 'poles' (denoted by 'x').

9.5 Extract Hardware Gains
This control, when pressed, causes the filter design values, such as sampling
frequency, filter order and polynomial coefficients to be extracted from the design and
copied to the individual **HARDWARE GAINS** controls. These controls have a
yellow background and the values of a0 … aN and b0 … bM. Here, N=M=Filter
Order.

9.6 Download to myDSP
This control, when pressed, will load the 'extracted' parameters into the DSP processor for
the implementation of the specified filter.

9.7 ELVIS Instrument Launcher Panel
The FGEN, Scope, Bode & DSA can be launched from this button panel as well.

9.8 Manual Gains Entry mode
'Manual Gain Entry' mode allows the user to vary individual hardware gains, and see
the corresponding filter design be displayed in the Graphic displays as described by
Figure 5. Any yellow control may be varied in this mode. This is especially interesting
when the **gains are being varied manually**, the user can see the **poles and zeros
moving about the unit circle in real time** and changes the **filter response (Magnitude
& Phase)**. Next, user can load and run that filter and see how it performs with real
signals in the 'real' world.
These values will override the values setup in the 'Filter Design to Gains mode' tab.

9.9 Filter structure
There are 2 filter structures implemented in the DSP unit. These are Direct Form 2 (DF2) and Direct Form 2 transposed. These are selected on this tab.

9.10 Order (0-10)
User can select the order of the filter from 0 to 10 only.

9.11 Tap select (0-11)
As well as viewing the output of the structure, the user can choose to view any one of the internal taps of the structure from 0 to 11 only.

9.12 Input scaling
A user can internally scale a signal, after sampling, by varying this control. This serves to maintain a large scale signal for sampling with mathematical amplitude reduction internally, to improve signal/noise performance.

9.13 Sampling Freq
The actual sampling frequency to be sent to the myDSP can be set here.
.
9.14 Tune Ts
It is used to tune the sampling period manually. It will vary the actual Ts value of sampling rate.
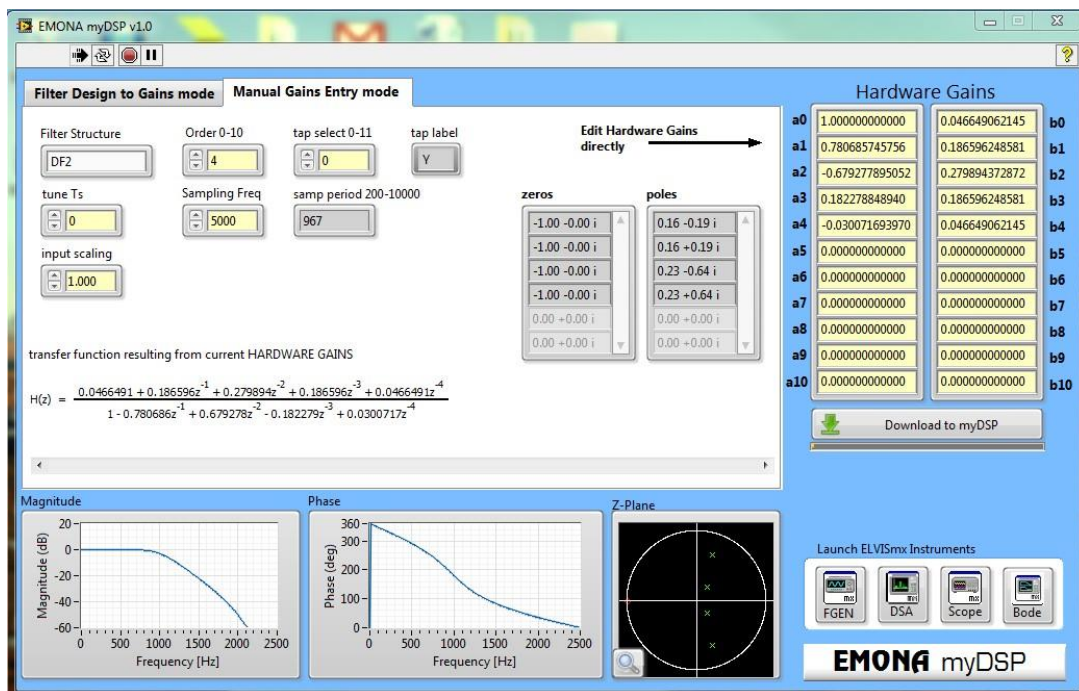
Figure 5: EMONA myDSP Soft Front Panel (SFP); manual mode

10. EXPERIMENTS

Experiment 1: Sampling, Aliasing & Nyquist

Part 1: through the time domain
In this exercise, the sampling of a sine wave can be observed. Configure the myDSP and myDAQ board as shown in Table 1 and Figure 2. Select design mode = Manual Gain Entry mode. Setup the myDSP to implement simple straight through sampling. The settings are shown in the screenshot as shown in Figure 6 below. Click on 'LOAD myDSP' when it is ready. With the structure set to order equal zero, then the input signal will be sampled by the ADC and directly output by the DAC. The settings of oscilloscope and the results are shown in Figure 6.
Setup the myDSP SFP to implement a simple sampling, with no delay elements.
MODE: manual gains entry
Filter Structure=DF2.
Sampling Frequency =10 kHz; order 0; tap 0; input
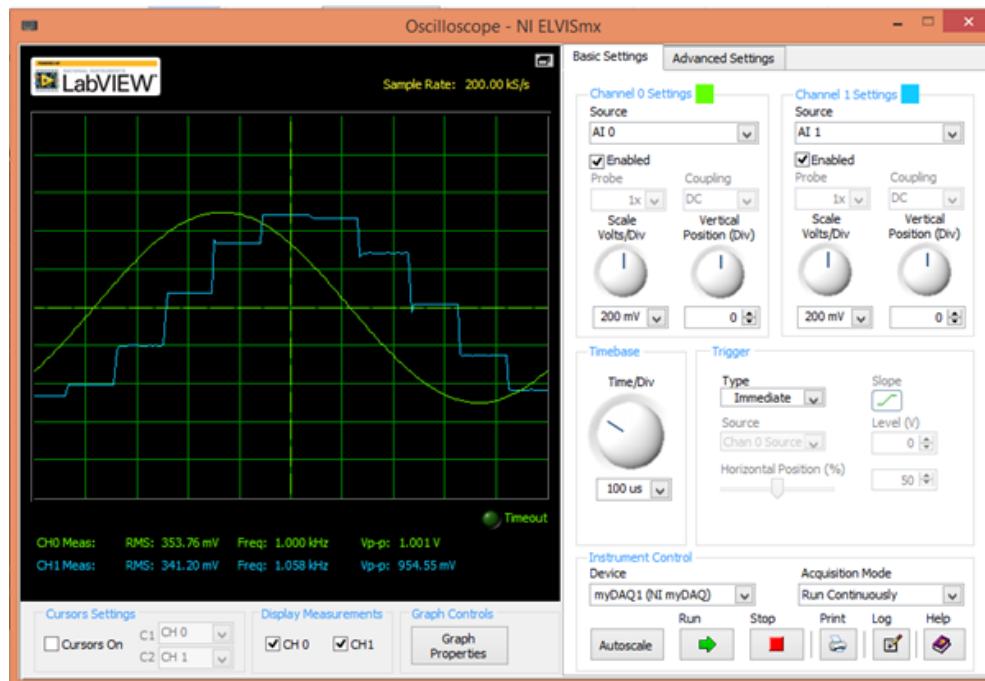Scaling=1; GAINS: unused.
Input Frequency= 1 kHz

Figure 6: Signal Sampling results at 1 kHz input signal and 10 kHz sampling

Task 1:
(a) Vary the input frequency at 200 Hz, 5 kHz, and 9 kHz with the same configurations of myDSP board and myDAQ and sampling frequency. Observe the difference and identify the aliasing.
(b)   Change the configuration of myDSP from DC unfiltered to DC filtered and vary again the input frequency in part (a). Observe the difference.
 Report your findings for part (a) and (b).

Part 2: through the frequency domain

Set the Function Generator frequency to 1 kHz. Use the same myDSP configuration SFP settings as stated in Part 1.

Stop the SCOPE and run the DSA (Dynamic Signal Analyzer or SpectrumAnalyzer), to view the signals in both the time and frequency domains as shown in Figure 7. Configure the settings of Spectrum Analyzer as shown in Figure 7.

Focus on the main harmonics signal at 10 and 20 kHz. Now move your observation point to the DAC UNFILTERED output, which is the sampled of 1 kHz sine wave. It can been seen that the spectrum change from single harmonics at 10, 20 and 30 kHz to a pair of harmonics at 1 kHz on either side of these points. The process can be repeated a few times to confirm the existence of harmonics.

Task 2:

Vary the input signal frequency at 5 kHz and 10 kHz. Observe the difference and report the findings.
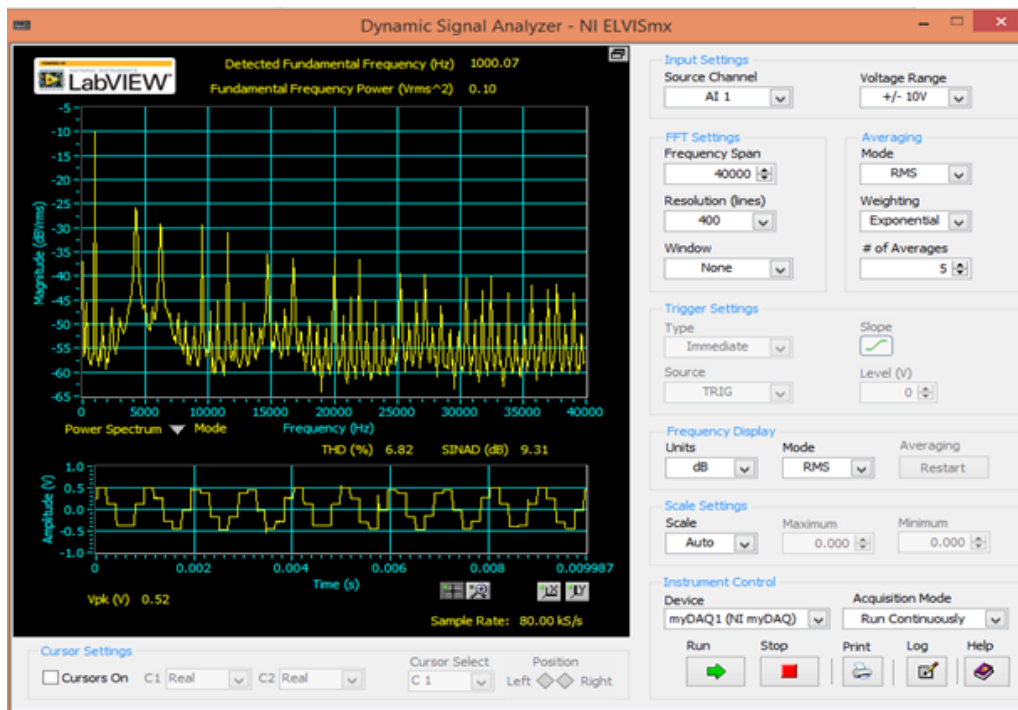

Figure 7: Signal Sampling Viewed in Frequency Domain

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Part B: Filter Implementation using NI MyDAQ & myDSP

Experiment 2 – Design the 10th Order IIR Bandstop Filter to eliminate power line noise

Objective: To design suitable IIR filter to eliminate 60 Hz power line noise with 2nd harmonic of the noise at 120 Hz.

Method:
Set the myDSP configuration as below as shown in Figure 8;
Mode: Filter Design to Gain Mode
Filter type: Bandstop
Design Method: Chebyshev
Sampling Frequency: 500 Hz
Filter specifications: fpass1=100, fstop1=110, fpass2=140, fstop2=130
Ripple specifications: passband=3 dB and stopband=40 dB
Filter Order=10th
Extract Hardware Gains.
Download to myDSP.
Obtain the Transfer function of the filter, zero-pole location, filter response and system stability. Repeat the experiment by reducing the filter order and sampling frequency. Report the observation.
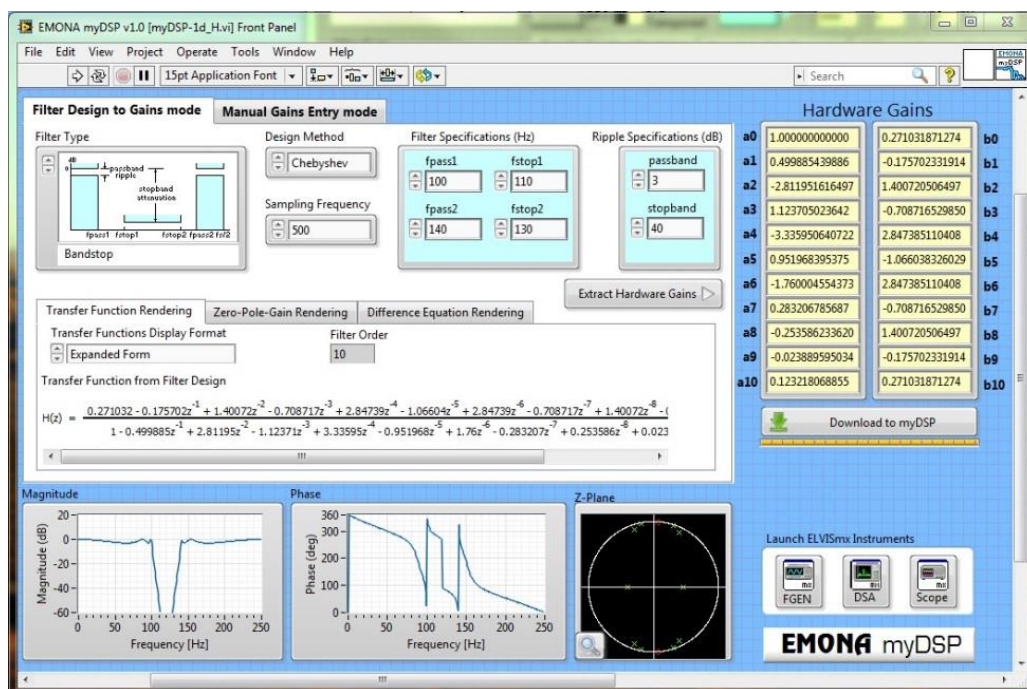

Figure 8: 10th IIR Bandstop Filter

Task:
The transfer function of the IIR Bandpass filter with sampling frequency of 1 kHz, stopband attenuation of 20 dB, passband ripple of 3 dB is given below;
$H(z) = 0.0675(1 - 2z^{-2} + z^{-4}) / (1 + 1.143z^{-2} + 0.4128z^{-4})$
Use Manual Gains Entry Mode to obtain filter response, zero-pole location and system stability.

Part C: Filtering Music using NI MyDAQ & myDSP

Objective: To remove the background noise from the music by designing a real-time filter using NI myDAQ and myDSP.

Method: 1. Set-up the experiment according to the Figure 1 and Figure 2 as shown below.
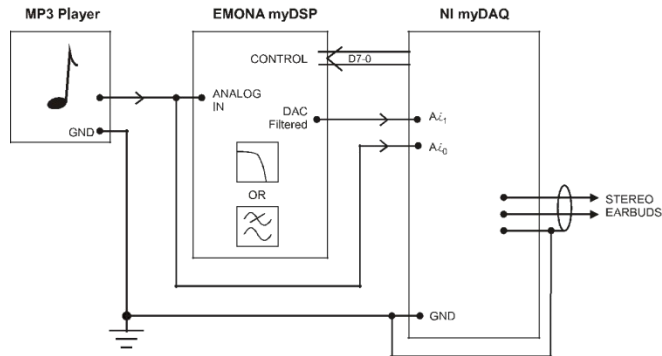

Figure 9: Block diagram for the connection of MP3 player, EMONA myDSP and NI myDAQ
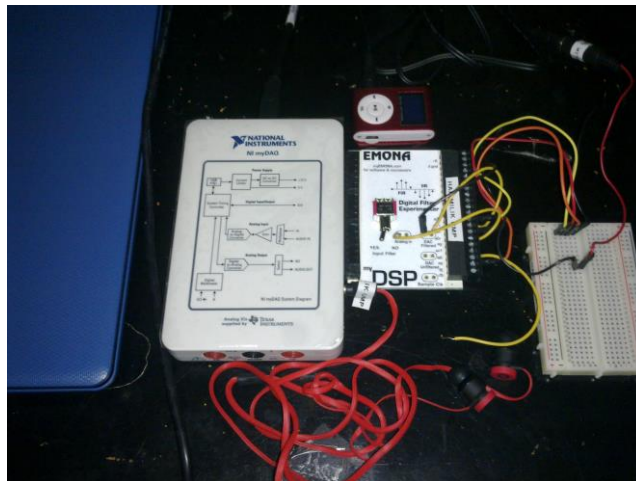

Figure 10: The complete devices connection

2. Open the Audio in-out Spectrum Analyzer as shown in Figure 11.
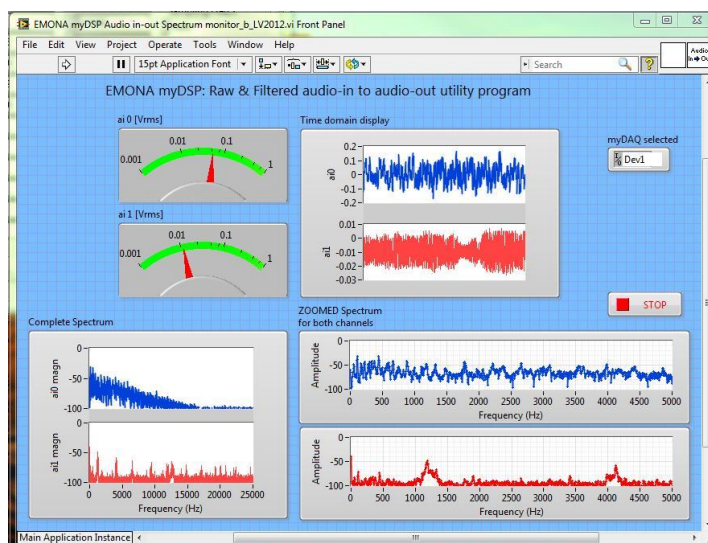

Figure 11: Audio in-out Spectrum Analyzer

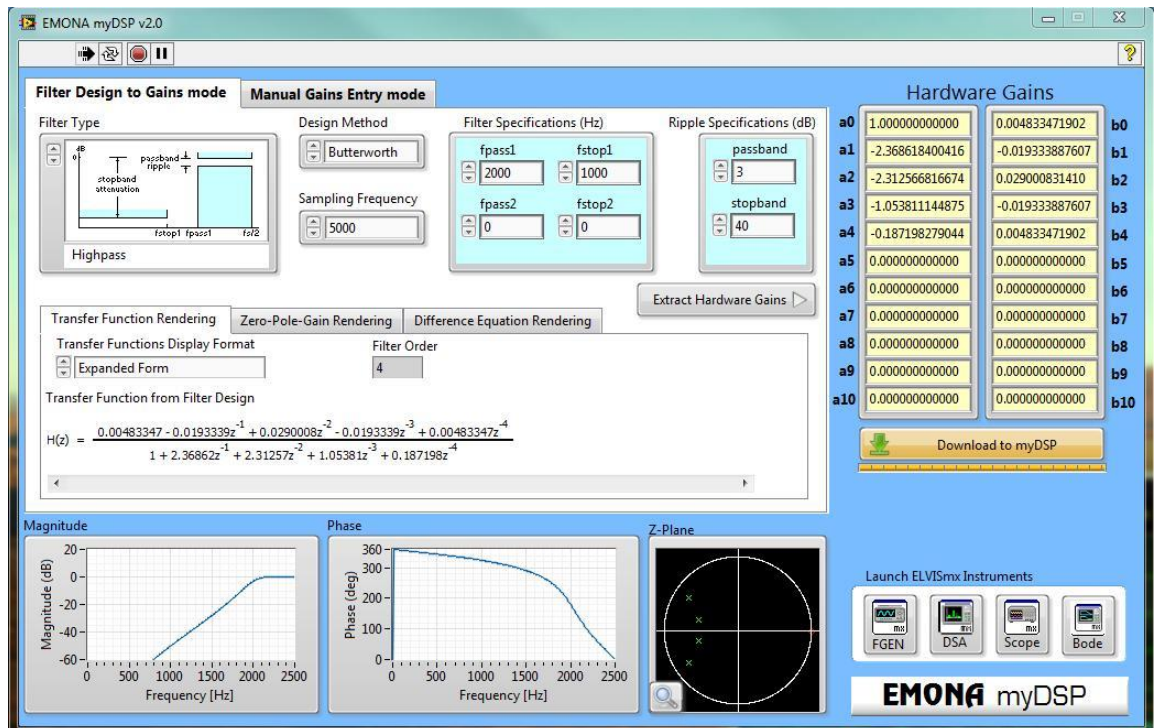3. Configure the EMONA myDSP as shown in Figure 12.

Figure 12: EMONA myDSP Configurations

4. Play some music from your MP3 player and listen to the stereo earbuds one at a time.
5. Note that one channel will be the original music, whilst the other channel will be the filtered music. You may find it easier to listen to one earbud at a time.
6. Try changing the filter design on the "myDSP SFP" to remain as a Lowpass, but with fpass1 = 200, and fstop1 = 800. This will also be a 4th order filter running at 5kHz. Press "EXTRACT" to compute the filter co-efficients, and then press" LOAD myDSP" to transfer the implementation coefficients to the hardware. You will now hear that most high pitched sounds have reduced and mainly low frequency sounds are output…as expected according to your design specifications.
7. The "spectrum monitor" program displays the spectrum of the processed signal. The **unfiltered input signal is in blue color** and **filtered output signal is in red color**.
8. Finally change the filter type to Highpass. Enter a fpass1=2000, and fstop1=1000. Leave sampling frequency as is at 5kHz. Switch to the 'manual gains entry mode' tab and review the selections there. Extract the hardware gains, and download to the myDSP. Listen to the high frequencies remaining in your music, and notice how much energy is present in the low frequencies and how noticeable it is when this is removed.
Report the findings.