TDT4258 Assignment 2 Group 1

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1 Abstract

For this assignment we wrote a program producing distinct sounds when buttons on the microcontroller where pressed. The program was witten in C without a Operation System. We used the internal DAC to make audible audio waves.

A high-level programing language like C gives you a nice abstraction from the hardware while still maintaining direct control. Compared to assembly it is much simpler to create abstract datatype and dealing with variables instead of registers.

Programming without an Operation System gives us full control over the hardware, but it does not provide any abstractions, like device drivers. This requires us to implement a partial driver for all the devices we want to use. In this assignment we use LEDs, buttons and DAC.

Contents

2 Introduction

The second assignment introduces us to a new hardware device, the ABDAC. We also utilize the the buttons and LEDs used in assignment one. This gives an introduction to programming with audio devices and how to produce audio digitally.

To the development environment from assignment one we add the GCC C-compiler.

The gist of this assignment is to produce different sounds when the buttons on the board are pressed. The program should be implemented in the C programming language without the support of a Operation System. We implemented two modes. The first mode is a 7-note piano. The second is a playback function which plays a different predefined sample for each button. The button SW0 is used to toggle between the modes.

In order to make the program energy efficient the CPU is set to sleep and the DAC is shut down when a tone is not playing.

3 Overview of the Solution

3.1 States

3.1.1 Modes

The program has two main modes. These are Piano and Playback. To switch between state the **SW0** is pressed.

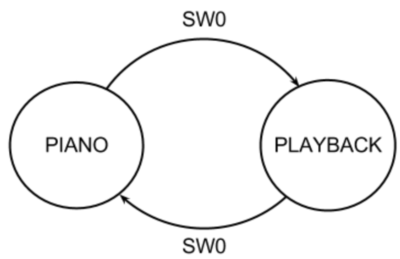


Figure 1: State diagram for modes

3.2 Datastructures

In order to play sounds the sound samples has to be saved. A sample contains four tracks where each track is a series of notes in a linked list.

3.2.1 Note structure

A musical note is represented in the program by the datatype note_t defined in **note.h**.

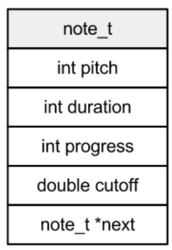


Figure 2: Datatype for notes

This type is used as a linked list to produce a track. The pitch is a number which relates to the frequency of the note. Pitches are defined in **tone.h** (C, D, .., C2, C3, ...). The duration is how long the note lasts, durations are also defined in **tone.h** (WHOLE, HALF, FORTH, etc.). The pogress field is a state variable used when the tone is played in order to know when its done, its always set to 0. The cutoff is to let different tones have different quality. The range of the cutoff is 0.0 - 1.0 where 0.5 is a very *staccato*[1] tone and 1.0 is *glissando*[2]. The value 0.875 is used for ordinary notes.

3.2.2 Sound Tracks

The **playback.c** file contains a array, *tracks*, of constant size 4. This array has pointers to the current point of each track. A track is a linked list of notes. The list is NULL terminated. As there are 4 tracks, a sound sample can play four tones at a time.

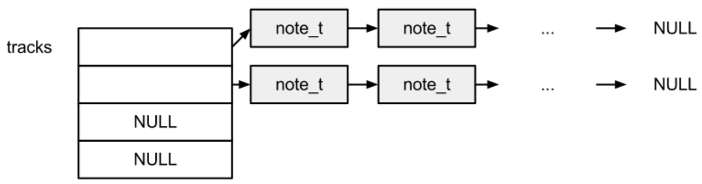


Figure 3: Illustration of tracks setup with 2 tracks

3.2.3 Sound Samples

The 7 different soundsamples are contained within a array of function pointers inside the interrupt.c file. Each function initilizes the tracks array by using the **set_track** function.

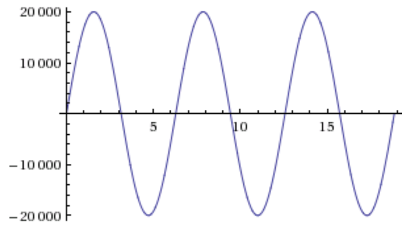
3.2.4 Sine Table

As the abdac interrupt routine is on a deadline, it has to conserve it's computing. Computing a **sin** function on every interrupt is wastefull and to timeconsuming. To make this a constant operation at runtime a sine table is computed on startup and stored in the **sine_table** in the **samples.c** file.

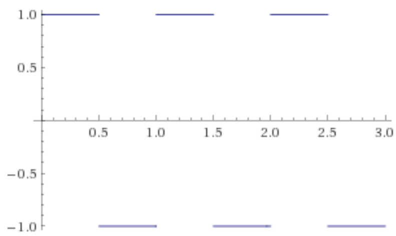
3.3 Waves

To make sound waves we need some functions producing wave signals. The following waves are implemented in **samples.c**.

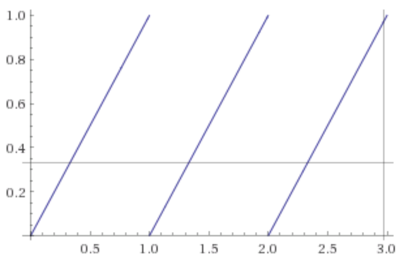
[1] A shortened duration of the tone
[2] When tones glides into each other



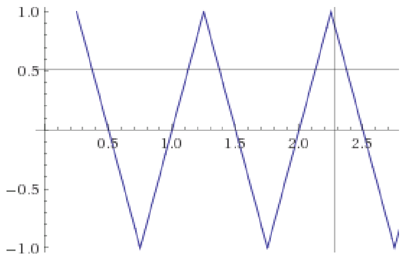
(a) Sine wave



(b) Square wave



(c) Sawtooth wave



(d) Triangle wave

4 Solution

4.1 Sound generation

To generate sound we used the internal ABDAC(Audio Bitstream Digital-to-Audio-Converter) on the AVR32 board. It takes a sequence of samples and converts it to an analog signal amplifies and outputs it. The ABDAC uses 2 of the pins on the PIO port B to send signals to the output, and the 6th clock of the Power Manager to generate interrupts that processes the samples. The clock is set up with oscillator 0 as the source and no division of the frequency. This gives us a clock of 20 MHz and a sample rate of 20MHz / 256 = 81.920 kHz on the ABDAC

```c
    // Register interrupt handler
    register_interrupt((__int_handler)(abdac_isr),
                AVR32_ABDAC_IRQ / 32, AVR32_ABDAC_IRQ % 32,
                    ABDAC_INT_LEVEL);

    // Disable PIO
    piob->PDR.p20 = 1;
    piob->PDR.p21 = 1;

    // Enable ABDAC
    piob->ASR.p20 = 1;
    piob->ASR.p21 = 1;

    // Set the clock to use Oscillator (OSC0 and OSC1 is 20MHz and 12MHz)
    volatile avr32_pm_t *sm = &AVR32_PM;
    volatile avr32_pm_gcctrl_t *clock = &sm->gcctrl[6];

    clock->oscsel = 0;
    clock->pllsel = 0;
    clock->cen = ON;
```

The ABDAC is turned of when it is not used, to save power and keep the output silent.

```c
    dac->CR.en = OFF;
    dac->IER.tx_ready = OFF;
```

To send samples to the ABDAC, in the interrupt routine, each of the stereo-channels are written with the corresponding sample data. We have only used mono sounds in our implementation, so the channels are written with equal samples.

```c
    dac->SDR.channel0 = sound;
    dac->SDR.channel1 = sound;
```

The waveforms that are used as base for the samples are sine, triangle, sawtooth, square and white noise. They are implemented in samples.c as mathematical functions of a counter that ticks for a constant length. To make it possible to play multiple tones at once, the soundwaves are accumulated before written to the channels.

```
for (i=0; i<7; i++) {
    sound += get_tone_pitch(i);
}
```

This pretty much works the same way for playing multiple sounds at once in the playback mode, but the playback mode also allows accumulating sounds with different base waveforms.

5 Discussion

This assignment was easier getting started with, because we knew the tools, and the programming language. Off course handling I/O devices without an Operating System was a new experience. We found having a little knowledge of music- and wave-theory quite favorable, but the recitation slides and the compendium worked as good summary and reinstatements of the theory.

6 Conclusion

During this assignment we have experienced the advantage of higher level programming languages. We followed the recommended approach, and found solving the previous assignment i C to be a fraction of work compared to using assembly. As with the last assignment, we were reminded about the challenges when working with analog signals, and some new challenges were faced when working with digital audio, and real-time requirements.