GEB1 - A ROBUST DSP PLATFORM FOR AUDIO AND GUITAR SIGNAL PROCESSING IN EDUCATION

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ABSTRACT

The development of a small, robust and cheap embedded hardware is presented, based on the `TMS320C6722` signal processor. The system is tailored to the particular needs of audio and especially guitar effect processing, and is dedicated for educational use. The hardware realization, with focus on component selection, and the software framework are explained. To demonstrate the utility in education, we give examples of successfully implemented student works on our DSP platform.

1. INTRODUCTION

Lectures on digital signal processing are very mathematical and often difficult to access. The probably best way to impart understanding to students is to face them with real DSP hardware to gain practical experience. Audio effects, in this context, are an excellent educational material which helps to acquire the techniques and tools of digital signal processing and system theory in general [1]. The first algorithms are implemented quickly and the success is easily checked by listening. Especially the real-time processing of electric guitars is a popular exercise among students and a suitable topic for bachelor or master theses.

Several available DSP-boards are specified for audio applications. In case of the Texas Instruments C6x family, first of all the starter kits (e.g. `TMS320C6713 DSK`) are qualified when starting with DSP programming. A drawback is the low mechanical robustness, simply because these hardware cards are designed for table use. Other DSP boards like the PADK[1] offer many input and output channels at high quality and lots of possibilities, but at an accordingly higher purchase price.

In all cases additional peripherial equipment is necessary. If an electric guitar is considered as signal source, a preamplifier and impedance converter is strongly recommendable.

With audio effect processing in mind, it is often desirable to change parameters continuously, for instance a stereo panner or a user controllable gain. To be equipped for such tasks, accessory turning knobs have to be interfaced as well.

Not any of the available DSP boards meets all these (special) requirements. Due to this, the idea arose to create an own, ready-to-use DSP platform which is tailored to the demands of audio, and especially guitar effect processing.

The target specifictions are:

- compact, small, but robust at low cost,
- limited to two channels (stereo) and restricted to the essentials (to keep it cheap and manageable),
- ready-to-use as floor controllable box (like normal analog guitar stompbox) and
- a software framework that allows easy integration of own algorithms.

[1]Professional Audio Development Kit, Lyrtech Incorporated

Figure 1: Picture of the GEB1. The four knobs in the rear are the potentiometers and the encoder, ahead the two foot switches.

The developed embedded system (figure 1) got the name *Guitar Effect Board 1* - GEB1. In this paper we describe the implemetation in hard- and software and give examples for successful use in academic environments.

2. AUDIO AND GUITAR EFFECTS

To understand the purposes this board is designed for, a short introduction to audio effect processing has to be given.

The collective term “audio effect” is used for all accessory devices that have the intent to audibly modify a music signal. In principle, effects can be used either for a single instrument or for the whole mix. The term “guitar effect” specifies all audio effects that are designed for use in particular with electric guitars.

Most effects have their origin in analog, vintage equipment of the 1950s to 1970s. In that time a variety of effect units, often in form of floor-operated “pedals”, were invented. In this context, typical guitar amplifiers have to be treated as effects too, because they do not simply reproduce the instruments signal but change its sound explicitly. Many guitar effects go beyond the bounds of LTI systems: for instance the utilization of nonlinear distortions to enhance the harmonic spectrum or equalization filters with a time-varying

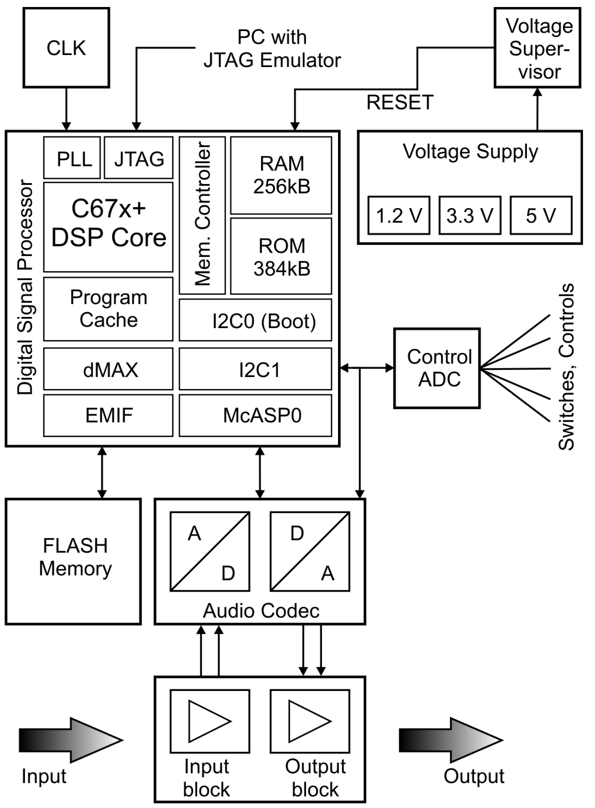

Figure 2: Functional overview of the embedded system.

frequency response to give a spatial impression.

The task of digital audio effects is in many cases the simulation of these effect units and amplifiers, to imitate the sound of the reference systems as true to original as possible. A good overview over audio effects is presented in [2]. A review article with focus on modeling techniques for nonlinear amplifiers is given in [3].

3. HARDWARE DESCRIPTION

A functional overview of the GEB1 is depicted in figure 2, where the main blocks of the system are pulled together. In the following, these blocks and their interrelations are explained.

3.1 Input Stage

Typical guitar pickups are signal sources with a high impedance. For that reason the input impedance of the board is chosen to $Z_i \approx 1\,\mathrm{M\Omega}$, a value which is high enough to ensure a good matching. Both stereo inputs are equipped the same, with a non-inverting operational amplifier and a simple attenuator. The preamplification is then adjustable in the range $G = -6\,\mathrm{dB}$ to $22\,\mathrm{dB}$ via trimpots which can be tuned through small holes in the front panel. The frequency response is flat except for a low cut caused by the coupling capacitors. A half `OPA2134` amplifier is used (the second half is used for the outputs, see section §3.4).

3.2 AD/DA Conversion

The analog/digital (ADC) and digital/analog conversion (DAC) are performed by the 24-bit delta sigma codec `PCM3060`. The control over the device (clock settings and other configurations) is arranged from the DSP via the I^2C bus[2]. For the digital audio data the I^2S format[3] is used. The sampling frequency F_s is adjustable in the range 16 kHz to 96 kHz, for most applications $F_s = 48\,\mathrm{kHz}$ was found to be sufficient.

3.3 Signal Processing

The heart of the GEB1 is the embedded digital signal processor (DSP) of type `TMS320C6722`. The decision for this IC was made, among other things, in consideration of prototyping expenses and the limited prospects in educational environments: this device, with its 144-pin quad flatpack (RFP) package, can still be soldered manually in contrast to other chips which are only available as ball grid array (BGA) packages.

The following interfaces and controllers are used: the digital audio data is handled by the McASP[4] interface and the dMAX[5]. During boot operation the program code is read from an external memory device via the EMIF[6]. Finally, the I^2C bus links the audio codec and the control ADC.

3.4 Output Stage

In most cases the output is connected to a mixing console or amplifier input. The output level is adjusted to 6 dB V (max) to have enough headroom. To suppress the high frequency noise caused by the delta sigma converters, the output amplifier features a low pass characteristic. Left and right channel are identical.

3.5 Booting and Memory

The integrated standard 4-megabit flash memory allows the user to store executable program code for stand-alone usage. This memory is accessed exclusively (using the EMIF interface) during boot operation. While the default boot mode is set to 'flash', other start sources are still selectable by changing the jumper configuration on board.

In this first version no additional RAM is integrated. This is a possible extension for an upcoming redesign.

3.6 User Controls

The six control elements (potentiometers, footswitches, encoder) are sampled by the 8 channel AD-converter `ADS7830` and queried via the I^2C bus. The 8-bit resolution is sufficient to give the potentiometers the required "analog feel".

For visual feedback two light emitting diodes are installed. These elements, which are switched by two GPIO pins of the McASP, are freely programmable. Applications could be the acknowledgement for a completed operation, indication of operation mode and many more.

[2] Inter-integrated circuit
[3] Inter-IC sound interface
[4] Multi-channel audio serial port
[5] Dual data movement accelerator
[6] External memory interface

3.7 Voltage Supply and Supervision

Four different DC voltage levels have to be provided on the hardware.

- 9 V. The board is designed for use with an external DC feeding. Any simple DC power supply in the range 7 V to 12 V with at least 150 mA is working. Nevertheless, a stabilized source is recommended to avoid power line hum. After HF filtering and a few bulk capacitors, this input voltage powers directly the operational amplifiers. All other voltages are derived from this supply.
- 5 V. This voltage is solely needed for the analog part of the codec. A simple voltage regulator is used.
- 3.3 V. This level powers most semiconductors, e.g. the digital part of the codec and the flash memory. It is also the I/O supply voltage of the DSP. With the `LP2985` again a voltage regulator is integrated.
- 1.2 V. This is the core voltage for the signal processor. This supply is generated using a switching regulator (switching frequency $f_{sw} = 285\,\text{kHz}$).

In addition, separated ground levels for analog and digital circuits are provided, with special care of the analog ground plane. Both potentials are connected via a $0\,\Omega$ resistor near the audio codec.

The core and I/O voltages of the processor are observed by the `TPS3106` supervisor circuit, which ensures that the processor is only taken out of reset once the voltages have reached steady state after power up.

3.8 Mechanical Construction

The board is explicitly designed for stage use and therefore has to be very robust. Hence, the sensitive electronics are housed by a small diecast aluminium box. To allow the user to change parameters or switch between different settings, the following control elements are provided:

- two foot-operated stomp switches,
- one solid encoder with six positions and
- three rugged potentiometers.

As connectors for the audio signals 1/4" phone jacks are used. Compared to the tiny surface mounted components are these parts relatively large, but standard in electric guitar equipment. For power supply the commonly used coaxial DC sockets are integrated. The printed circuit board (PCB) is easily accessed after removing the four skrews from the rear panel. It is mounted to the enclosure both to the side walls through the phone jacks and to the frontpanel through the potentiometers. A picture of the inner life is shown in figure 3.

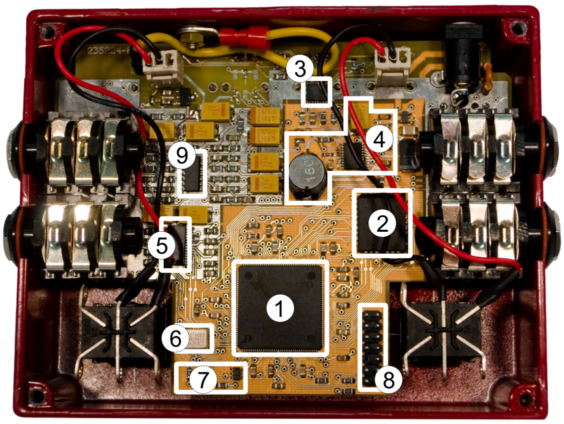

Figure 3: Picture of inner life. Highlighted components: (1) DSP, (2) flash, (3) control ADC, (4) switching regulator, (5) audio codec, (6) oscillator, (7) supervisor, (8) JTAG port and (9) OPA.

4. SOFTWARE FRAMEWORK

The programming of the GEB1 is done in C using the development environment *Code Composer Studio*. For debugging the JTAG[7] port and an emulator (e.g. Spectrum Digital SD 510USB) have to be used.

The framework divides into two parts, the `main` function and the interrupt controlled `dmax_isr` routine. During start up, `main` executes all necessary module initializations, including PLL controller, McASP interface and dMAX. After finishing these operations, an infinite loop is entered, in which all non prioritary tasks can take place. Dedicated tasks are the querying of the control elements or the setting of indicator LEDs, see section §3.6.

The `dmax_isr` routine is responsible for the signal processing. All effect algorithms take place here. The routine is initiated by an interrupt depending on the input. In general, two strategies are common:

1. **Sample-by-sample processing**
   Every new sample causes a new interrupt. All computations for this sample have to be done before the new sample is disposed, i.e. within the sampling interval $T = \frac{1}{F_s}$.
2. **Block-by-block processing**
   Every N samples (e.g 128 or 256) a new interrupt is released. The computation of the whole block has to be finished before the next block of N samples stands by.

The block diagram in figure 4 illustrates this context exemplary for a multi-effect application. The interrupt routine reads out the input samples and calls different "presets", dependend on the position of the encoder. Each preset consists of one or more effect algorithms, each capsuled in an own routine, which are arranged in a cascade. When all algorithms have been processed, the output samples are released. An example for a realistic effect chain is

compressor $\rightarrow$ amplifier model $\rightarrow$ loudspeaker simulation.

The framework makes it easy for beginners to get into the programming. The students can directly start to work on sample level, without considering challenges like how to address the audio codec. The input samples come in, the user defines some processing and then assigns the values to the outputs.

5. EDUCATIONAL USE

The first version of the described hardware unit was issued in 2008 as main part of the first authors diploma thesis. At that time the software included the simulation of rotating speakers (*Leslie* effect), a Vox guitar amplifier model and some simple basic effects like stereo delay or tremolo.

In the meantime several students worked successfully on the GEB1 during their project- or bachelor thesis work. The following tasks have been implemented:

[7] Joint Test Action Group, IEEE 1149.1 Standard Test Access Port

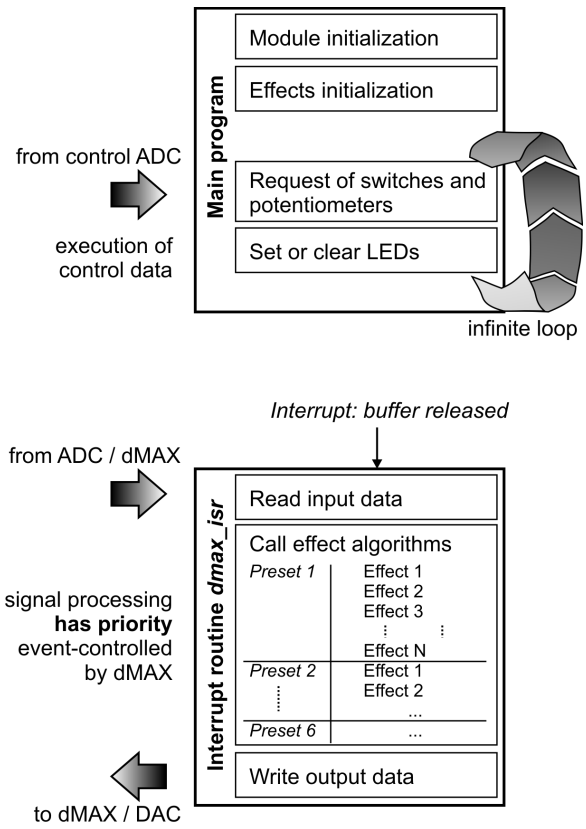

Figure 4: Framework for multi-effect processing.

- **Multi-chorus.** A computationally light chorus effect was implemeted featuring up to eight independent voices. This was a development "from scratch" with the aim to create a good and natural sounding chorus effect. The signal processing was block-based.
- **Univibe.** Based on the *behavior modeling*-technique a vintage guitar effect of the 1970s was emulated using block-based processing. For this effect two frequency notches, realized by first-order all-pass filters, are (slowly) varied in their center frequencies. The expansion to two channels with slightly different parameters animates the sound in a unique matter.
- **Fender tone stack and Marshall preamp.** Two subcircuits found in typical guitar amplifiers were the subject of this thesis: the tone control circuit of an early Fender amplifier and the distorting preamplifier of the Marshall JCM900. Using state-space descriptions circuit-based models of these devices were implemented, that emulate the sound true to original in real-time. A detailed description is presented in [4].
- **Feedback Canceller.** In this work the GEB1 was for the first time not used for effect processing. With the objective of suppressing acoustical feedbacks in small public address systems, a robust algorithm was implemented featuring automatic detection of suspicious frequencies and cancellation, using narrow-band notch filters. For fast reaction to arising feedbacks the processing was done sample-by-sample.

6. DISCUSSION AND OUTLOOK

The developed platform is a useful and field-tested equipment - but has room for improvement. In the upcoming redesign some changes will take place. First of all, the currently used audio codec will be replaced by a different device with a higher signal-to-noise ratio. Regarding the modeling of (high gain) guitar amplifiers, this is an expedient investment, because the input signals (and input noise) receive a very high amplification by the algorithms. This leads to an annoying noise floor in the current assembly, when using higain models. Other changes intend to increase both electrical and mechanical stability. Additional features include SDRAM and USB support e.g. for parameter exchange with a host computer.

For further use many interesting applications are imaginable. Beside even more audio and guitar effects, telecommunication tasks are also possible, e.g. coding algorithms or base band transmission.

7. CONCLUSION

We have presented the development of a DSP platform tailored to audio and especially guitar effect processing. This platform, named *Guitar Effect Board 1* (GEB1), contains an embedded signal processor, an audio codec and a flash memory. The realized hardware features six (free programmable) user control elements and is small, robust and cheap. The special needs of audio effects processing were explained, as well as their realization in the design.

We affirmed that effect algorithms are an excellent educational material that can be used to learn the techniques of digital signal processing and system theory. The utility in education was demonstrated by examples of projects which were successfully implemented on our DSP platform by students. A short outlook specifies possible improvements that will be considered in an upcoming redesign.

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