System Requirements Document

Audio digital signal processor

BeCreative Minor

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April 17, 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:** • Two RCA audio inputs which work on a line level of 4dBu(±1,74V) • Two 6,35mm TRS plug audio inputs which work on a line level of 4dBu(±1,74) • Two XLR audio which work on a line level of 22dBu(±9,75)	Must	Proposed
U2	**Outputs:** • Two RCA audio outputs which work on a line level of 4dBu(±1,74V) • Two 6,35mm TRS plug audio outputs which work on a line level of 4dBu(±1,74V) • 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: • Distortion • Reverb • Gain • Equalizer • 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 (± 1 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 ($\pm$9,75 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: • Phaser • Tremelo • Flanger • Fuzz • Overdrive • Chorus • Compressor • Wah • Looper • Wow and flutter • Modulator • Echo • Fade in/out • 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