Final Degree Project

Bachelor's degree in Industrial Technology Engineering

Music Quartet based on a PSoC

REPORT

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Abstract

This report details the migration process of the Quartet code from a Microchip PIC24 microcontroller, which was previously migrated by Pau Mendieta from a Microchip PIC18 microcontroller, to a CY8CKIT-042-BLE PSoC 4 microcontroller manufactured by Cypress Semiconductor. It also explains how several improvements, both in sound quality and program functionality, have been implemented into the final program once the migration was completed.

The original Quartet code, developed by Victor Timofeev, synthesized 4 different voices (two guitars, a violin and a bass) coordinated by a conductor task, each of which played from its own music sheet, which was stored in ROM. The sound samples were generated using amplitude modulation: the characteristic waveform of the instrument (also stored in ROM) and its amplitude envelope (generated through software) were multiplied, and later exported through an 8-bit PWM running at 78 kHz. These could later be played on speakers with the help of an RC filter.

The project was carried out by continuously testing the software on the actual microcontroller. Moreover, data was extracted from these tests and compared with the simulation of Pere Domenech's PIC18 code. A great use of the example projects provided by Cypress Semiconductor, which demonstrate the features of the PSoC 4 through simple applications, has also been made. It was also essential to analyze different signals with an oscilloscope at the laboratory.

The program obtained after completing the project exports the audio signal through a 16-bit PWM running at 92 kHz and includes an extra voice (a second violin), more accurate characteristic waveforms and improved amplitude envelopes, as well as added functionality, such as an extra octave and a wider note frequency range.

The report first presents the basic operation of the original code, which was carefully studied at the beginning of the project. Then, it explains how the OS (FreeRTOS) and the program were migrated, along with the extensive troubleshooting process that followed. Moreover, it details the improvements that were made once the code worked correctly. Finally, suggestions are made for future improvements.

Contents

ETSEIB

1. Glossary

BLE: Bluetooth Low Energy

BPM: Beats Per Minute

CAN: Controller Area Network

CPU: Central Processing Unit

DAC: Digital to Analog Converter

IDE: Integrated Developing Environment

ISR: Interrupt Service Routine

LED: Light-Emitting Diode

OS: Operating System

PIC: Peripheral Interface Controller

PSoC: Programmable System on a Chip

PWM: Pulse Width Modulation/Modulator

RC: Resistor-Capacitor

RF: Radio Frequency

ROM: Read-Only Memory

RTOS: Real Time Operating System

SCB: Serial Communication Block

SRAM: Static Random Access Memory

TCPWM: Timer Counter Pulse Width Modulator

UART: Universal Asynchronous Receiver-Transmitter

UDB: Universal Digital Block

USB: Universal Serial Bus

2. Preface

2.1. Prior works related to this project

The origin of this project goes back to Victor Timofeev's usage example of the OSA RTOS **[1]**. Timofeev designed a program, which he named "Quartet", that was able to synthesize a 4-channel melody and export it using an 8-bit PWM running at 78 kHz so that, with the help of an RC filter, it could be played on speakers or headphones. The original code was written for a PIC16 microcontroller and used the OSA RTOS as an operating system.

In 2015 Pere Domenech, a student at ETSEIB (Escola Tècnica Superior d'Enginyeria Industrial de Barcelona), succeeded in migrating Timofeev's code so that it could be executed using a PIC18 microcontroller **[2]**. He chose this microcontroller in particular because it is widely available at the Department of Electronics at said university.

Two years later another student, Pau Mendieta, migrated Pere Domenech's code so that a PIC24 could run it **[3]**. This meant a change from an 8-bit CPU to a 16-bit CPU and, therefore, he expected to improve the sound quality by increasing the PWM resolution. In the end, however, only a 9-bit PWM running at 31.25 kHz was achieved and, thus, the audio quality did not improve significantly. He also made the necessary changes so that the code could run on FreeRTOS **[4]**, a more universal operating system which is easier to migrate.

Other related projects include Juan Gallostra's *RF Music Festival: orquesta basada en microcontroladores PIC18 y RF* **[5]** and Joan Calvet's *CAN Music Festival: Orquestra Basada en Microcontroladors PIC18 i un bus CAN* **[6]**. The former designed a musical orchestra where each instrument was individually synthesized by four different PIC18 microcontrollers and the conductor sent commands or notes to each instrument through RF. The latter pursued the same objective, but using a CAN bus instead of RF as a means of communication.

2.2. Motivation

The reasons that justified the creation of this project are the following:

- Learn how to program a microcontroller in C language.
- Learn how to use a PSoC and get familiarized with its specific IDE.
- Understand how a RTOS, more specifically FreeRTOS, works.
- Apply the knowledge and skills learned in an engineering degree to a more practical case.
- Upgrade an existing code so that it produces an improved result.
- Learn a method of synthesizing music.
- Deepen the personal knowledge in the fields of electronics, programming and music.

2.3. Previous requirements

Since the original code is written in C language, it is essential to have a certain degree of understanding of this programming language before the actual project can begin. In this case, an introductory tutorial was consulted **[7]**. Knowing the basics of digital electronics is also necessary.

Moreover, one has to familiarize themselves with the PSoC specific IDE, PSoC Creator 4.1 **[8]**, in order to develop the code, transfer it to the processor and manage the chip's configurable hardware blocks. With regard to these components, some of their datasheets have to be studied carefully, namely the PWM **[9]**, TCPWM **[10]**, SCB **[11]**, Clock **[12]**, Interrupt **[13]** and Pins **[14]** components.

In addition, it is important to study the pin distribution on the kit **[15]** whenever cable connections have to be made and to understand the fundamental behavior and characteristics of the FreeRTOS operating system **[16]**.

3. Introduction

3.1. Objectives

This is the first time a PSoC is used in the Department of Electronic Engineering at ETSEIB. Therefore, the first objective of this project is to learn how to use a CY8CKIT-042 PSoC 4 Pioneer Kit and acquaint oneself with its features.

The second aim is to adapt both the operating system and the Quartet code so that the PSoC 4 is able to run them correctly. Since FreeRTOS is compatible with this microcontroller, this migration should not be excessively difficult.

Another important objective is to make use of the PSoC 4's advantages over the other microcontrollers to improve the resulting sound quality. This can be done in numerous ways, as it will be explained further on, but the most important one is to increase the PWM bit resolution.

Lastly, another important feature of the PSoC is that it includes configurable analog and digital blocks along with the CPU core. It was thus initially thought that some of the operations that were originally implemented through software could be accomplished with hardware instead, hence relieving some of the CPU's computational effort. However, it was concluded that the blocks required to do this were only available in a more complex version of the PSoC used.

3.2. Scope of the project

A significant amount of time in this project will be devoted to learning about the C programming language, FreeRTOS, the features of the PSoC, etc. It will also be essential to study carefully and understand the original Quartet code in order to migrate it correctly.

All software modifications will be constantly tested using the corresponding hardware. This is made possible thanks to the USB connection between the PSoC 4 and a PC, which allows a fast debugging, testing and analysis of the code. As a downside, unlike the PIC microcontrollers, the code cannot be simulated, it must be always tested on the actual chip.

Moreover, modifications of the original Quartet code will be considered, especially those that help improve the sound quality of the synthesized instruments. Nevertheless, these will only be studied once the migration process has been completed.

The project will conclude once the code has been correctly migrated and the sound quality of the final result is empirically verified to be superior to the older projects mentioned beforehand. Finally, further improvements will be suggested for future projects.

4. Basic Quartet Operation

Before introducing the migration process, it is essential to understand the basic operation of the Quartet program.

4.1. The Conductor

The conductor is a task whose purpose is to coordinate all instruments and set the tempo of the song. To do so, the conductor sends a semaphore to each instrument for every command or note in their note list and then creates a delay that sets the tempo.

4.2. The Instruments

There are a total of 4 voices (a violin, two guitars and a bass) each of which is represented by a task. These tasks read and execute every command/note when the conductor tells them to.

Every voice has a note list stored in ROM (located in the bach1067.h file), where every element is either a command (pause, set octave, repeat, etc) or a musical note. In the sinus.h file there is a table (also stored in ROM) for every instrument that contains one digitized period of its characteristic waveform (see Fig. 4.1).

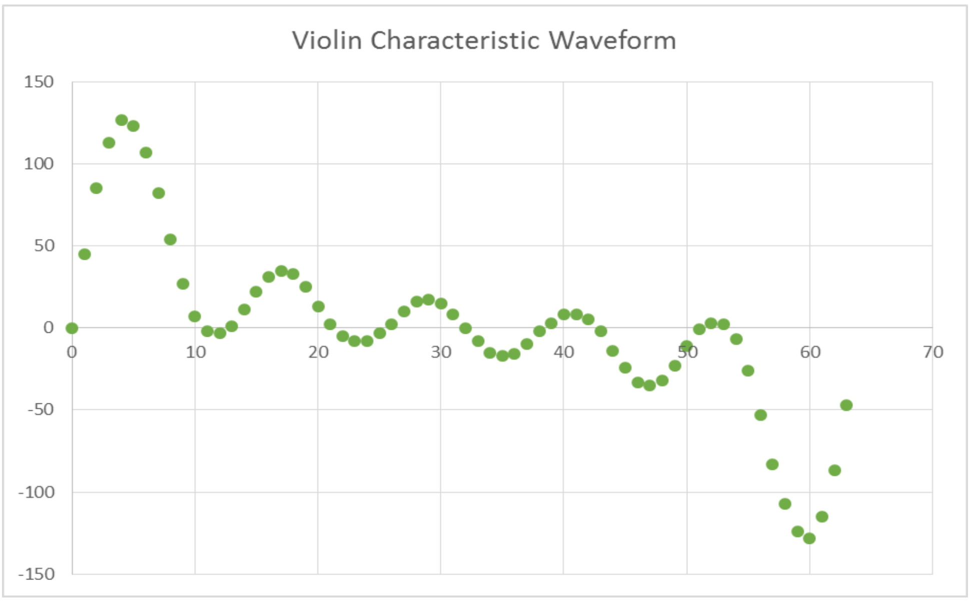


Figure 4.1: Violin characteristic waveform. Source: Victor Timofeev's data in sinus.h

4.3. The Synthesizer

The sound synthesizing is done during processor interrupts using a technique known as amplitude modulation synthesis (see Fig. 4.2). It consists in multiplying two signals: the modulator signal and the carrier signal. In this case, the carrier signal is a periodical repetition of the instrument's characteristic waveform (see Fig. 4.3) and the modulator signal is the instrument's envelope (see Fig. 4.4). This is done to simulate the natural evolution of sound. In the original code, this multiplication was done through a macro called MUL(), which was written in assembly code so as to speed up calculations.

Once every sample for each voice is generated, they are added and the resulting value is divided by 4.

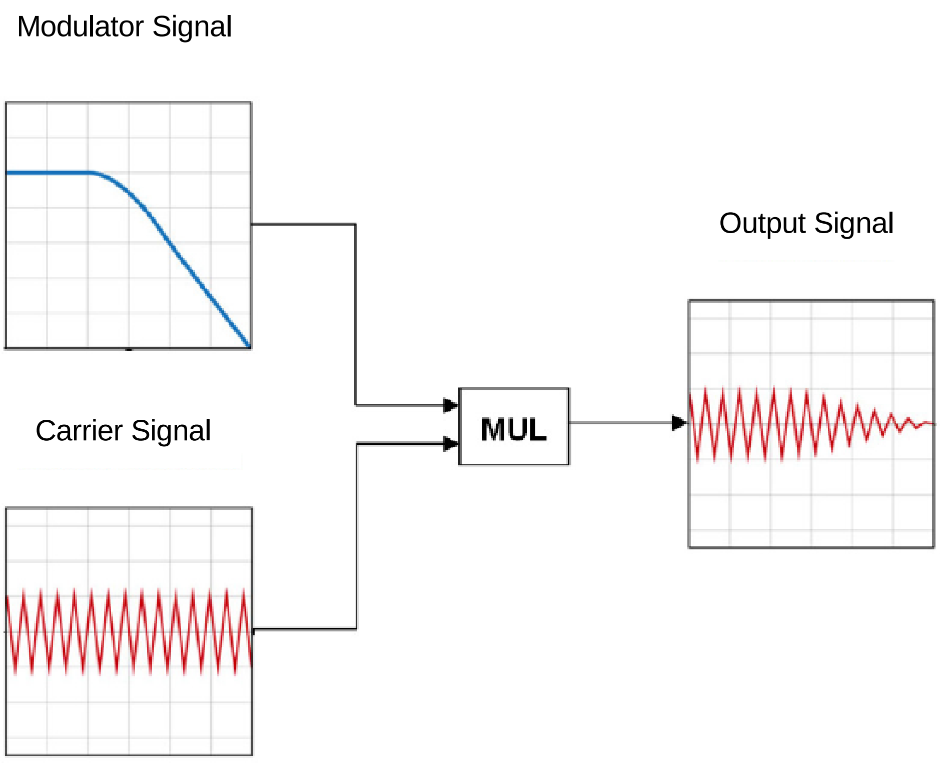


Figure 4.2: Amplitude modulation synthesis conceptual diagram. Source: [2]

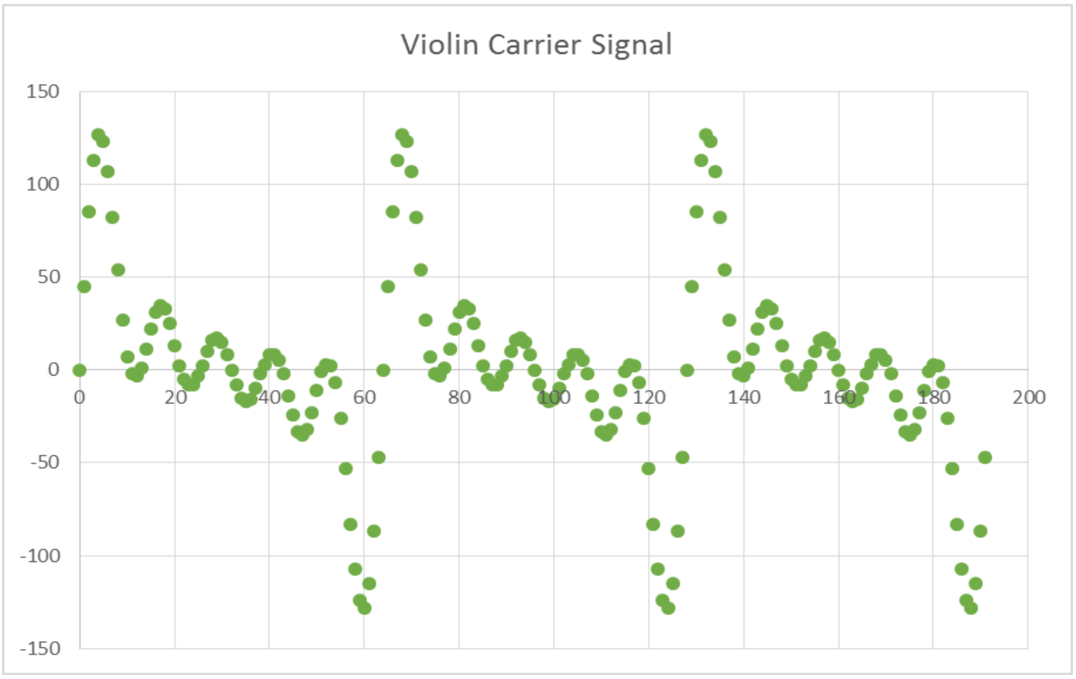


Figure 4.3: Violin carrier signal. Source: Victor Timofeev's data in sinus.h

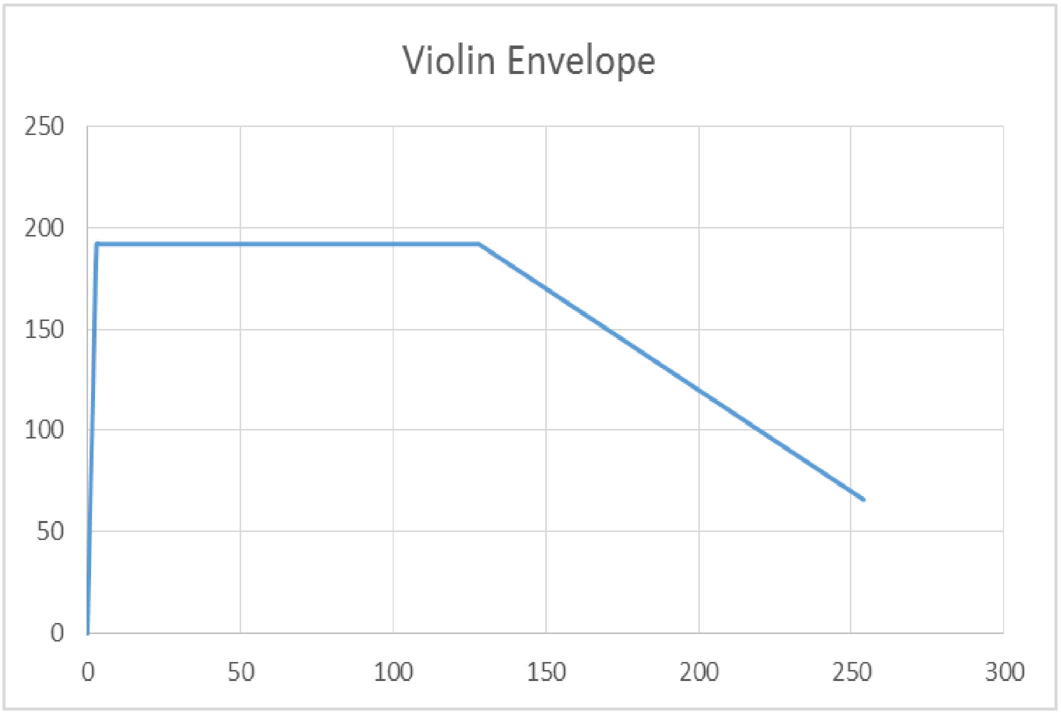


Figure 4.4: Violin envelope. Source: Victor Timofeev's original Quartet code

4.4. Pulse Width Modulation

Once the synthesizer has generated the output signal, this information is stored and sent within the width of a pulse (see Fig. 4.5). This is known as pulse width modulation. Finally, the resulting PWM signal passes through a low-pass filter, which eliminates the low voltage fraction of the signal, and thus the waveform is recovered and reproduced through speakers.

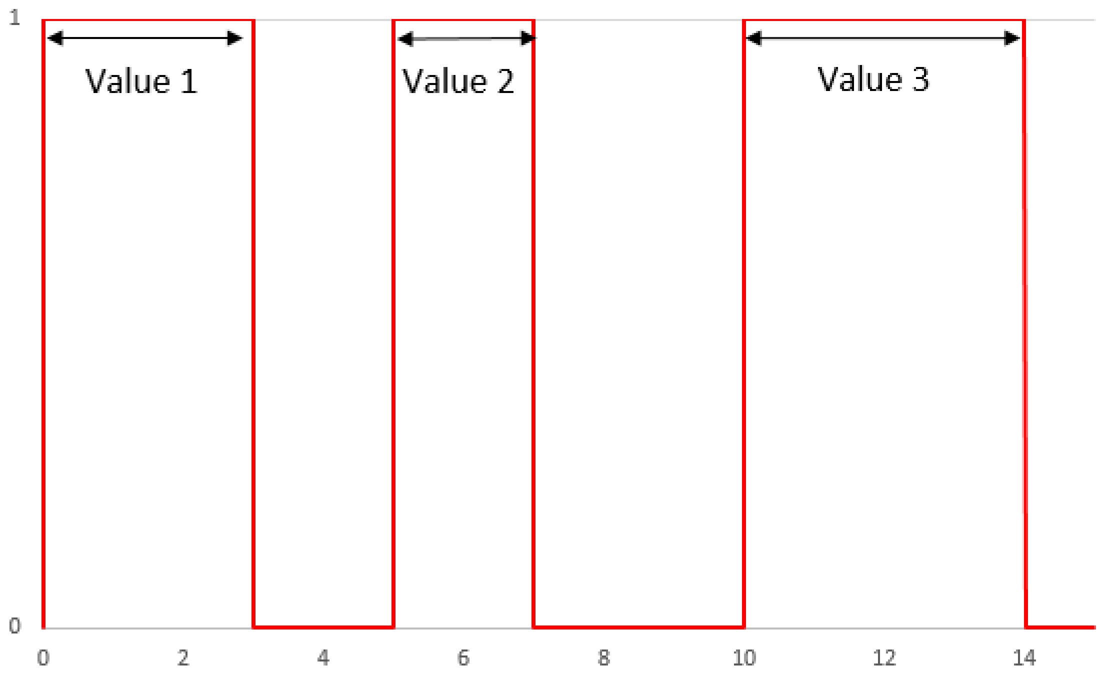


Figure 4.5: Example of information being stored and sent using PWM. Source: own.

5. Software migration

5.1. FreeRTOS migration

The migration process began by testing an example project provided by Cypress Semiconductor in which FreeRTOS was specifically implemented for the PSoC 4 BLE **[17]**. The project was built using PSoC Creator 4.1 **[8]**.

This example basically detects when the SW2 switch or the CapSense Slider are pressed and sends a message through one UART accordingly. Once tested correctly it was possible to move on to the configuration of the OS.

5.2. FreeRTOS configuration

For the Quartet code to work correctly it is necessary to change some parameters inside the FreeRTOS configuration. This is done by editing the FreeRTOSConfig.h header file. Apart from disabling pre-emption (see section 5.2.1), at first all other options were left at their default setting.

5.2.1. Co-operative scheduler

FreeRTOS allows the pre-emption of tasks, which means that during a tick interrupt (the OS's own software-generated interrupt) a higher priority task can force the running task out of running state, replacing it.

In case of the Quartet code, a context switch (changing which task is running) only happens when the instruments enter the blocked state to wait for the conductor to give a semaphore to them. Therefore, the co-operative task scheduler was chosen, which is equivalent to disabling pre-emption.

5.3. Introducing the TCPWM block

The next logical step in the migration process is to test the PSoC 4's TCPWM block and understand how it works. In order to do this, a TCPWM example provided by Cypress Semiconductor was consulted **[18]**.

In this example project, the brightness of a LED is controlled using the TCPWM block. Once it was successfully tested in separate, all necessary code and configuration was then moved to the FreeRTOS example to check if both projects were compatible. More specifically, it was necessary to:

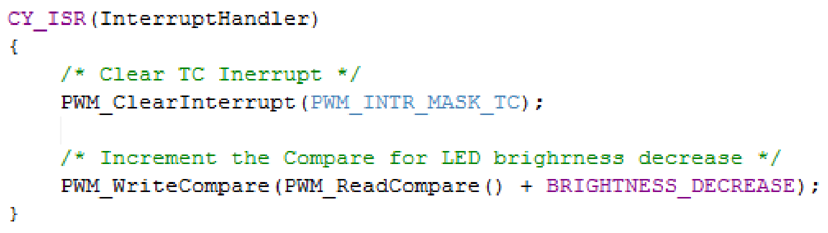
- Replace all previously existing blocks in the TopDesign.cysch file with the TCPWM block (see Fig. 5.2).
- Enable the interrupt component linked to the TCPWM block (TC_ISR). This is done using the following command in the main() function:

  ```
  TC_ISR_StartEx(InterruptHandler);
  ```

- Start the TCPWM component and configure its parameters. This is also done in the main() function, even though the latter can also be done by directly editing the block's properties:

  ```
  PWM_Start();

  PWM_WritePeriod(65300u);
  ```

- Edit the ISR function so that it handles the TC_ISR interrupt (see Fig. 5.1).
- Remove unnecessary code from the first example.



```
CY_ISR(InterruptHandler)
{
    /* Clear TC Inerrupt */
    PWM_ClearInterrupt(PWM_INTR_MASK_TC);

    /* Increment the Compare for LED brighrness decrease */
    PWM_WriteCompare(PWM_ReadCompare() + BRIGHTNESS_DECREASE);
}
```

Figure 5.1: ISR function that handles the TCPWM interrupt. Source: [18]

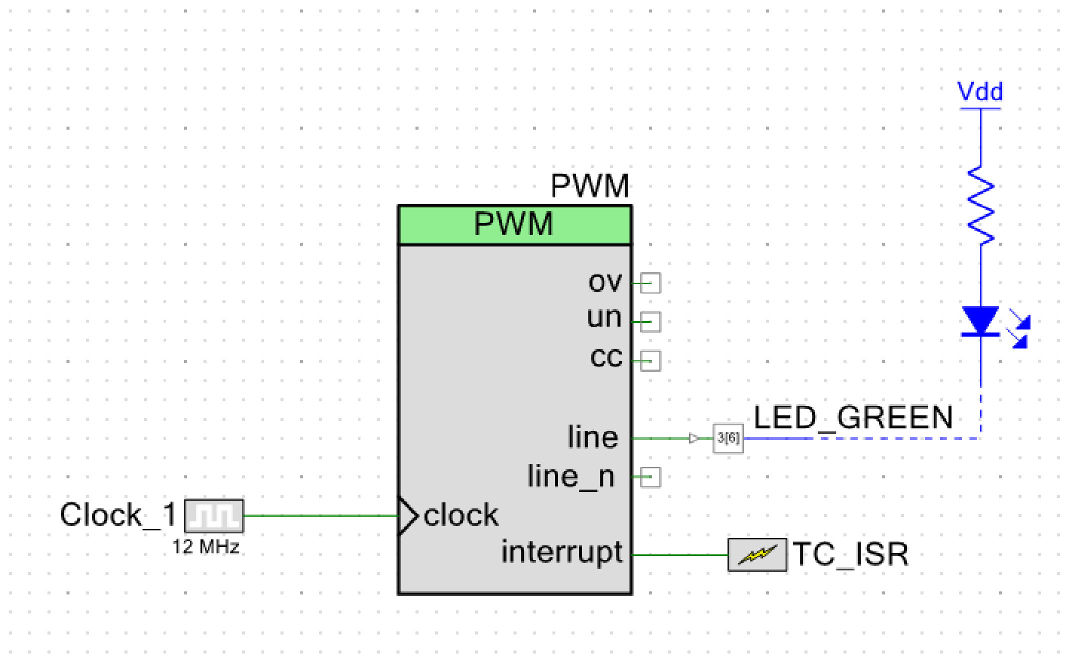


Figure 5.2: TCPWM block, located in the TopDesign.cysch file, which controls the brightness of a green LED. Source: [18]

After completing these steps, the TCPWM example worked as expected inside the FreeRTOS project. Now that the operating system's functions were available and the TCPWM block was implemented, all in the same project, it was possible to proceed to the migration of the actual Quartet code.

5.4. Quartet code migration

The first step in the code migration was to copy all Quartet-specific files (sinus.h, bach1067.h and elochka.h) into the project's folder. After doing so, Pau Mendieta's code **[3]** had to be implemented into the project's main.c file. Nevertheless, some modifications were still necessary.

5.4.1. Changes made to the amplitude modulation

As explained in the previous chapter, the final step in the synthesizing process is to multiply two signals, which was originally done using assembly language to multiply two variables. Since migrating this piece of code would not be trivial (the assembly language would be drastically different) and the PSoC 4's processor is more powerful than the PIC24's, this operation was implemented as a simple product between two variables in C language, which is more costly (computationally).

5.4.2. Changes made to the PWM

All of the code related to the sound synthesizing had to be introduced into the function that handles the interrupts while respecting the structure seen in section 5.3 (see Fig. 5.1). The main difference is that the interrupt has to be cleared at the start of the function and that the pulse width is modulated with the PWM_WriteCompare(temp_dac) command, where temp_dac is the variable that holds the final synthesized sample of all 4 voices.

It is also important to fit the period of the PWM so that its frequency matches the desired value. In order to achieve a PWM frequency of 78 kHz while using a 24 MHz clock, the period has to be 300 (see calculation below).

$$PWM\ period = \frac{Clock\ frequency}{PWM\ frequency} = \frac{24000}{78} \cong 300$$

6. Quartet troubleshooting

The aim of the migration process described in the previous chapter was to achieve the same acoustic result with the PSoC 4 as the one obtained with the PIC microcontrollers. The migrated program did produce sound, but it was significantly different than what was expected. The main issues were:

- The sound was highly saturated.
- The notes played by the program were of a much higher frequency than they should be.
- The melody could only be distinguished when playing one instrument at a time. If two or more instruments were played simultaneously they fell out of synchronization and the sound became distorted.
- The tempo was far slower than the original.
- The sound had an overall poor quality.

In this chapter, the process of detecting and solving the mistakes that caused these faults will be explained in detail. This has been the most challenging and time-consuming part of the project. Hence, to simplify it, several characteristics of the resulting sound (listed below) have been evaluated for every possible fix made to the project, in accordance with the legend presented in table 6.1.

- **Clarity:** A highly saturated audio signal has a loud cracking sound added to it (similar to white noise). This makes it bothersome to listen to and sometimes even makes it difficult to identify the melody that is being played. High clarity will imply little saturation and vice versa.
- **Accuracy of note frequency:** Musical notes have a characteristic frequency. Therefore, a synthesizer that works correctly should generate an audio signal with the frequency of the desired note.
- **Quality:** This is a more subjective parameter. Higher quality here will imply a more detailed and realistic sound.
- **Correct interaction between instruments:** Troubleshooting is made easier by analyzing a single voice, but it is also important to check that the sum of all 4 voices plays correctly.

Symbol	Meaning
+	Slight improvement
++	Significant improvement
+++	Critical improvement
/	No difference
-	Slight worsening
--	Significant worsening
---	Critical worsening
?	Not tested

Table 6.1: Legend of the symbols used to describe the improvement/worsening of the audio result

Evaluation tables are always relative to the best configuration that has been found at a particular point. Obviously, changes that have a positive effect on the result are kept and those that do not are discarded.

6.1. Adjusting temp_dac

At first, it was thought that temp_dac, the variable that holds the final synthesized sample, was not properly adjusted to the range of values accepted by the PWM. Since the PWM period was set to 300, any temp_dac value outside the range of 0-300 would produce unexpected results.

In the original code, once all of the 4 voices were added the result was divided by 4 and, later, an offset of 128 was added. This last operation was done because the waveform tables include negative values. However, since the multiplication operation used in this case is slightly different, the values produced by the migrated code were of a higher order, and thus the variable had to be divided by a higher number.

Several combinations of divisor and offset were tested and evaluated empirically. With a single active voice, the best result was achieved by dividing by 512 (shifting 9 bits to the right) and adding an offset of 220. The sound became slightly less saturated, but most of the issues persisted (see table 6.2). It was therefore reasonable to assume that there was a more serious problem with the migrated code that remained undetected and was responsible for the poor quality of the result.

Clarity	+
Accuracy	/
Quality	/
Correct interaction	?

Table 6.2: Sound evaluation after adjusting the temp_dac variable

6.2. Increasing clock speed

Before starting a more detailed analysis of the migrated code's behavior, it was decided that it would be interesting to increase the frequency of the clock that controls the TCPWM block. Until now, it had been running at the same frequency as the external clock (EXTCLK), 24 MHz.

However, the TCPWM block allows clocks to operate up to a maximum of 48 MHz **[10]**. Choosing this clock speed means that, if the same PWM frequency is kept, the PWM period should now be 600, thus doubling its resolution. In other words, the temp_dac variable would be able to hold double the amount of different values than it previously could. This would be equivalent to a 9-bit resolution.

After testing this modification (see recording #1 of the annex), the quality increased slightly but so did the saturation (see table 6.3). Still, overall this version of the migrated code sounded better.

Clarity	-
Accuracy	/
Quality	+
Correct interaction	?

Table 6.3: Sound evaluation after increasing the clock speed to 48 MHz

6.3. Extracting data from the PSoC 4

Once the PWM resolution and the adjustment of the temp_dac variable were discarded from being responsible for the major sound issues, a more detailed and comprehensive analysis of the code’s behavior was deemed necessary.

Up to this point all data had been extracted from the PSoC using breakpoints in PSoC Creator 4.1. This procedure was enough to check the state of variables at certain points of execution, but it made saving large amounts of data almost impossible (saving data from each interrupt, for example). Therefore, it was necessary to find another way to extract information.

This was done by following an online tutorial that demonstrated how to implement the printf C function into the PSoC (by default the function compiles without errors, but it does nothing when called) **[19]**.

First, a UART (SCB mode) block was placed into the TopDesign.cysch file with its standard configuration. Then, it was assigned to the correct pins, P1.5 and P1.4 (see Fig. 6.1) and, afterwards, a cable connection was made from P1.5 to P12.6 and from P1.4 to P12.7 (see Fig. 6.2).

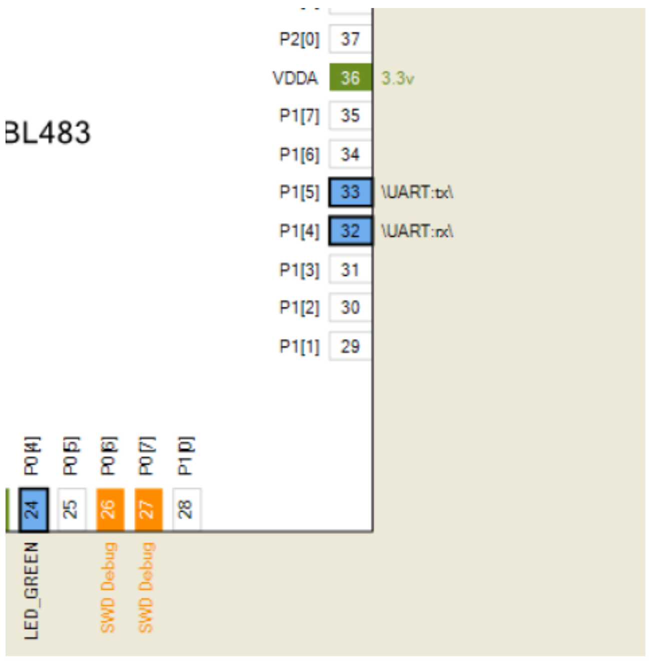


Figure 6.1: Pin assignment of the UART block. Screenshot taken from PSoC Creator 4.1.

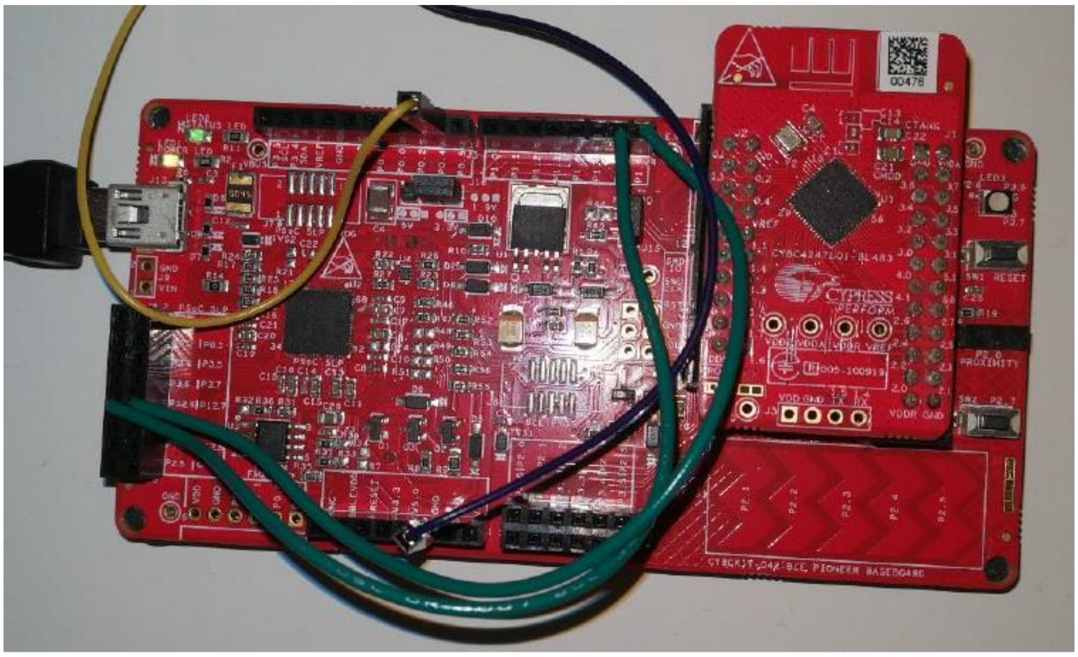


Figure 6.2: PSoC 4 after making the cable connections for the UART block (green cables). Source: own.

The following steps consisted in increasing the heap size, which is done in the “System” tab in the .cydwr file, and adding the _write() function to the main.c file (see Fig. 6.3). Moreover, the UART block has to be started with the UART_Start() command.

```
int _write(int file, char *ptr, int len)
{
    int i;
    file = file;
    for (i = 0; i < len; i++)
    {
        UART_UartPutChar(*ptr++);
    }
    return len;
}
```

Figure 6.3: Additional code necessary for the implementation of the printf function. Source: [19].

Finally, the program PuTTY **[20]** was installed in order to receive the information that the PSoC would be transmitting through USB.

6.4. Comparison between PSoC 4 and PIC18

In order to discover what had gone wrong in the migration process, a comprehensive comparison of the PSoC and PIC18's behavior was done. The PSoC's data was extracted using the UART block (as seen in section 6.3), while the PIC18 was analyzed by simulating Pere Domenech's code **[2]** with the use of MPLAB IDE **[21]**.

6.4.1. Testing conditions

Both systems were tested in the following conditions:

- Only one voice (the violin) was playing.
- The only note played was b0 (in Quartet notation) with a base scale of 0 and a duration of 4.
- The variables whose values were stored for every interrupt were:
    - **S2.f:** Works as an index for selecting values from the tables in sinus.h.
    - **temp1:** Holds the value read from the tables in sinus.h.
    - **S2.t:** Increases by one every 64 interrupts and is used to generate the amplitude envelope.
    - **temp2:** Holds the value of the amplitude envelope.
    - **temp_dac:** Holds the value after the amplitude modulation and before it is adjusted.

6.4.2. S2.f

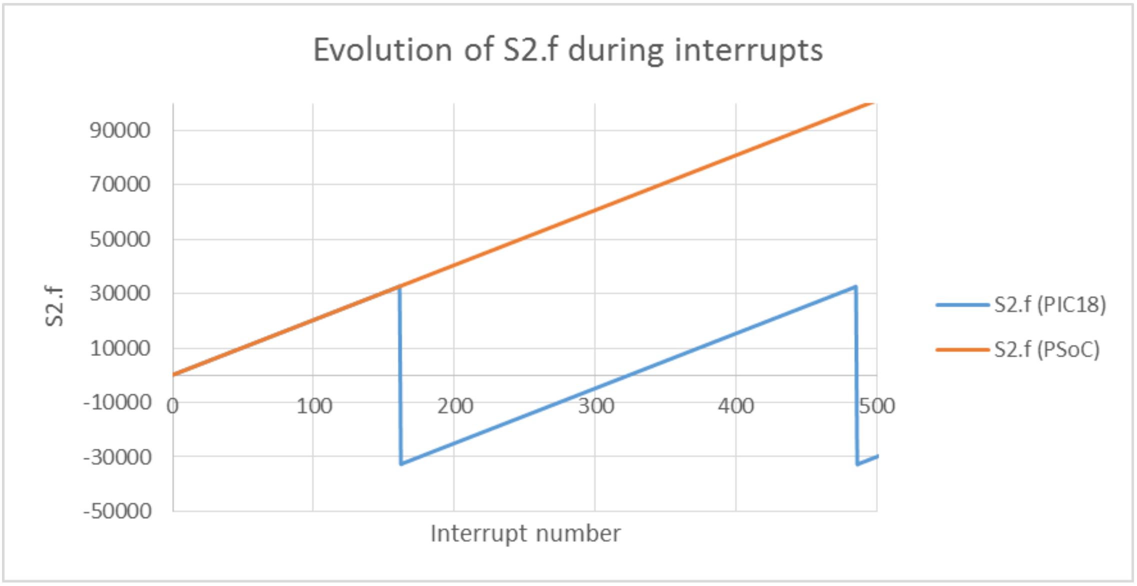


Figure 6.4: Evolution of the S2.f variable during interrupts both in the PSoC and the PIC18. Source: own.

The different evolutions of S2.f shown in Fig. 6.4 are simply due to how each microcontroller treats the integer type. On the one hand, the PIC18's integer type is 16 bits and thus has to reset itself to its lowest value once it reaches its highest permitted value (32768). On the other hand, the PSoC integer type is 32 bits by default, and allows S2.f to keep increasing.

Since this variable is used as an index, this difference should not be important. In any case, the code was executed after changing S2.f into int16 in the PSoC to check whether there was a difference in sound, but there was none.

6.4.3. Temp1

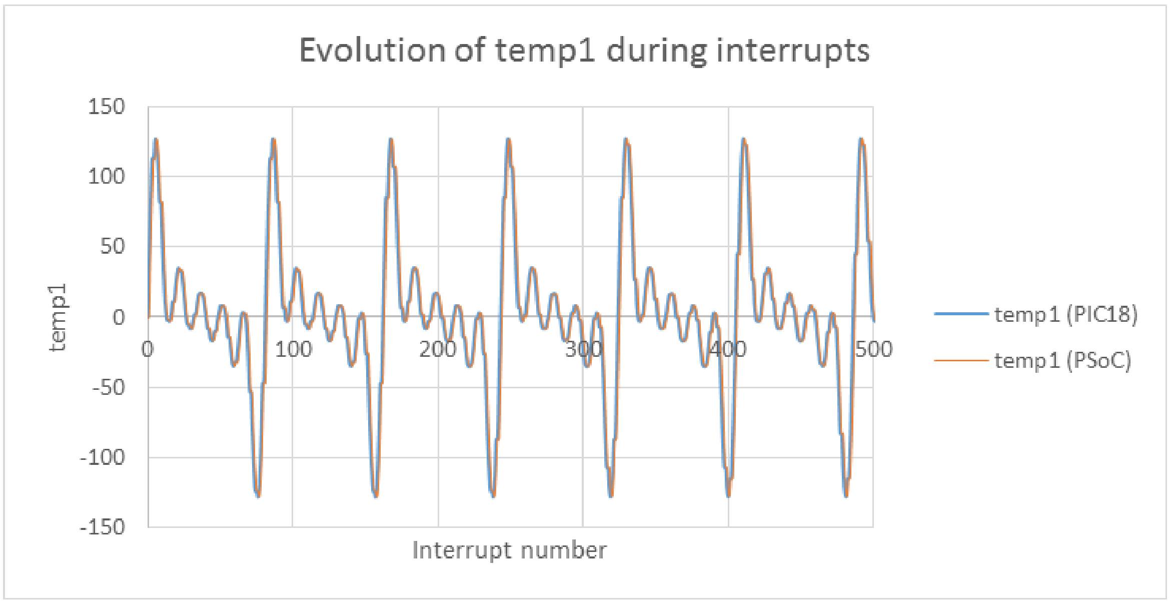


Figure 6.5: Evolution of the temp1 variable during interrupts both in the PSoC and the PIC18. Source: own.

As Fig. 6.5 illustrates, temp1 is read correctly from sinus.h.

6.4.4. S2.t

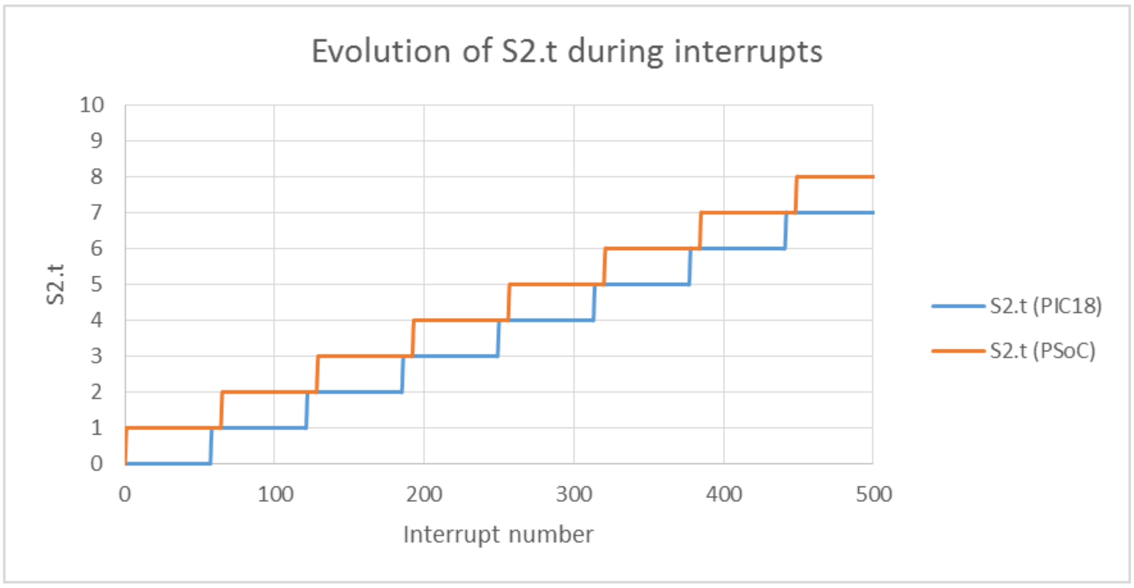


Figure 6.6: Evolution of the S2.t variable during interrupts both in the PSoC and the PIC18. Source: own.

Fig. 6.6 shows how the behavior of S2.t is the same, with only a phase difference between them. Several tests were made and this phase difference kept changing. This caused the envelope’s evolution to change, but only slightly, as it will be seen in the following sections.

It was possible to fix this phase difference by modifying the counter that controls the variable, but it did not have any influence on sound.

6.4.5. Temp2

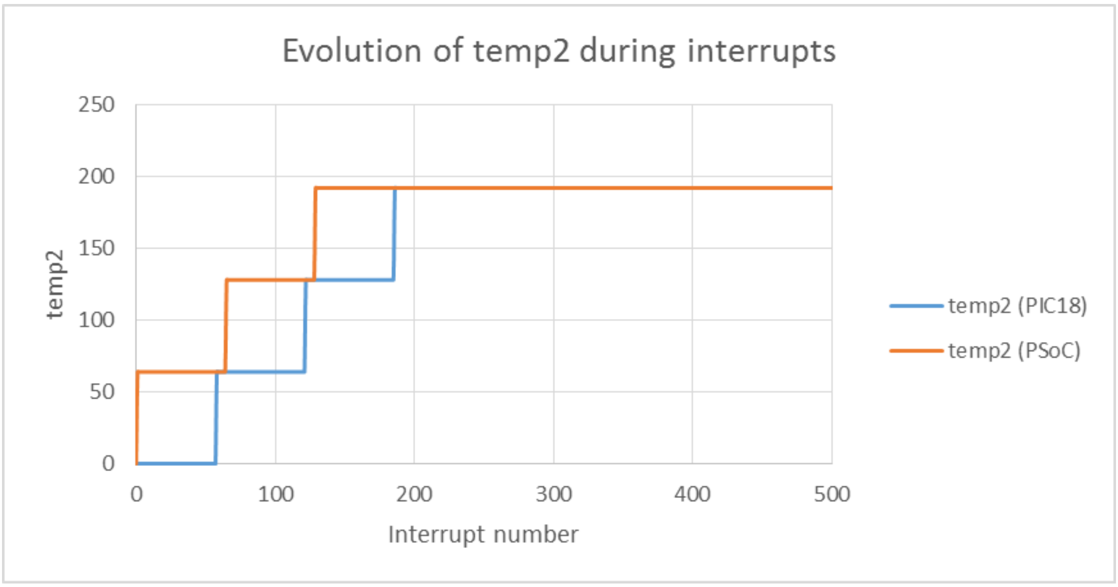


Figure 6.7: Evolution of the temp2 variable during interrupts both in the PSoC and the PIC18. Source: own.

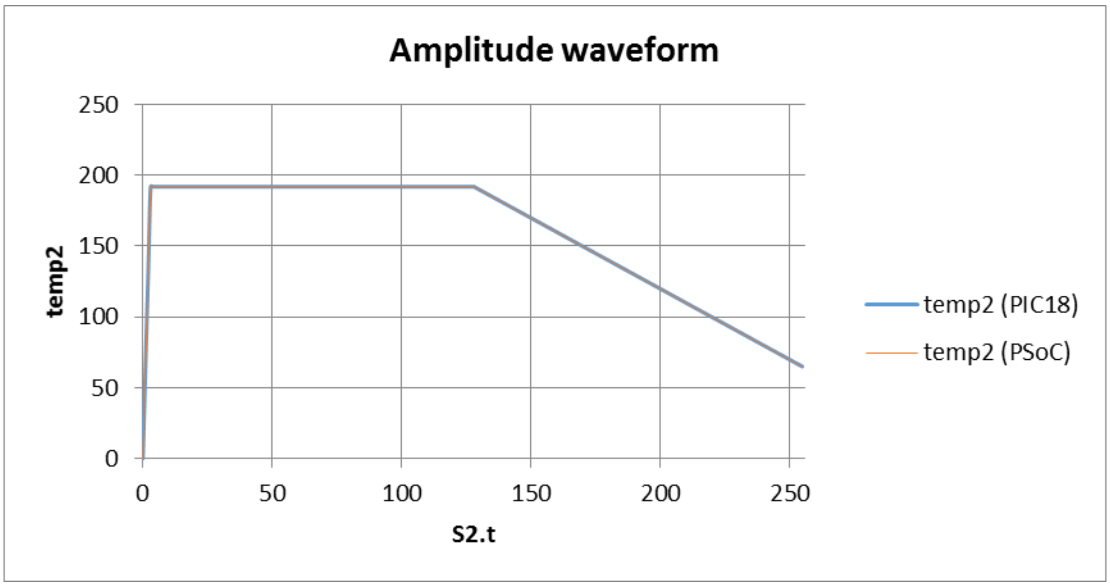


Figure 6.8: Temp2 plotted as a function of S2.t both in the PSoC and the PIC18. Source: own.

Temp2 displays the same phase difference due to S2.t (see Fig. 6.7), but the amplitude envelope is correct, as Fig. 6.8 shows.

6.4.6. Temp_dac

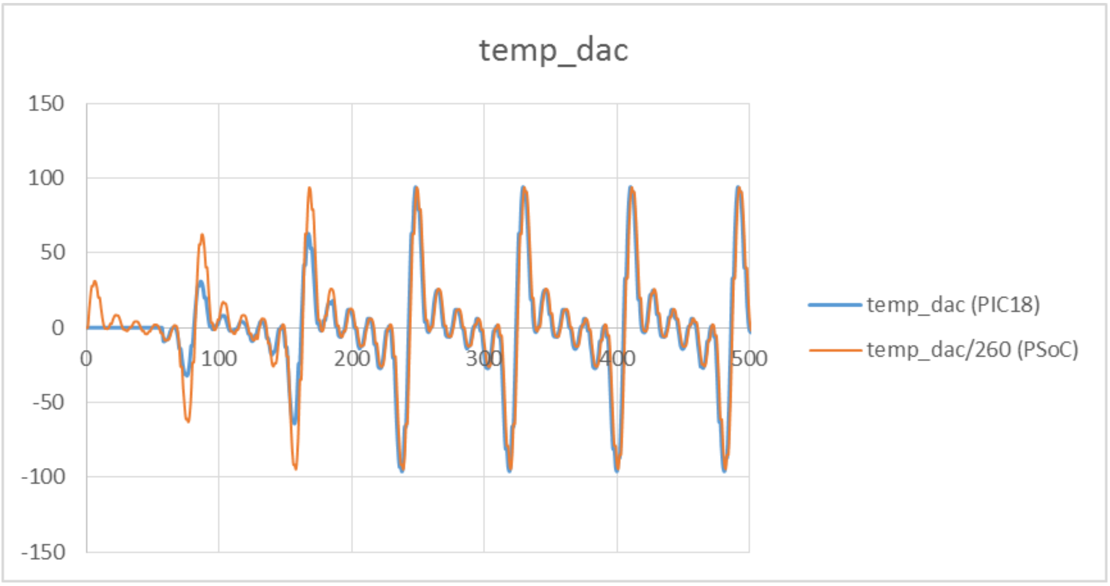


Figure 6.9: Evolution of the temp_dac variable during interrupts both in the PSoC and the PIC18. Source: own.

Here the PSoC's temp_dac had to be divided by 260 so that it could be compared to the PIC18. The difference seen in figure 6.9 is again due to the phase difference in S2.t. Otherwise, the variable evolution is correct.

6.4.7. Comparison conclusions

After comparing the behavior of the most important variables in the synthesizing process, it was concluded that:

1. The migrated program reads values from the tables in sinus.h correctly.
2. The migrated program creates the amplitude envelopes correctly.
3. The migrated program carries out the modulation operation correctly. There is only a difference in scale, which forces a different final adjustment of the temp_dac variable.

6.5. Temporal analysis

Since the comparison done in section 6.4 did not reveal any significant differences, it was decided to carry out a temporal analysis of the PSoC's behavior.

6.5.1. PWM frequency verification

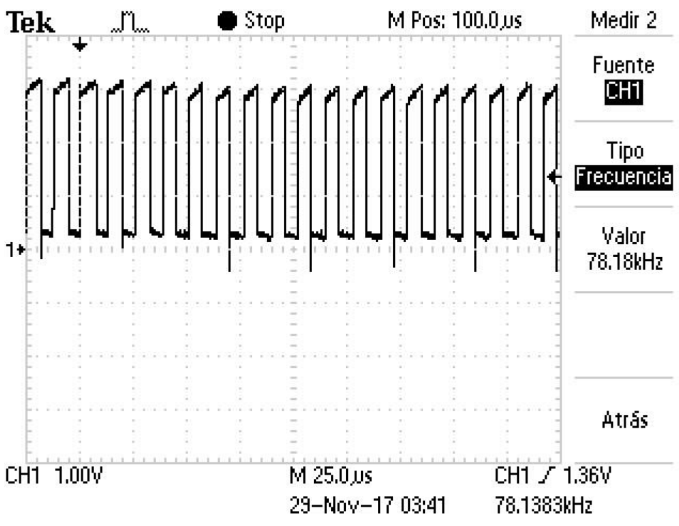


Figure 6.10: Oscilloscope capture of the PWM signal. Source: own.

The PWM signal was analyzed in the laboratory using an oscilloscope. As seen in Fig. 6.10, its frequency is correct: 78 kHz.

6.5.2. Interrupt frequency

Here one of the critical differences between the original and migrated programs was detected. Since the interrupt component in the PSoC had been linked to the TCPWM block and the "Interrupt on terminal count" option was selected, the interrupts were also happening at a 78 kHz frequency. However, in the original code the interrupt frequency was 10 kHz.

In order to fix this, a separate timer that controlled the time between interrupts had to be implemented.

6.5.3. Timer implementation

An example project provided by Cypress Semiconductor **[22]** was consulted as a starting point to implement the timer into the project. Once compiled, properly setup and programmed, this example had a LED blinking every second and changing color every 3 seconds.

After testing it, some modifications were made so that every second an interrupt happened and either switched the LED on or off (without changing color). This simplified example project would be easier to introduce into the main project.

First, the Timer Counter block was copied into the main project. Its period was changed to 4896 so that, once connected to the 48 MHz clock and the interrupt component, it produced interrupts with a frequency of 10 kHz. The resulting TopDesign.cysch file is shown in Fig. 6.11.

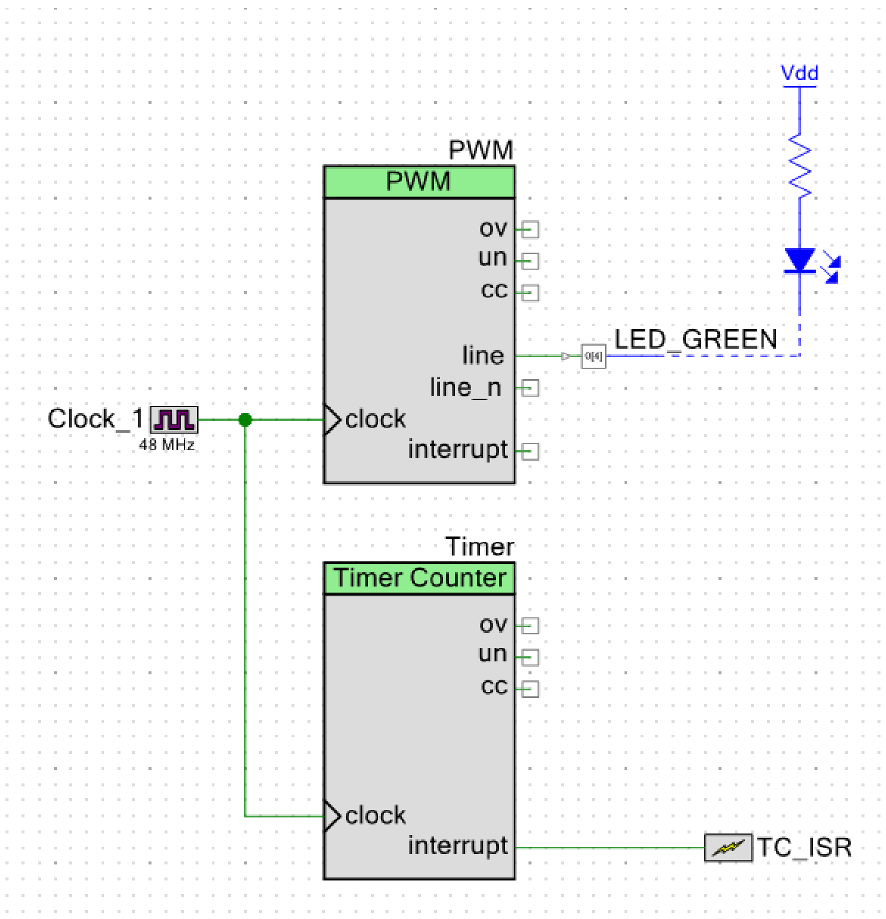


Figure 6.11: TopDesign.cysch file of the main migrated project after introducing the Timer Counter block. Source: Adaptation of [18] and [22].

Some modifications to the code in main.c were also necessary:

- The Timer block has to be started in main():

```
Timer_Start();
```

- The interrupt associated with the timer has to be cleared at the start of the interrupt handler function (replacing the PWM interrupt):

```
Timer_ClearInterrupt(Timer_INTR_MASK_TC);
```

The modified project was tested with positive results (see recording #2 of the annex): the note accuracy improved significantly, even though it was not yet correct, and so did the sound quality (see table 6.4). Still, the program did not sound as intended.

Clarity	/
Accuracy	++
Quality	++
Correct interaction	?

Table 6.4: Sound evaluation after changing the interrupt frequency.

6.5.4. Interrupt frequency verification

Just as with the PWM, an oscilloscope was used to measure the frequency of interrupts. In this case, however, an output pin had to be configured (see Fig. 6.12). It was initialized to 0 and was set to 1 (OutputPinSW_Write(1u)) at the start of the interrupt handler function and back to 0 (OutputPinSW_Write(0u)) at the end of it.

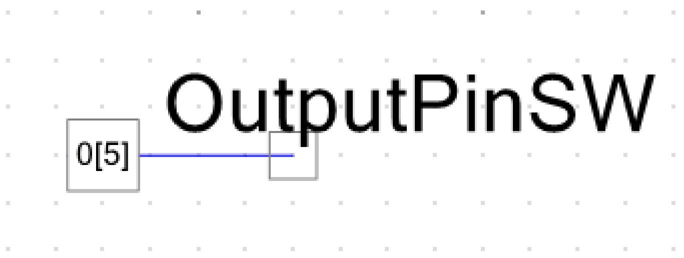


Figure 6.12: Software output pin in the TopDesign.cysch file. Source: Own.

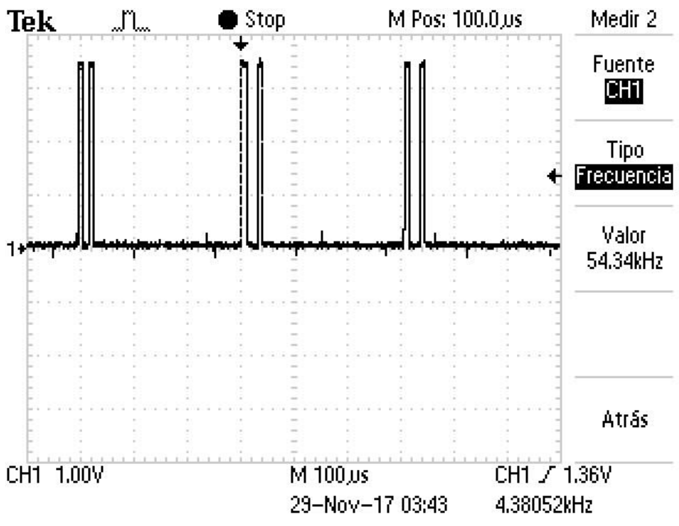


Figure 6.13: Oscilloscope capture of the interrupt signal. Source: own.

In Fig. 6.13 the high level represents the amount of time spent inside an interrupt. This result is quite surprising: Not only is the interrupt frequency wrong (54 kHz), but also there is a second, unexpected interrupt that happens almost immediately after the first.

After further testing, it was concluded that the OS was responsible for this, because, if it was disabled (by never starting the task scheduler), the interrupt behavior was perfectly correct (see Fig. 6.14).

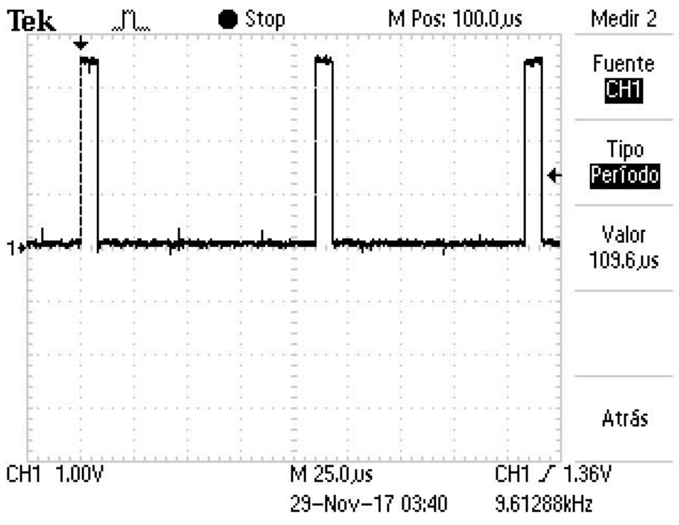


Figure 6.14: Oscilloscope capture of the interrupt signal while FreeRTOS was disabled. Source: own.

6.5.5. Tickless idle

After learning that the OS was somehow interfering with the interrupts, it seemed reasonable to examine the FreeRTOS configuration file carefully. Everything seemed to be in order, except for this line:

```
#define configUSE_TICKLESS_IDLE         2
```

The FreeRTOS reference manual **[23]** was consulted so as to learn more about it. Apparently, if the tickless idle is active (set to 1 or 2), and only the Idle task (the task that runs when no other tasks are able to) is running, then the microcontroller enters a low power state. This means that the tick interrupt, during which the OS can select a new task to enter the running state, is stopped. This could explain the weird behavior experienced before.

Testing (see table 6.5) revealed that the interaction between instruments was perfectly correct (see recording #3 of the annex): all of them were audible and synchronized. Sound quality also seemed to improve. As a downside, there was more saturation than before.

Clarity	-
Accuracy	+
Quality	++
Correct interaction	+++

Table 6.5: Sound evaluation after disabling the tickless idle.

The case of note accuracy bears special mention. The root of this issue lies in a macro called 'Hz' located in elochka.h:

```
#define      Hz   *64L*256/10000
```

This macro's purpose is to translate a frequency measured in Hertz into the number of points in the sinus.h tables that have to be skipped during each interrupt. In other words: the program is designed to be coherent with the note frequencies only when the interrupt frequency is 10 kHz. This is why the notes played are now correct.

6.6. Final corrections

6.6.1. PWM block variation

The saturation noted in the previous section became worrying, as there was no way of adjusting the temp_dac variable that completely solved it. Therefore, a different block that could also work as a PWM was tested. The TCPWM block that was being used previously was replaced by the PWM block **[9]** (see Fig. 6.15).

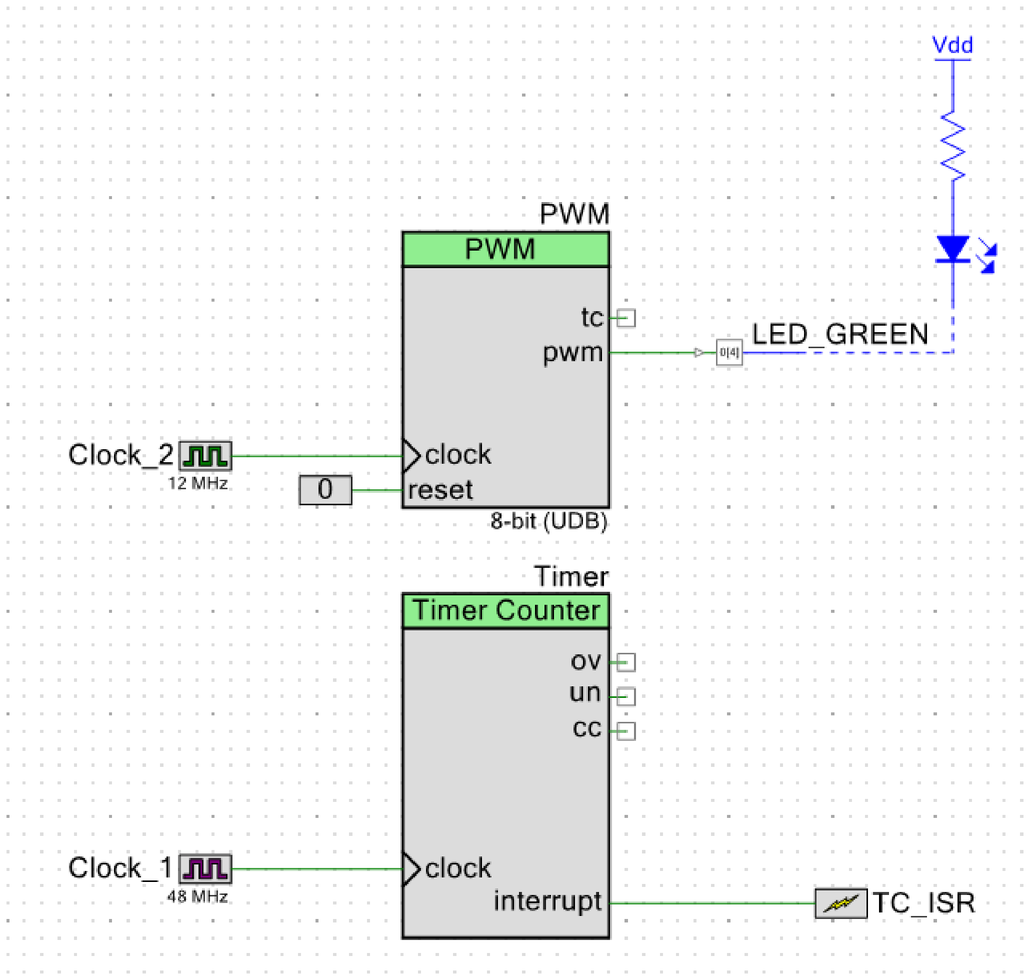


Figure 6.15: TopDesign.cysch file of the main migrated project after replacing the TCPWM block with a PWM block. Source: Adaptation of [18] and [22].

The main disadvantadge of this new block is that the maximum clock frequency it allows is 12 MHz. This means that the PWM resolution will be lower (around 7-8 bits) because the period will necessarily be lower too.

Clarity	+++
Accuracy	/
Quality	---
Correct interaction	/

Table 6.6: Sound evaluation after switching the PWM block.

After evaluating the resulting sound (see table 6.6), the saturation disappeared (see recording #4 of the annex). However, the quality got significantly worse.

6.6.2. Sound quality correction

Further analysis of the code revealed that the sinus.h tables were not being read correctly now. More specifically, the negative values were being converted into positive ones. To solve this, all of the tables in sinus.h, along with the temp1 variable, were changed into the integer type (in the original code they belonged to the character type).

Clarity	/
Accuracy	/
Quality	+++
Correct interaction	/

Table 6.7: Sound evaluation after changing the sinus.h tables into the integer type.

After verifying that the tables were now being read correctly, the sound was re-evaluated (see table 6.7). It seemed that, with the increase in quality achieved, the sound result was finally equivalent to that of the previous projects (see recording #5 of the annex). Therefore, the migration process had finished.

7. Quartet improvements

This chapter will be devoted to explaining the improvements made to the migrated Quartet program. Most of them are designed to improve sound quality, but some add extra functionality to the program as well.

7.1. 16-bit PWM

The most effective way to improve sound quality was to increase the PWM bit resolution. In order to do this, an existing project by Ganesh Raaja was consulted **[24]**. This project consisted in achieving a high resolution PWM by combining lower resolution PWM blocks (see Fig. 7.1).

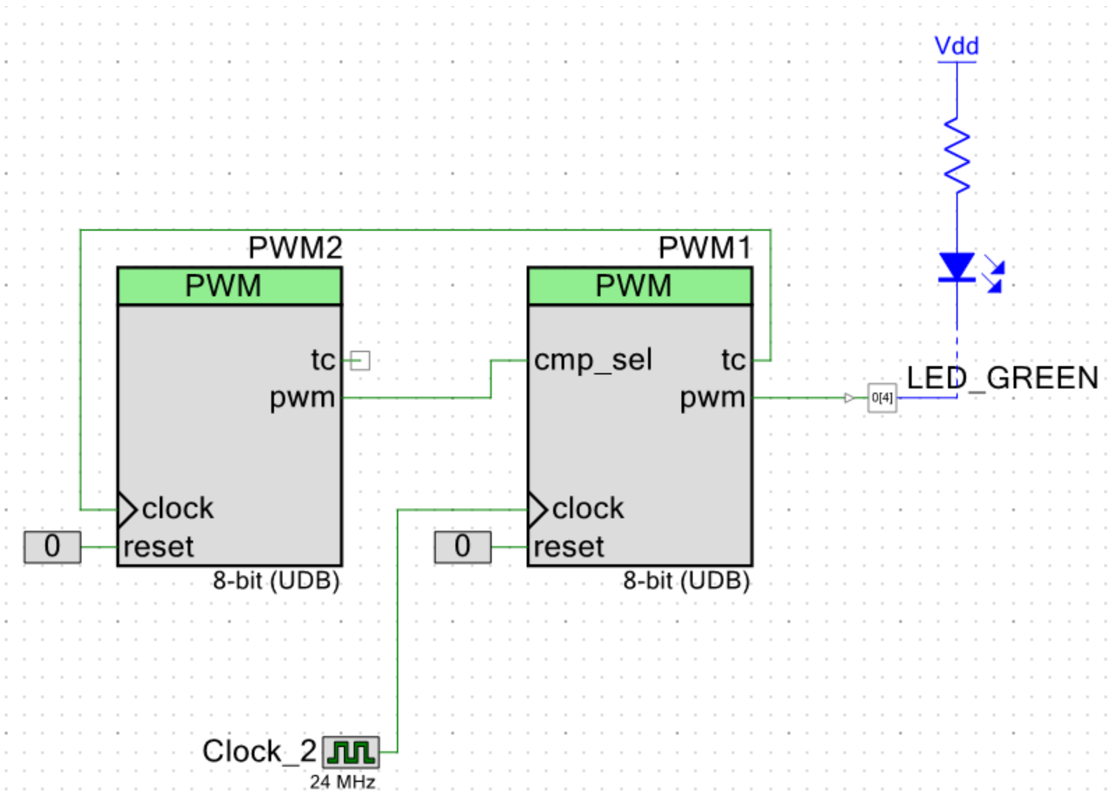
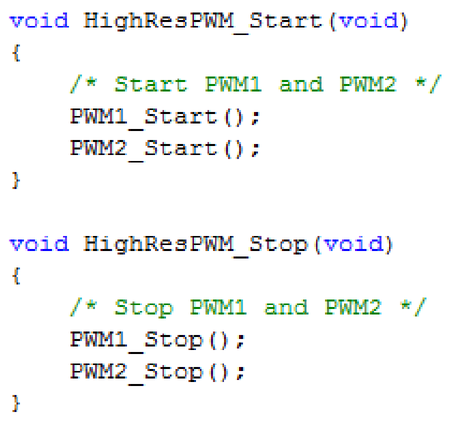


Figure 7.1: Implementation of the 16-bit PWM. Source: Adaptation of [24].

The PWM1 block has two 8-bit outputs with a difference in duty cycle of one clock. The output is selected by the hardware cmp_sel signal, which is the output signal of the PWM2 block. A 24 MHz clock is used for the PWM1 block (because it was the maximum frequency it allowed), while the PWM2 clock is the terminal count output from PWM1.

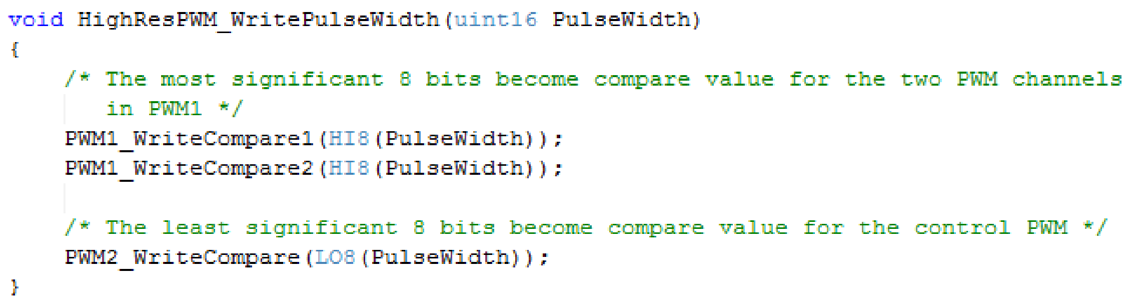
There are also several functions that need to be added to the code in main.c. The first two are the functions that start and stop the entire 16-bit PWM (see Fig. 7.2).



```
void HighResPWM_Start(void)
{
    /* Start PWM1 and PWM2 */
    PWM1_Start();
    PWM2_Start();
}

void HighResPWM_Stop(void)
{
    /* Stop PWM1 and PWM2 */
    PWM1_Stop();
    PWM2_Stop();
}
```

Figure 7.2: Functions that start (top) and stop (bottom) the 16-bit PWM. Source: [24].

It is also necessary to implement the function that writes the compare value (pulse width) to the 16-bit PWM (see Fig. 7.3).



```
void HighResPWM_WritePulseWidth(uint16 PulseWidth)
{
    /* The most significant 8 bits become compare value for the two PWM channels
       in PWM1 */
    PWM1_WriteCompare1(HI8(PulseWidth));
    PWM1_WriteCompare2(HI8(PulseWidth));

    /* The least significant 8 bits become compare value for the control PWM */
    PWM2_WriteCompare(LO8(PulseWidth));
}
```

Figure 7.3: Function that writes the compare value to the 16-bit PWM. Source: [24].

The sound that resulted from introducing the high resolution PWM improved greatly when compared to the previous version of the program (see recording #6 of the annex).

7.2. New amplitude envelopes

The next upgrade consisted in changing the amplitude envelopes along with the way they were generated. Instead of being created using simple operations during interrupts, they will be stored in ROM, having already been calculated previously. This will allow less linear envelopes that will give the sound a more natural evolution.

Fig. 7.4, 7.5 and 7.6 show the original and new amplitude envelopes for every instrument.

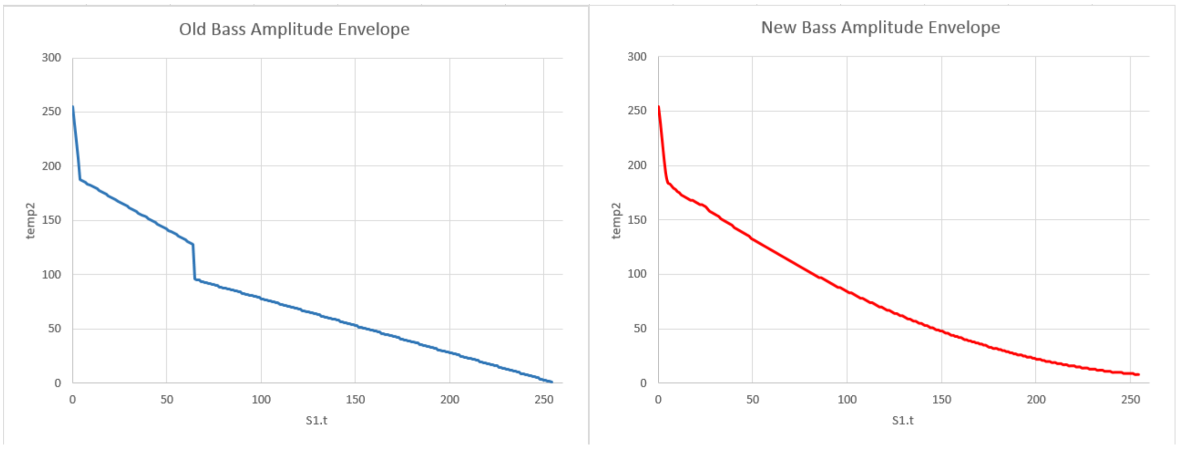


Figure 7.4: Comparison between the original and new amplitude envelopes of the bass. Elaborated from Victor Timofeev's code.

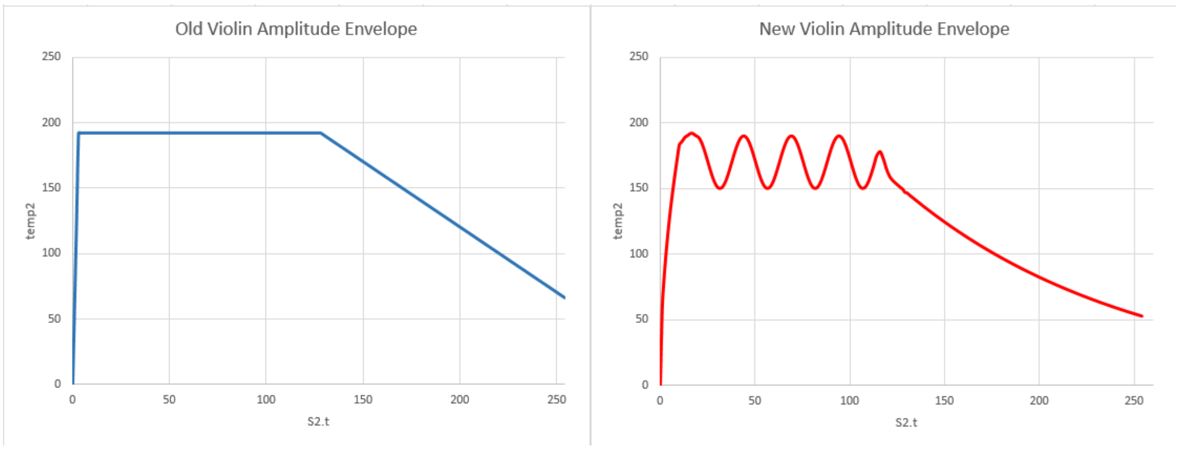


Figure 7.5: Comparison between the original and new amplitude envelopes of the violin. Elaborated from Victor Timofeev's code.

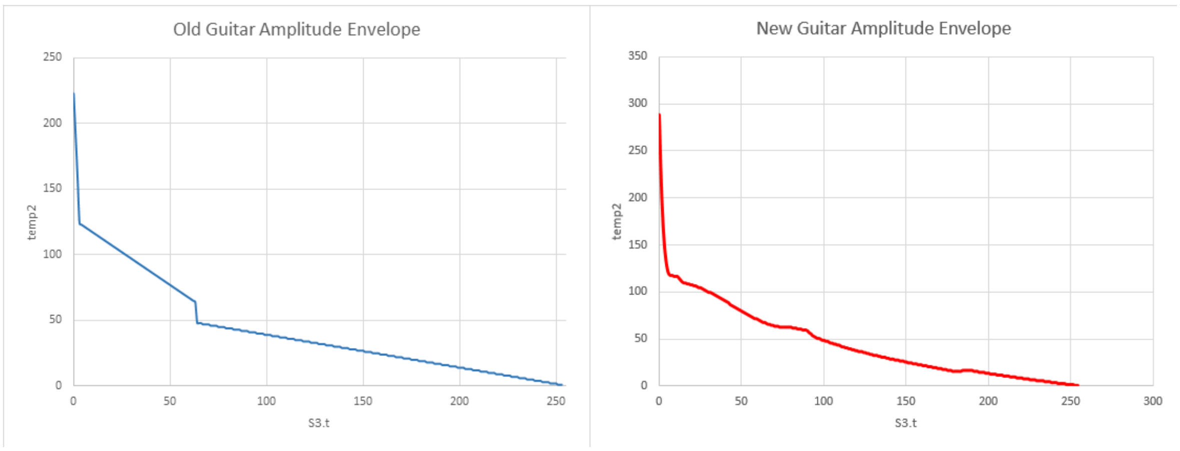
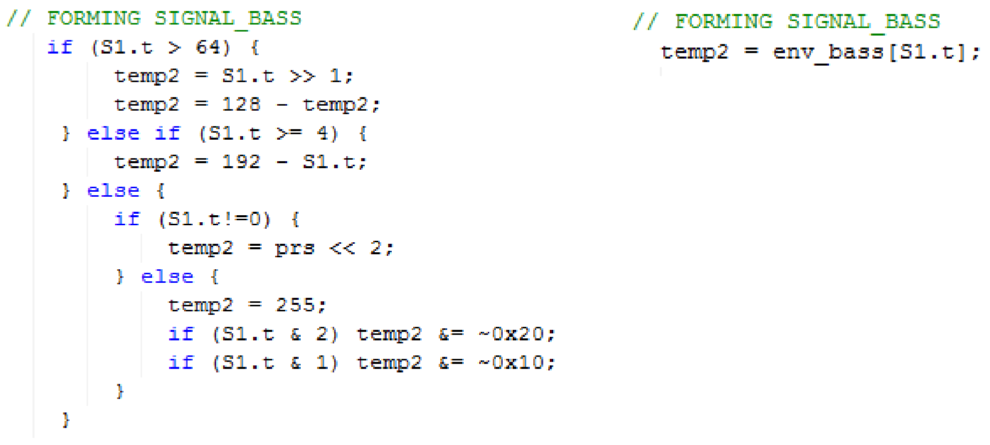


Figure 7.6: Comparison between the original and new amplitude envelopes of the guitar. Elaborated from Victor Timofeev's code.

These new envelopes were stored in arrays inside the sinus.h file. Lastly, the code which previously generated the envelope had to be replaced (see Fig. 7.7).



```
// FORMING SIGNAL_BASS
if (S1.t > 64) {
    temp2 = S1.t >> 1;
    temp2 = 128 - temp2;
} else if (S1.t >= 4) {
    temp2 = 192 - S1.t;
} else {
    if (S1.t!=0) {
        temp2 = prs << 2;
    } else {
        temp2 = 255;
        if (S1.t & 2) temp2 &= ~0x20;
        if (S1.t & 1) temp2 &= ~0x10;
    }
}
```

```
// FORMING SIGNAL_BASS
  temp2 = env_bass[S1.t];
```

Figure 7.7: Comparison between the original (left) and new (right) code inside the synthesizer that deals with the bass's envelope. Adapted from Victor Timofeev's code.

The change in sound was subtle, but when listening carefully it was verified that the instruments sounded more natural (see recording #7 of the annex). In fact, the effect can be observed in the actual recordings (see Fig. 7.8). The original waveforms in the audio recording were more linear and artificial, whereas the new ones present more variability.

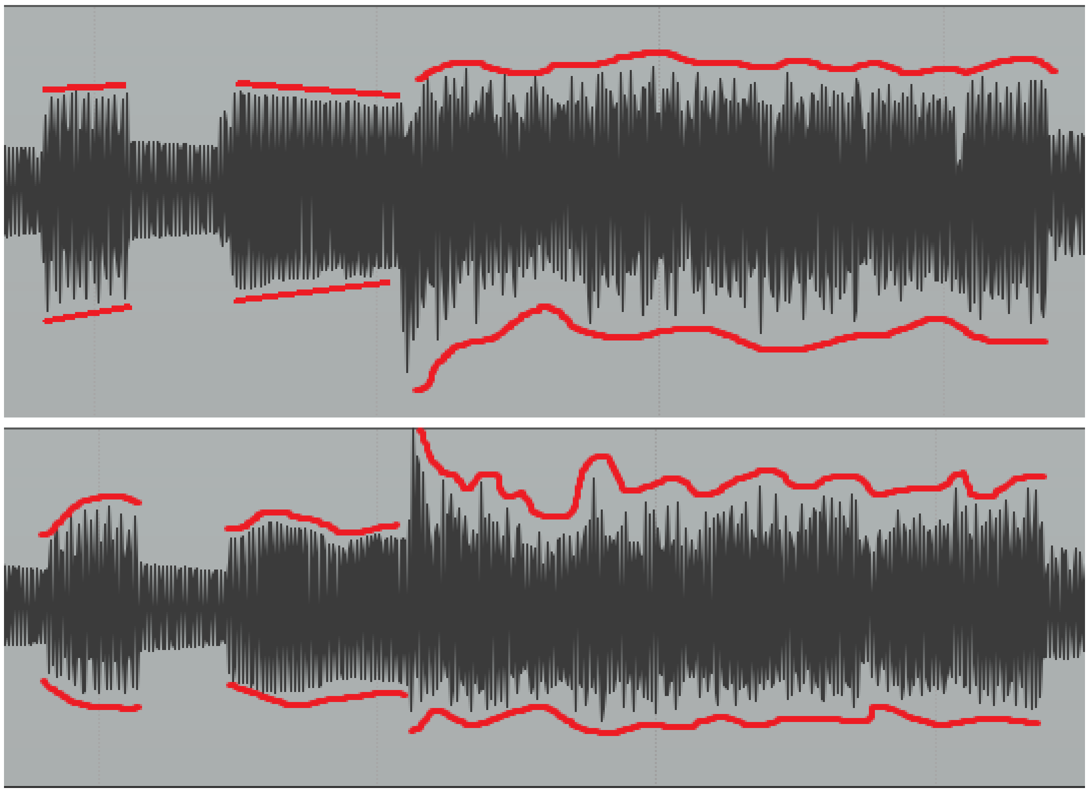


Figure 7.8: Comparison between the 16-bit PWM recording with the original envelopes (top) and with the new ones (bottom). Source: own.

7.3. Waveform tables with 128 data points

Another way of improving the sound quality of the program is to increase the number of data points that make up one period of the characteristic waveform of every instrument. More specifically, the original amount of 64 values will be doubled. The result will be more realistic sounding instruments.

7.3.1. Data interpolation

The first step in this process is to interpolate the original data points to create a continuous function. The original values were previously amplified (multiplied by 4) so that there was room to create intermediate points. This was done for different pieces of the entire set of points, as finding a function that described the entire waveform would be too complicated. Fig. 7.9, 7.10 and 7.11 compare the original and new waveforms for every instrument.

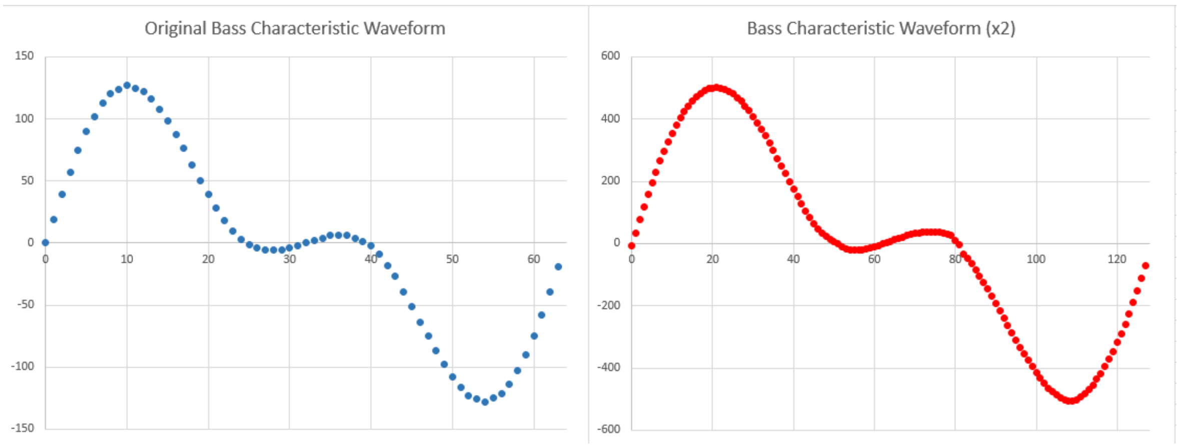


Figure 7.9: Comparison between the original bass waveform and the new one with double the amount of data points. Elaborated from Victor Timofeev's data in sinus.h.

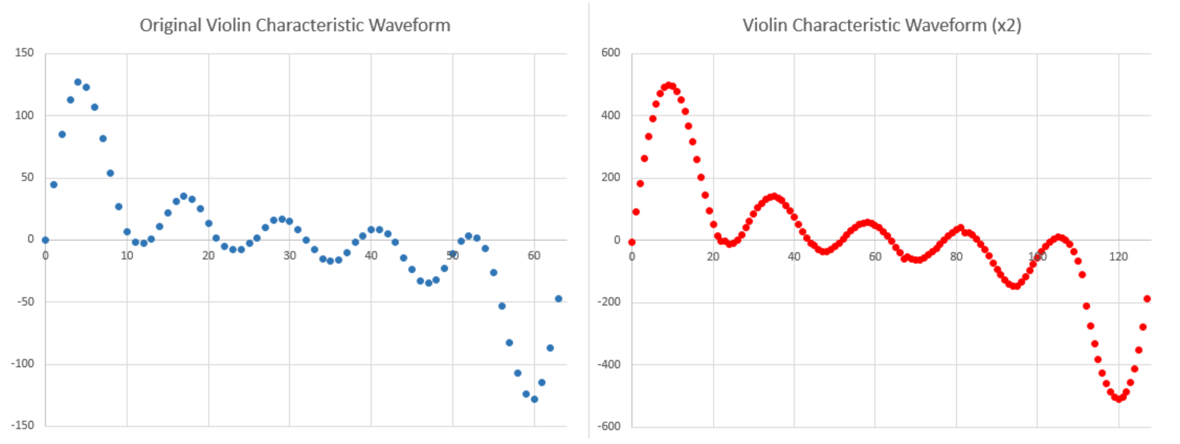


Figure 7.10: Comparison between the original violin waveform and the new one with double the amount of data points. Elaborated from Victor Timofeev's data in sinus.h.

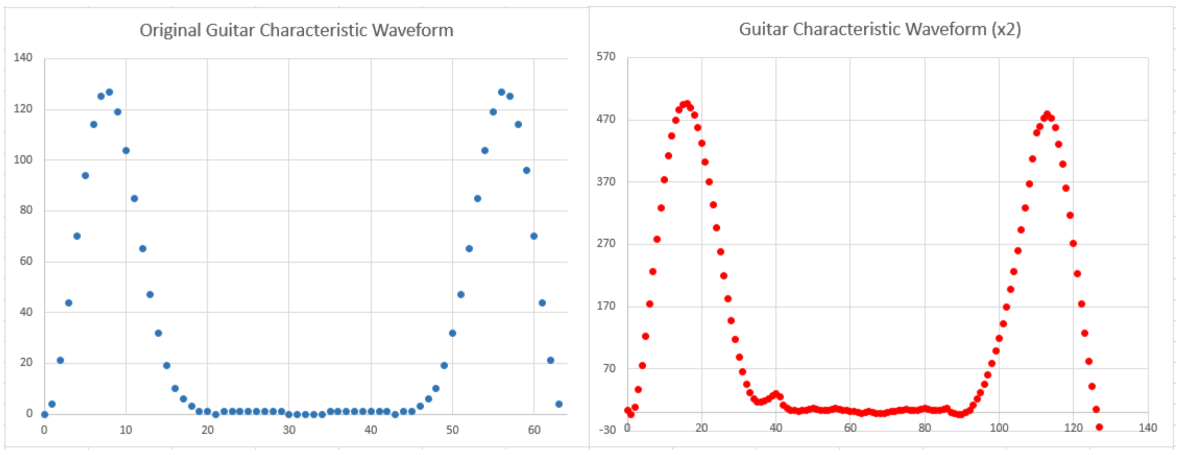


Figure 7.11: Comparison between the original guitar waveform and the new one with double the amount of data points. Elaborated from Victor Timofeev's data in sinus.h.

7.3.2. Data reading method

Before going into the modifications needed to implement the new sinus tables, it is essential to understand how and when the program reads every data point.

The data in the sinus tables represents one period of each instrument's waveform. This implies that, in order to play a note with a characteristic frequency of *F* [Hz], the entire waveform has to be read *F* times per second. If the waveform is comprised of 64 individual data points, a single point has to be read 64·*F* times per second.

However, since data points can only be read during interrupts, it is more practical to speak of interrupts instead of seconds. As seen in previous chapters, the interrupt frequency is 10 kHz: 10000 interrupts happen every second. Therefore, a single data point has to be read 64·*F*/10000 times per interrupt.

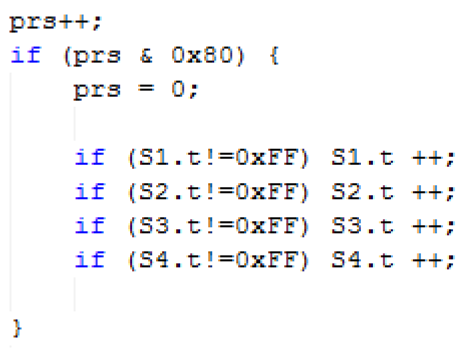
7.3.3. Program modifications

With 128 points, every single point will have to be read 128·*F*/10000 times per interrupt. However, if the interrupt frequency is kept constant, the program will simply read data points faster and consequently skip the interpolated ones, thus creating exactly the same sound as before. Therefore, the interrupt frequency must be doubled so that these extra points become useful.

The line that reads from the sinus table has to be changed (for every voice) so that the index resets at 127 instead of 63, ensuring that the entire waveform is read:

```
temp1 = bass[*((char*)&S1.f+1) & 0x7F];
```

The counter that controls when the next point of the envelope is selected must also be changed, otherwise the volume will decay too fast. Previously it counted up to 64 before reading the next point of the envelope; now it must count up to 128 (see Fig. 7.12):



```
prs++;
if (prs & 0x80) {
    prs = 0;

    if (S1.t!=0xFF) S1.t ++;
    if (S2.t!=0xFF) S2.t ++;
    if (S3.t!=0xFF) S3.t ++;
    if (S4.t!=0xFF) S4.t ++;

}
```

Figure 7.12: Fragment of code where the counter prs controls the increase of Sx.t variables, which in turn serve as an index for the temp2 tables. Adapted from Victor Timofeev's code.

Even though mathematically the "Hz" macro in elochka.h could have been kept as it was, conceptually it made more sense to change it:

```
#define      Hz   *128L*256/20000
```

After testing the code a notable improvement in sound was confirmed (see recording #8 of the annex).

7.4. Waveform tables with 256 data points

After verifying it was possible to create more accurate waveform tables, it was decided to double the amount of data points again. This will now require an interrupt frequency of 40 kHz. Since the rest of the procedure is identical, it will only be mentioned briefly.

7.4.1. Interpolated data

The resulting data can be found in Fig. 7.13, 7.14 and 7.15.

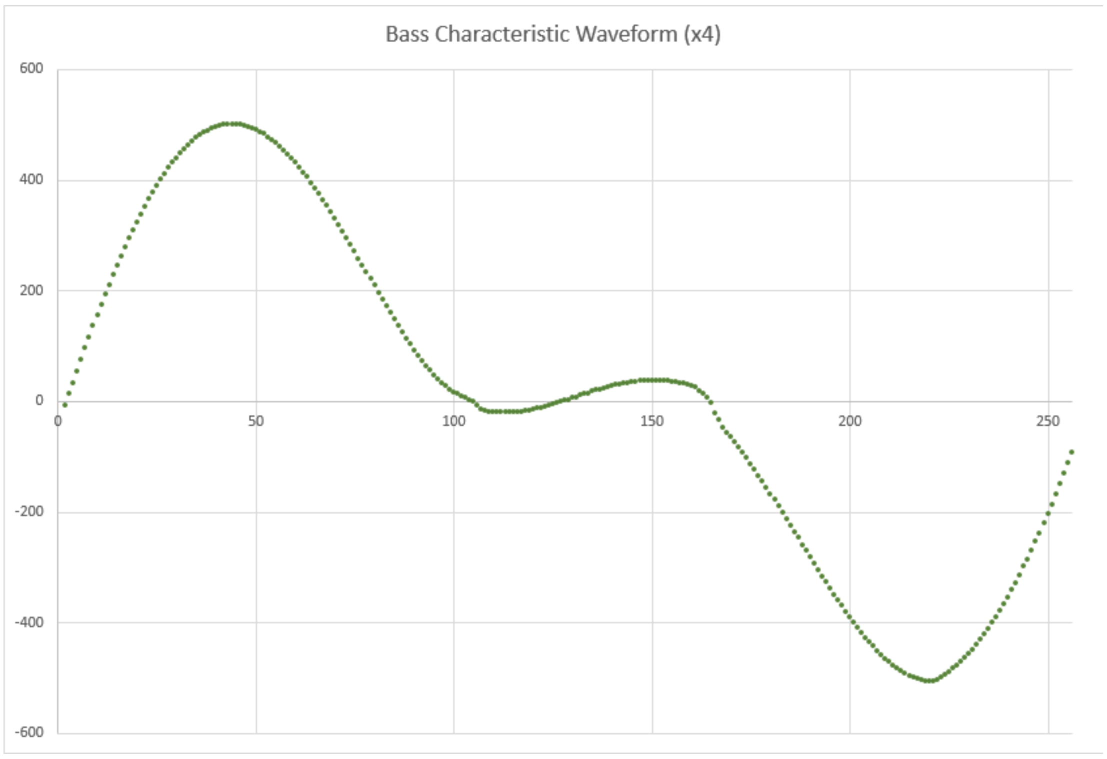


Figure 7.13: Bass characteristic waveform made up of 256 data points. Elaborated from Victor Timofeev's data in sinus.h.

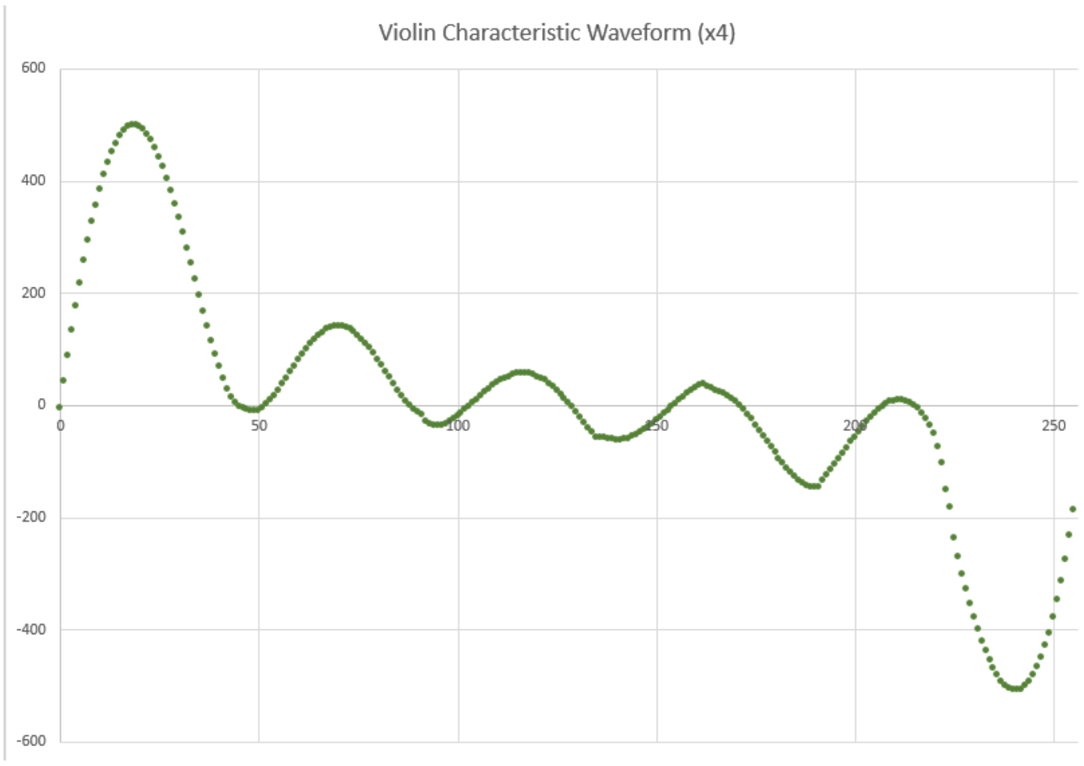


Figure 7.14: Violin characteristic waveform made up of 256 data points. Elaborated from Victor Timofeev's code.

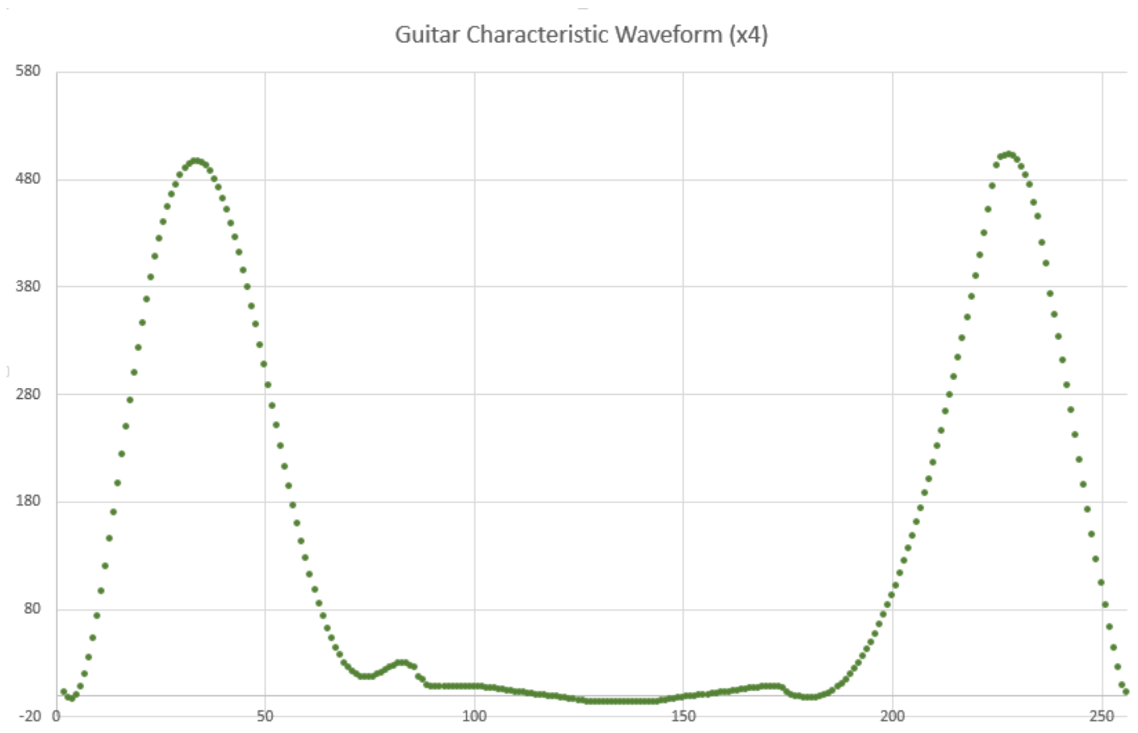


Figure 7.15: Guitar characteristic waveform made up of 256 data points. Elaborated from Victor Timofeev's code.

7.4.2. Code modifications

The modified lines (changed in the same manner as in section 7.3.3) are presented below:

```
temp1 = bass[*((char*)&S1.f+1) & 0xFF];
```
(this has to be done for every voice)

```
if (prs & 0x100)
```

```
#define     Hz   *256L*256/40000
```

The result was another significant improvement in sound (see recording #9 of the annex).

7.5. Waveform tables with 512 data points

The next step could be doubling again the number of data points, but it would require a significant modification of the table reading method. This is because the line that generates the index requires a conversion into the character type, which can only hold values up to 255. Therefore, unless the data-reading was re-structured, it would not be possible to read up to 512 data points because the index would never surpass 255.

It would also require an interrupt frequency of 80 kHz, which surpasses the PWM frequency initially desired (78 kHz). However, it is also true that, as it will be explained later, the PWM frequency at this point was actually higher. In the end, it was decided not to introduce this modification because it would be too troublesome for the expected improvement in sound.

7.6. Broadening the note frequency range

The play() instruction, which is used to play a certain note with a specific duration, stores these two sources of information in the same variable. It does so by assigning bits 5 and 6 to the duration and bits 0-4 to the note. Since notes are represented with its index from the frequency table (C_2, the lowest available note, is represented by 0; $C\#_2$ is represented by 1 and so on), any note with an index higher than 31 (0001 1111 in binary) will start mixing up both sources of information. In short, the higher notes inside the frequency table cannot actually be played.

7.6.1. Duration modifications

In order to solve this, the duration information was assigned to bits 6 and 7 instead. This implied a modification of the play() instruction (in elochka.h):

```
#define play(note,duration)     (((duration-1) << 6) | note)
```

It was also necessary to change the line that read the frequency table so that it included the extra bit assigned to note information (in main.c):

```
f = Freq[S->cBaseNote + (cmd & 0x3F)];
```

Finally, the line that recovered the duration information had to move it an extra bit (in main.c):

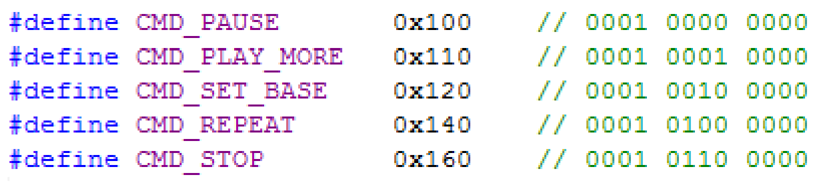
```
n = cmd >> 5;
```

7.6.2. Change into the integer type

Because of the extra bit, which surpasses the character type maximum capacity (255), all variables that store commands have to be changed into the integer type. This includes all of the notelists, as well as the variables “cmd” and “n”.

7.6.3. Command re-structure

Previously, bit 7 was used to distinguish commands (stop, repeat, etc) from notes. Now this information had to be moved to the next bit, bit 8. Doing this meant re-defining all of the commands (see Fig. 7.16).



```
#define CMD_PAUSE       0x100   // 0001 0000 0000
#define CMD_PLAY_MORE   0x110   // 0001 0001 0000
#define CMD_SET_BASE    0x120   // 0001 0010 0000
#define CMD_REPEAT      0x140   // 0001 0100 0000
#define CMD_STOP        0x160   // 0001 0110 0000
```

Figure 7.16: Definition of all commands after changing the discriminating bit from bit 7 to 8. Adapted from Victor Timofeev's code.

7.7. Adding an extra octave

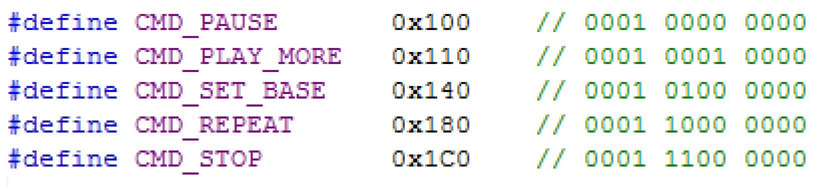
Another modification made was adding an extra lower octave. The procedure to do this consisted in:

1. Adding the notes belonging to this new octave at the start of the frequency table.

2. Since now some instruments will have to go 3 octaves up, and originally the program was designed to go up a maximum of 2 octaves, more bits have to be saved from the setbase() command:

   ```
   S->cBaseNote = (char)(cmd & 0x3F);
   ```

3. Commands have to be re-structured again (see Fig. 7.17):



```
#define CMD_PAUSE       0x100   // 0001 0000 0000
#define CMD_PLAY_MORE   0x110   // 0001 0001 0000
#define CMD_SET_BASE    0x140   // 0001 0100 0000
#define CMD_REPEAT      0x180   // 0001 1000 0000
#define CMD_STOP        0x1C0   // 0001 1100 0000
```

Figure 7.17: Definition of all commands after adding the extra octave. Adapted from Victor Timofeev's code.

7.8. Adding an extra voice

The last improvement made to the Quartet program consisted in adding another voice. The idea was to introduce another violin that played an octave lower than the original one, thus creating a harmonized effect between both (see recording #10 of the annex).

7.8.1. Notelist and sound channel

The first step was to copy the original violin notelist, rename it and set the octave to 2 instead of 3. Also, the switch for this new voice was created:

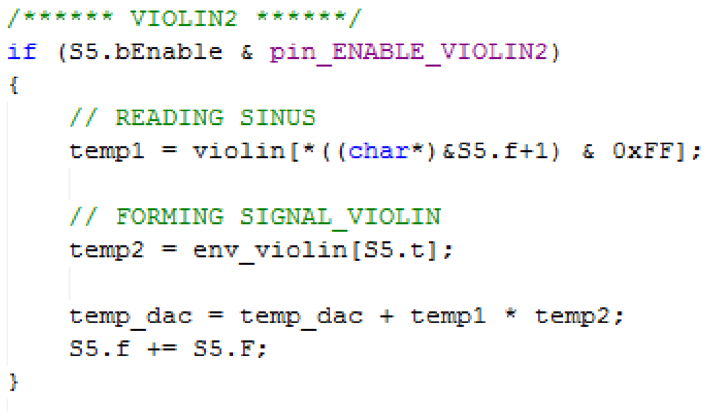
```
#define pin_ENABLE_VIOLIN2  1       // Switch violin2 channel ON/OFF
```

Furthermore, a new sound channel had to be defined:

```
TSound S5;  // For channel 5 (violin2)
```

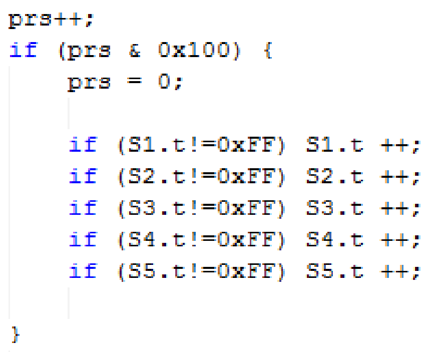
7.8.2. Synthesizer modifications

Additional code had to be introduced into the interrupt handler function so that the new voice was correctly synthesized (see Fig. 7.18 and 7.19).



```
/****** VIOLIN2 ******/
if (S5.bEnable & pin_ENABLE_VIOLIN2)
{
    // READING SINUS
    temp1 = violin[*((char*)&S5.f+1) & 0xFF];

    // FORMING SIGNAL_VIOLIN
    temp2 = env_violin[S5.t];

    temp_dac = temp_dac + temp1 * temp2;
    S5.f += S5.F;
}
```

Figure 7.18: Fragment of code that synthesizes the sound sample of violin2. Adapted from Victor Timofeev's code.



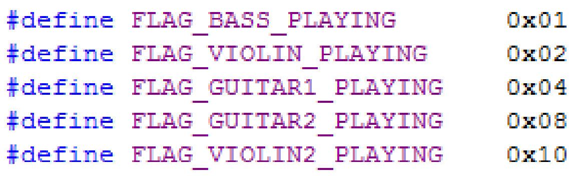
```
prs++;
if (prs & 0x100) {
    prs = 0;

    if (S1.t!=0xFF) S1.t ++;
    if (S2.t!=0xFF) S2.t ++;
    if (S3.t!=0xFF) S3.t ++;
    if (S4.t!=0xFF) S4.t ++;
    if (S5.t!=0xFF) S5.t ++;

}
```

Figure 7.19: Fragment of code that controls the evolution of the indexes of the envelope tables. Adapted from Victor Timofeev's code.

7.8.3. FreeRTOS related modifications

The next steps, which were more related to the OS, consisted in:

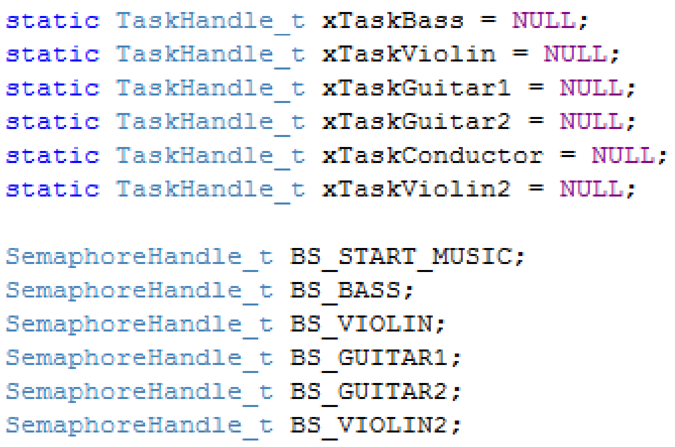
1. Creating a flag that indicates that violin2 is playing (see Fig. 7.20). It was assigned to the next available bit (0001 0000).



```
#define FLAG_BASS_PLAYING       0x01
#define FLAG_VIOLIN_PLAYING     0x02
#define FLAG_GUITAR1_PLAYING    0x04
#define FLAG_GUITAR2_PLAYING    0x08
#define FLAG_VIOLIN2_PLAYING    0x10
```

Figure 7.20: Definition of flags for all 5 voices. Adapted from Pau Mendieta's code.

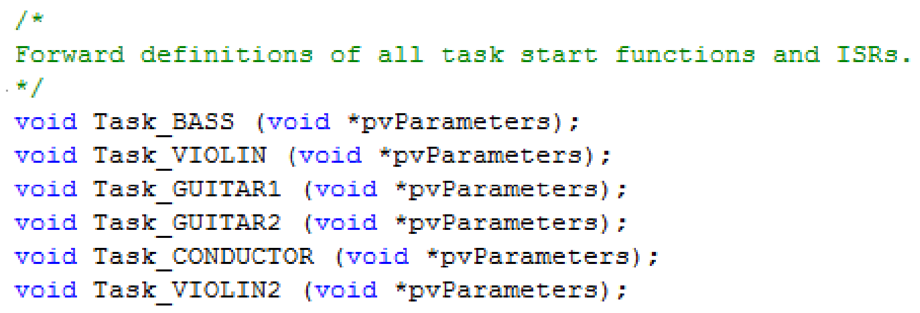
2. Creating a task handle and a binary semaphore for the new violin2 task (see Fig. 7.21).



```
static TaskHandle_t xTaskBass = NULL;
static TaskHandle_t xTaskViolin = NULL;
static TaskHandle_t xTaskGuitar1 = NULL;
static TaskHandle_t xTaskGuitar2 = NULL;
static TaskHandle_t xTaskConductor = NULL;
static TaskHandle_t xTaskViolin2 = NULL;

SemaphoreHandle_t BS_START_MUSIC;
SemaphoreHandle_t BS_BASS;
SemaphoreHandle_t BS_VIOLIN;
SemaphoreHandle_t BS_GUITAR1;
SemaphoreHandle_t BS_GUITAR2;
SemaphoreHandle_t BS_VIOLIN2;
```

Figure 7.21: Definition of all task handles and binary semaphores for all 5 voices. Adapted from Pau Mendieta's code.

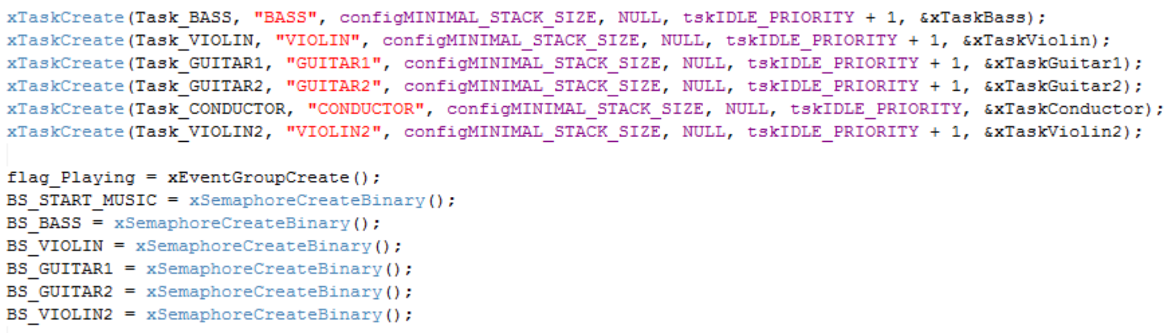
3. Defining the new task in the Task_Defs.h file (see Fig. 7.22).



```
/*
Forward definitions of all task start functions and ISRs.
*/
void Task_BASS (void *pvParameters);
void Task_VIOLIN (void *pvParameters);
void Task_GUITAR1 (void *pvParameters);
void Task_GUITAR2 (void *pvParameters);
void Task_CONDUCTOR (void *pvParameters);
void Task_VIOLIN2 (void *pvParameters);
```

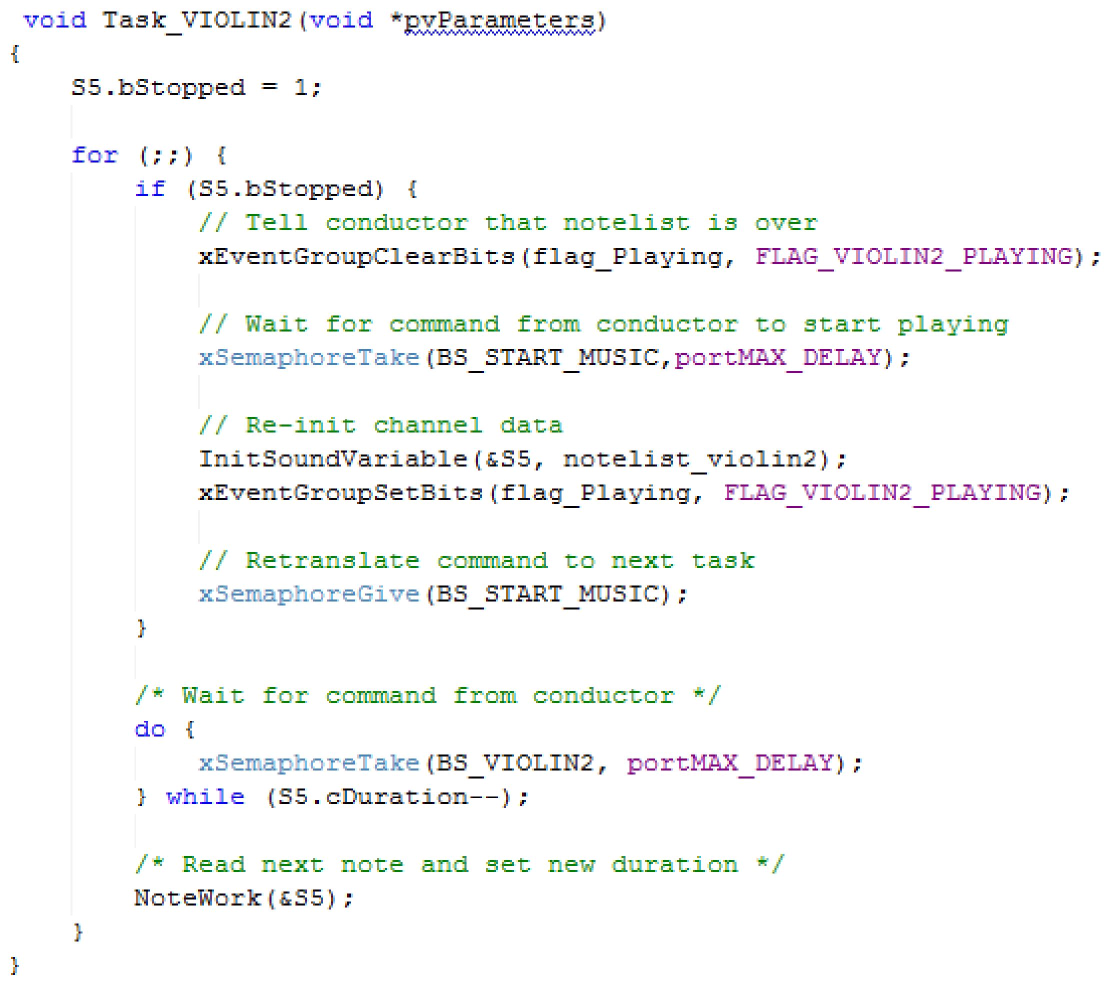
Figure 7.22: Definition of all tasks in Task_Defs.h. Adapted from Pau Mendieta's code.

4. Creating the new task and binary semaphore in the main() function (see Fig. 7.23). The code for the new task can be seen in Fig. 7.24.



```
xTaskCreate(Task_BASS, "BASS", configMINIMAL_STACK_SIZE, NULL, tskIDLE_PRIORITY + 1, &xTaskBass);
xTaskCreate(Task_VIOLIN, "VIOLIN", configMINIMAL_STACK_SIZE, NULL, tskIDLE_PRIORITY + 1, &xTaskViolin);
xTaskCreate(Task_GUITAR1, "GUITAR1", configMINIMAL_STACK_SIZE, NULL, tskIDLE_PRIORITY + 1, &xTaskGuitar1);
xTaskCreate(Task_GUITAR2, "GUITAR2", configMINIMAL_STACK_SIZE, NULL, tskIDLE_PRIORITY + 1, &xTaskGuitar2);
xTaskCreate(Task_CONDUCTOR, "CONDUCTOR", configMINIMAL_STACK_SIZE, NULL, tskIDLE_PRIORITY, &xTaskConductor);
xTaskCreate(Task_VIOLIN2, "VIOLIN2", configMINIMAL_STACK_SIZE, NULL, tskIDLE_PRIORITY + 1, &xTaskViolin2);

flag_Playing = xEventGroupCreate();
BS_START_MUSIC = xSemaphoreCreateBinary();
BS_BASS = xSemaphoreCreateBinary();
BS_VIOLIN = xSemaphoreCreateBinary();
BS_GUITAR1 = xSemaphoreCreateBinary();
BS_GUITAR2 = xSemaphoreCreateBinary();
BS_VIOLIN2 = xSemaphoreCreateBinary();
```

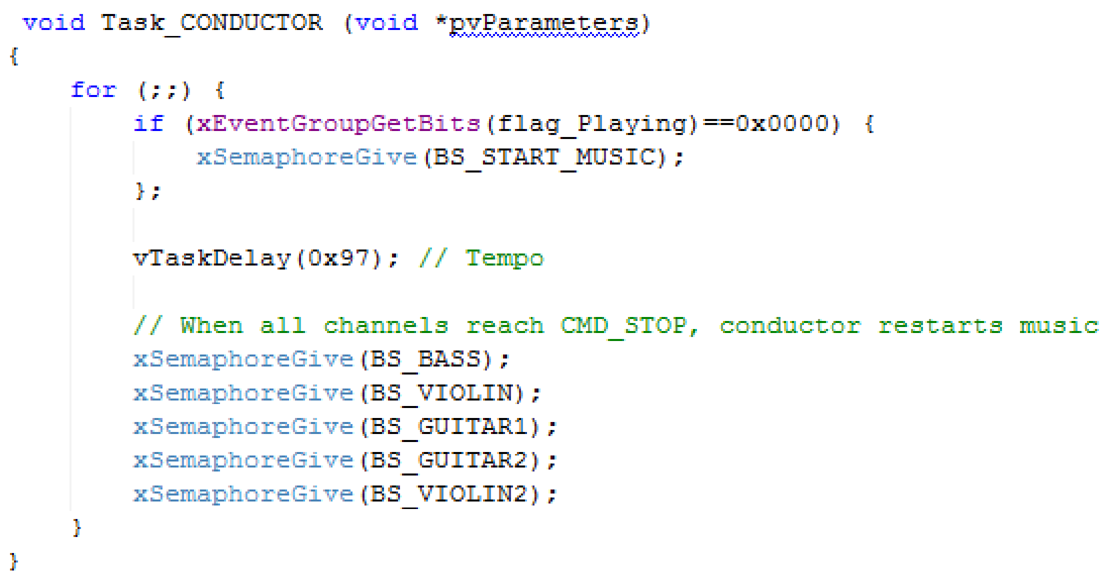
Figure 7.23: Creation of all tasks and binary semaphores. Adapted from Pau Mendieta's code.



```
 void Task_VIOLIN2(void *pvParameters)
{
    S5.bStopped = 1;

    for (;;) {
        if (S5.bStopped) {
            // Tell conductor that notelist is over
            xEventGroupClearBits(flag_Playing, FLAG_VIOLIN2_PLAYING);

            // Wait for command from conductor to start playing
            xSemaphoreTake(BS_START_MUSIC,portMAX_DELAY);

            // Re-init channel data
            InitSoundVariable(&S5, notelist_violin2);
            xEventGroupSetBits(flag_Playing, FLAG_VIOLIN2_PLAYING);

            // Retranslate command to next task
            xSemaphoreGive(BS_START_MUSIC);
        }

        /* Wait for command from conductor */
        do {
            xSemaphoreTake(BS_VIOLIN2, portMAX_DELAY);
        } while (S5.cDuration--);

        /* Read next note and set new duration */
        NoteWork(&S5);
    }
}
```

Figure 7.24: Violin2 task. Adapted from Pau Mendieta's code.

5. Editing the conductor task so that it gives the semaphore to the violin2 task as well (see Fig. 7.25).



```
 void Task_CONDUCTOR (void *pvParameters)
{
    for (;;) {
        if (xEventGroupGetBits(flag_Playing)==0x0000) {
            xSemaphoreGive(BS_START_MUSIC);
        };

        vTaskDelay(0x97); // Tempo

        // When all channels reach CMD_STOP, conductor restarts music
        xSemaphoreGive(BS_BASS);
        xSemaphoreGive(BS_VIOLIN);
        xSemaphoreGive(BS_GUITAR1);
        xSemaphoreGive(BS_GUITAR2);
        xSemaphoreGive(BS_VIOLIN2);
    }
}
```

Figure 7.25: Conductor task after adding the extra voice. Adapted from Pau Mendieta's code.

6. Re-adjusting the temp_dac variable.

8. Final Quartet Analysis

A detailed analysis of the final improved Quartet program, with and without the extra voice, will be presented in this chapter.

8.1. Resource usage (4 voices)

SRAM	8.984 KB (54.8%)
Flash	16.720 KB (12.8%)
Stack	1280 bytes
Heap	1024 bytes
UDB	50%

Table 8.1: Resources used by the final improved 4-voice Quartet program.

Table 8.1 shows the amount of PSoC's resources consumed by the final program. The SRAM and flash values show that there is enough space to add more instruments or improvements. UDB, which measures the amount of digital blocks used, shows that additional blocks could be included.

8.2. Temporal analysis (4 voices)

8.2.1. PWM frequency

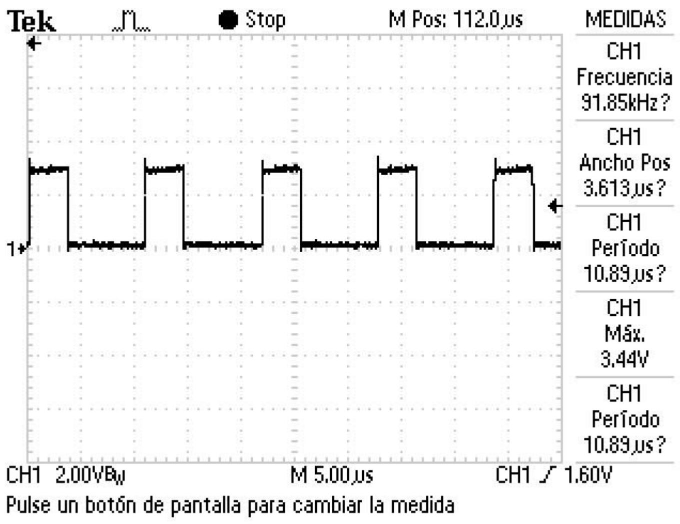


Figure 8.1: Oscilloscope capture of the final PWM signal. Source: own.

The PWM frequency, as shown in Fig. 8.1, has increased to 91.85 kHz because of the introduction of the high resolution PWM. Since it is higher than the original (78 kHz), there is no problem with it.

8.2.2. Interrupt frequency

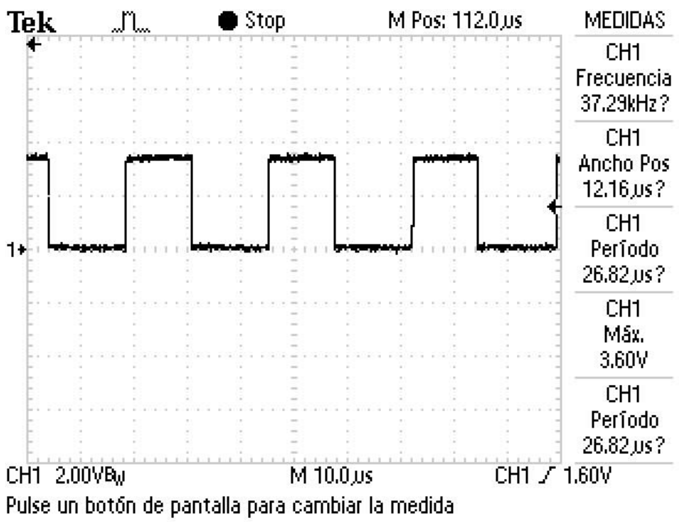


Figure 8.2: Oscilloscope capture of the final interrupt signal. Source: own.

The interrupt frequency is approximately 40 kHz (see Fig. 8.2), as was expected. The time spent in high level is the duration of an interrupt. The rest of it is available for the OS to carry out its own operations, which in this case is more than enough.

8.2.3. Tempo

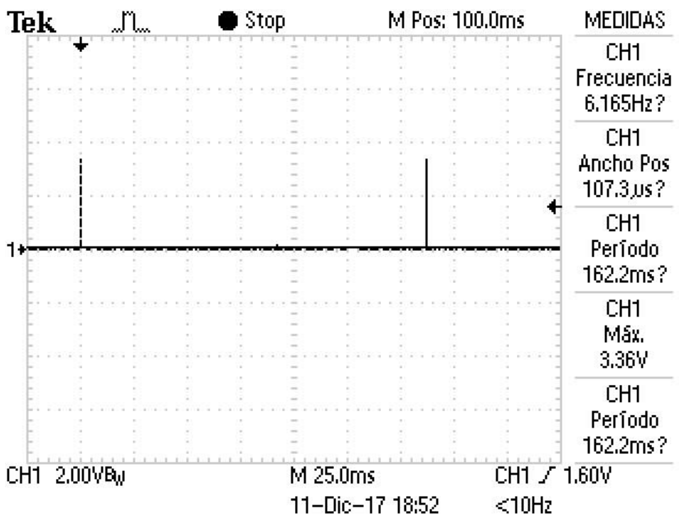


Figure 8.3: Oscilloscope capture that shows the time between two consecutive eighth notes. Source: own.

The peaks shown in Fig. 8.3 represent the processing of two consecutive eighth notes. Therefore, using the time between these two notes (162.2 ms), the tempo of the song can be calculated:

$$\frac{1}{0.1622 \cdot 2} \cdot 60 \cong 185\ BPM$$

8.2.4. Computational effort of synthesizing a sound sample

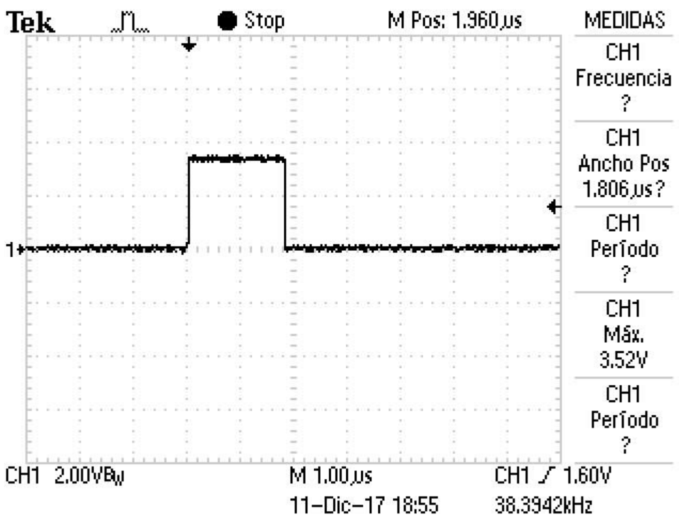


Figure 8.4: Oscilloscope capture that shows the amount of time required to synthesize a sound sample of a single voice. Source: own.

Fig. 8.4 shows that it takes the processor 1.8 µs to synthesize one sound sample of the bass. Assuming it takes more or less the same to synthesize every voice, the processor would need around 7.2 µs to generate the sample for all 4 voices, which accounts for 60% of the total interrupt time (see Fig. 8.2 for the total interrupt time).

8.2.5. Computational effort of adjusting temp_dac

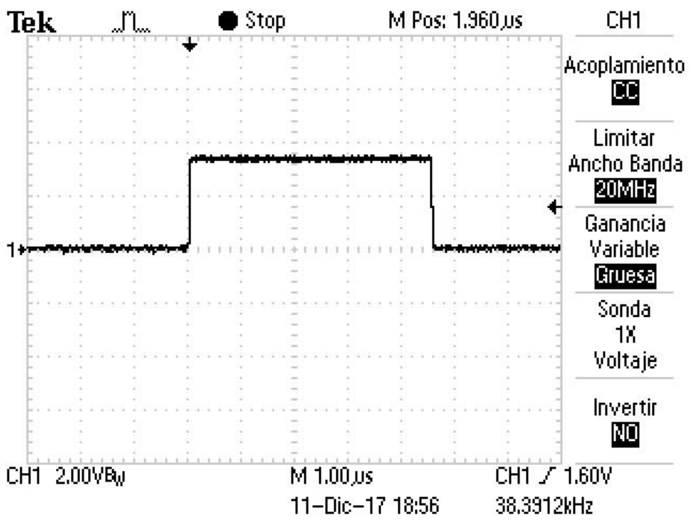


Figure 8.5: Oscilloscope capture that shows the amount of time required to adjust the temp_dac variable. Source: own.

It is interesting to study the amount of time required to adjust the temp_dac variable because in the final program this is done using a mathematical division. This was not possible in the PIC microcontrollers due to the high computational cost of the operation, so instead it was done using the binary right shift operator (>>), which is equivalent to a division by a power of 2. The mathematical division, however, gives more freedom when adjusting temp_dac, and thus allows most of the PWM resolution to be utilized.

Fig 8.5 reveals that the time spent making this adjustment is around 4.5 µs, 37% of the total interrupt time. Therefore, it clearly is a costly operation, even though in this case this is not a problem due to how much time is available (interrupts take only about 50% of the available time, as seen in Fig 8.2).

8.2.6. Command processing

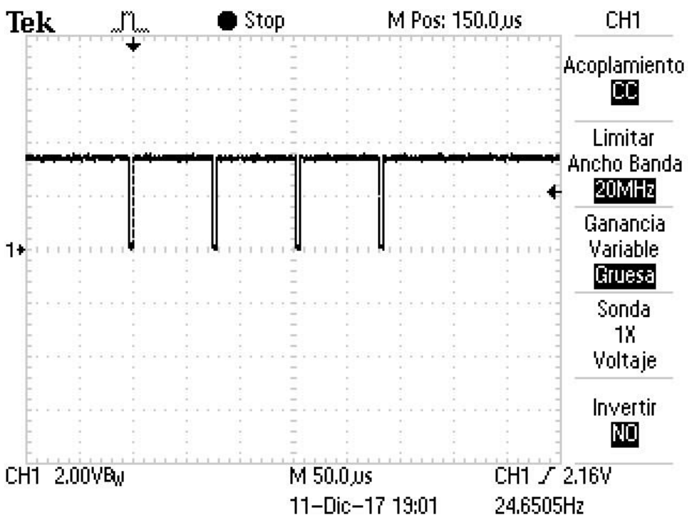


Figure 8.6: Oscilloscope capture showing the amount of time during which the instruments are waiting for a command (high voltage). Source: own.

The time spent in low voltage in Fig 8.6 represents the moment when the instruments are receiving a command from the conductor. The rest of it is spent waiting for the command. This shows how fast commands are processed.

8.3. Resource usage (5 voices)

SRAM	9.008 KB (55.0%)
Flash	17.624 KB (13.4%)
Stack	1280 bytes
Heap	1024 bytes
UDB	50%

Table 8.2: Resources used by the final improved 5-voice Quartet program.

As it can be seen in table 8.2, the addition of an extra voice barely has an impact on the resources consumed by the program.

8.4. Temporal analysis (5 voices)

8.4.1. Interrupt duration

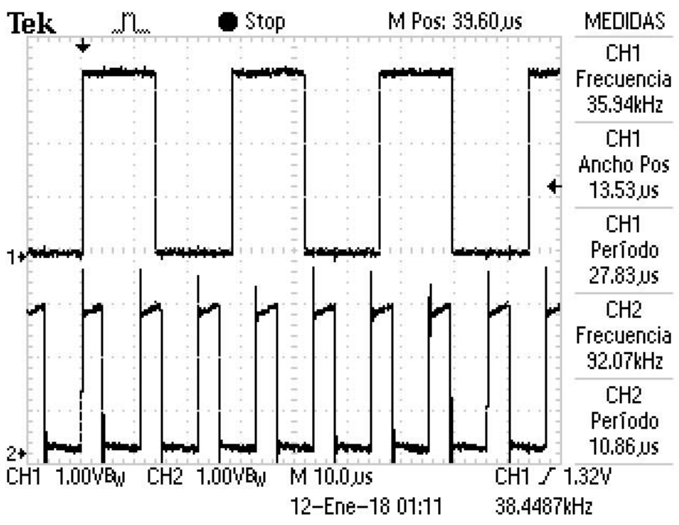


Figure 8.7: Oscilloscope capture showing the interrupt signal (top, CH1) and the PWM signal (down, CH2). Source: own.

Fig. 8.7 shows that an entire interrupt now lasts 13.53 µs, which is 1.37 µs more than the 4-voice version. There is still enough time for the OS to carry out its tasks.

9. Future improvements

In this chapter, suggestions for future improvements of the Quartet project will be presented.

9.1. Bluetooth

One of the important features of the PSoC that has not been used in this project is its capacity to communicate with external devices through Bluetooth. Such feature could be used to send music sheets (in the form of a header file) from a smartphone to the PSoC directly, for example.

9.2. Utilizing hardware blocks

As mentioned in chapter 3, one of the objectives of this project that could not be fulfilled consisted in relieving the CPU by using the configurable hardware blocks to carry out some of the necessary operations of the Quartet code. This would require a PSoC with the following blocks:

- **Waveform generators [25]**: These would allow the characteristic waveforms of the instruments to be generated with no CPU cost and without occupying ROM storage space.

- **Modulation block [26]**: This block would carry out the multiplication between the characteristic waveform and the amplitude envelope and would result in a significant decrease in CPU usage. Another option is to use an analog multiplier **[27]**.

- **High resolution DACs [28]**: These would replace the PWM blocks and would not require an RC filter. The new analog output signal would thus be more pure.

9.3. Adding new instruments

In this project, a new voice has been added to the Quartet program. This could be taken a step further, by adding a completely new instrument. However, it would require a certain degree of musical knowledge (or experimentation with real instruments) in order to be able to define its characteristic waveform and envelope.

9.4. Multiple sound outputs

Another possible improvement would be to have separate sound outputs for every voice, so that each of them could be heard on separate speakers. This would help create a more realistic orchestral sensation, as well as allowing a higher bit resolution for every voice (their samples would no longer have to be added together and divided).

Also related to this is the introduction of a stereo output, which would allow each speaker to play different audio. This would allow voices to be panned either to the right or left, making them easier to distinguish from one another.

9.5. Widening the range of note durations

Currently, the Quartet only offers 4 options for the note duration (1, 2, 3 and 4), where 1 is an eight note and 4 is a half note. It is possible to create longer notes by using the playmore() instruction, but shorter ones cannot be played.

Therefore, another improvement could be re-structuring the play() instruction so that it allows a wider range of note durations, especially so that faster notes could be played.

10. Planning

July	
W30	Prior works consulting
August	
W32	C language learning
	PSoC Creator 4.1 installation and testing
September	
W35	C language learning
W36	Original Quartet code examination
W37-W38	FreeRTOS manual consulting
W39	FreeRTOS migration
October	
W40	Quartet migration
W41	Quartet troubleshooting
W42-W43	PSoC 4 vs PIC18 comparison
November	
W44	Temporal analysis
W45	Final migration corrections
	Note frequency range broadening
W47	Introduction of the 16-bit PWM
December	
W48	New amplitude envelopes
W49	Waveform tables with 128 and 256 data points
W50	Addition of the extra voice and extra octave

Table 10.1: Project planning

11. Budget

Concept	Units	Unit price [€/unit]	Total price [€]
CY8CKIT-042-BLE PSoC 4 kit	1	40.61	40.61
PSoC Creator 4.1	-	0.00	0.00
MPLAB IDE	-	0.00	0.00
PuTTY	-	0.00	0.00
PC	1	700.00	700.00
Work hours	277	40.00	11080.00
			11820.61

Table 11.1: Project budget

12. Environmental impact

Cypress Semiconductor's products comply with the RoHS directive (Restriction of Hazardous Substances Directive) **[29]**. This means that the PSoC 4 does not include more Lead, Mercury, Hexavalent Chromium, Cadmium, PBB and PBDE than the European Union allows.

The project also has an acoustic impact, which can be dangerous to human hearing. This is why an adequate resistance has to be selected for the RC filter, so that the audio's volume is within acceptable levels.

Therefore, it can be concluded that this project has a minimal impact on the environment.

Conclusion

The objectives proposed at the start of the project have all been fulfilled at its completion. Some expectations, such as those of the resulting sound quality, have even been exceeded. The final program has a vastly improved sound quality, thanks to a significant increase in bit resolution and the addition of more accurate waveforms and amplitude envelopes. An extra octave, another voice and a wider note frequency range have also been added successfully.

The project as a whole has contributed to:

- Learning how to use the PSoC Creator 4.1 IDE and a PSoC 4 microcontroller.
- Learning the C programming language and how to use it in a practical case.
- Learning how to troubleshoot a microcontroller application.
- Understanding how FreeRTOS works.
- Learning how an audio synthesizer works and how it can be improved.

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