TECNOLÓGICO DE MONTERREY Escuela de Ingeniería y Ciencias Departamento Mecatrónica	**MR3012. Mechatronics Laboratory** Prof. Israel Ulises Cayetano Jiménez **Lab session 2 – Introduction to DSP**

Laboratory Session 2

Introduction to Digital Signal Processing

Objectives

- Implement basic DSP using microcontrollers, MATLAB and LabVIEW
- Understand the difference between FIR and IIR filters.
- Understand the differences between time domain and frequency domain approaches.
- Introduce the concepts of Binaural Audio for VR.

Materials

- Arduino compatible board (preferably ESP32) with ADC and processing capabilities.
- Computer with MATLAB, LabVIEW and microphone.
- Waveform generator
- Oscilloscope with 1 probe
- Headphones/earphones

Procedure

Part 1 – Basic Signal Processing Implementation

1. Given $f_s = 100\ \mathrm{Hz}$, design the following digital filters:
   a) Second order Butterworth LPF with $f_c = 5\ \mathrm{Hz}$
   b) IIR HPF with $f_c = 1\ \mathrm{Hz}$ designed directly on the z-plane.
   c) Fourth order Moving Average (get the f_c for such a filter).
2. Obtain the Bode plot, impulse response, z-plane, and difference equations of each one of the filters.
3. Create a signal vector that represents a 2 Hz square signal that goes from 0 to 1.65 V and apply the filters directly on MATLAB with the filter function.
4. Obtain the FFT and PSD of both the input and output signals.
5. Generate the same signal with the waveform generator and input it to your board.
6. Using the oscilloscope, obtain the FFT of the input signal

7. Using difference equations, program each one of the filters directly on your board and show both the “live” input and filtered output on a Virtual Instrument.
8. Use the signal processing tools from LabVIEW (Point by Point) to apply the filters and once again show both the “live” input and filtered output.
9. Compare the three implementations (MATLAB, board, and LabVIEW).
10. Implement, physically, the output processed signal from filters ***a*** and ***c*** (DAC).

Part 2 – Binaural Audio using MATLAB

1. Look for a database with the HRTF and download it.
2. Define a virtual scene by deciding where to place 3 different audio sources (at least one of them should be a recording done by you) with respect to the spectator and when to play them.
3. Download or record the required audios.
4. Convert your audios into mono sound.
5. Using the filter command, apply the HRTF to each audio according to the location previously defined
6. Generate a single stereo audio from the different filtered audio sources. Listen to it with your earphones/headphones.

Deliverables

IEEE format report covering part 1 (no more than 5 pages using LaTeX).

IEEE format report covering part 2 (no more than 4 pages using LaTeX).

Appendix document including the implemented code and personal comments about the lab session.

Binaural audio file.

Additional resources:

Smith, A. (2017). *Binaural audio: What is it? How can you get it?*
Retrieved from: https://www.whathifi.com/advice/binaural-audio-what-it-how-can-you-get-it