# System Requirements Document

## Audio digital signal processor

### BeCreative Minor

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April 11, 2023

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# Abbreviation List

| Abbreviation | Explanation |
|---|---|
| DAC | Digital to analog converter |
| ADC | Analog to digital converter |
| RAM | Random acces memory |
| SINAD | Signal to noise and distortion |
| TRS | Tip ring sleeve connector(Jack) |
| FPGA | Field programmable gate array |

Table 1: List of commonly used Abbreviations

# Chapter 1:   Background

When listening to music it is of great importance that the speakers are tuned to the environment and the position of the listener. This is necessary to achieve the best experience. If the speakers are not correctly tuned to the surrounding environment, a digital signal processor (DSP) is used to correct this. A DSP is a specialized processor which is used for digital signal processing.

In the audio world a DSP is used to optimize a sound system. For example some speakers have some imperfections and a DSP can be used to correct for these imperfections. It is also often used to add more dynamics to sound.

The goal of this project is to research how to make an audio-DSP. This raises the main research question: "How to design an audio-DSP?". In the process of researching this an actual audio-DSP will be developed. From the main research question the following sub-research questions are derived:

- What is the best method for creating digital filters?
- What is the best method for creating digital effects?
- What is the most suitable anti-aliasing filter?
- What is the optimal needed roll-off for the anti-aliasing filter for a given bandwidth such that the noise can be negligible?
- What is the minimum sample frequency needed to capture the desired frequency spectrum?
- What is the minimum frequency range to be sampled to achieve sufficient detailed audio?
- What is the lowest allowable noise for decent audio?
- What ADC resolution is needed such that the quantization error and noise level are on par?
- What ADC and DAC architecture is most suitable for this application?
- What kind of processor is most suitable for this application?
- What is the permittable jitter for accurate audio?
- What is the maximum allowable ripple on the reference voltage for the ADC and DAC?
- How much RAM does the system need?
- What power supply topology is best suited for each part of the system?

The project is conducted during the minor BeCreative at Fontys. This minor takes 20 weeks and allows the students to have a budget of €300,-. Thus after 20 weeks starting from 6-2-2023 an audio-DSP will be delivered within a budget of €300,-. The audio system has some requirements to specify the final result. These requirements are derived with the "MoSCoW" method. It must be noted that the following requirements will be confirmed by the research that will be conducted.

# Chapter 2:   Requirements

| ID | Requirement | Priority | Status |
|---|---|---|---|
| **U1** | **Inputs:**<br>• Two RCA audio inputs which work on a line level of 4dBu(±1,74V)<br>• Two 6,35mm TRS plug audio inputs which work on a line level of 4dBu(±1,74)<br>• Two XLR audio which work on a line level of 22dBu(±9,75) | Must | Proposed |
| **U2** | **Outputs:**<br>• Two RCA audio outputs which work on a line level of 4dBu(±1,74V)<br>• Two 6,35mm TRS plug audio outputs which work on a line level of 4dBu(±1,74V)<br>• Two XLR signal outputs which work on a line level of 22dBu(±9,75) | Must | Proposed |
| **U3** | The system should have a bandwidth (±3 dB) of at least 20 Hz up and till 20 kHz without any filters applied. | Must | Proposed |
| **U4** | The system has an Audio sample rate of at least 192 kHz | Must | Proposed |
| **U5** | The ADC and DAC resolution is at least 16-bit | Must | Proposed |
| **U6** | Signal-to-noise and distortion (SINAD) is at least 100dB | Must | Proposed |
| **U7** | Anti-aliasing filter is a 6th order filter | Must | Proposed |
| **U8** | propagation delay of less than 100ms without any filters applied | Must | Proposed |
| **U9** | The system has two samplers | Must | Proposed |
| **U10** | The system has two input channels | Must | Proposed |
| **U11** | The system has two output channels | Must | Proposed |
| **U12** | The system has two signal processors | Must | Proposed |
| **U13** | User can select what input will be routed to what output channel via a user interface | Must | Proposed |
| **U14** | All inputs use at least 80% of the ADC resolution at their specified line level | Must | Proposed |
| **U15** | User can select 1 effect to be active in one channel at the same time | Must | Proposed |
| **U16** | The 6,35mm TRS plug audio inputs must have a variable gain | Must | Proposed |
| **U17** | User can configure each effect | Must | Proposed |
| **U18** | The system works standalone | Must | Proposed |
| **U19** | The user can configure each effect in the user interface | Must | Proposed |

| U20 | The system has a visual representation of the user interface | Must | Proposed |
|---|---|---|---|
| U21 | Effects configurable in each signal processor channel:<br>• Distortion<br>• Reverb<br>• Gain<br>• Equalizer<br>• Delay | Must | Proposed |
| U22 | An FPGA is used as processor | Must | Proposed |
| U23 | RAM is at least 2MB | Must | Proposed |
| U24 | The system has a bandwidth ($\pm1$ dB) of at least 20 Hz up and till 20 kHz without any filters applied | Should | Proposed |
| U25 | The ADC and DAC resolution is at least 24-bit. | Should | Proposed |
| U26 | Signal-to-noise and distortion (SINAD) is at least 120dB | Should | Proposed |
| U27 | propagation time delay of less than 10ms without any filters applied | Should | Proposed |
| U28 | The system has four input channels | Should | Proposed |
| U29 | The system has six output channels | Should | Proposed |
| U30 | The system has six signal processors | Should | Proposed |
| U31 | The system has a USB audio input | Should | Proposed |
| U32 | The system has a USB decoder | Should | Proposed |
| U33 | Six XLR signal outputs work on a line level of 22 dBu ($\pm9{,}75$ V) | Should | Proposed |
| U34 | The system is able to recover the last saved configuration of the effect and the channel routing after reboot | Should | Proposed |
| U35 | The system has equalizer presets e.g. Rock, Classical, Default, effect | Should | Proposed |
| U36 | The system has different effect presets | Should | Proposed |
| U37 | The system has default settings for channel routing and presets | Should | Proposed |
| U38 | User can select up to 4 effects to be active in one channel at the same time. | Should | Proposed |
| U39 | Effects configurable in each signal processor channel:<br>• Phaser<br>• Tremelo<br>• Flanger<br>• Fuzz<br>• Overdrive<br>• Chorus<br>• Compressor<br>• Wah<br>• Looper<br>• Wow and flutter<br>• Modulator<br>• Echo<br>• Fade in/out<br>• Delay (at least 4 seconds) | Should | Proposed |
| U40 | Local power supplies for different parts of the system | Should | Proposed |
| U41 | A custom PCB is designed for the local power supplies, front-end and back-end of the system | Should | Proposed |
| U42 | Signal-to-noise and distortion (SINAD) is at least 140dB | Could | Proposed |

| U43 | The user can configure custom presets for the equalizer, effect and channel routing via the user interface | Could | Proposed |
|------|------|------|------|
| U44 | propagation time delay of less than 100µs without any filters applied. | Could | Proposed |
| U45 | User can select up to 10 effects to be active in one channel at the same time | Could | Proposed |
| U46 | Touch screen user interface | Could | Proposed |
| U47 | Zero-crossing detection to make adjusting the filters happen glitch free | Could | Proposed |
| U47 | Option to reverse polarity of the output signals | Could | Proposed |
| U48 | Self-made mains power supply | Won't | Proposed |