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DIPLOMA THESIS

**Implementation of a new PWM approach
for class-D digital audio amplifier**

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In Prague December 12, 2010

Declaration

I declare that I have created my Diploma Thesis on my own and I have used only literature cited in the included reference list.

In Prague, _____

signature

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Abstract

In this diploma thesis I would like to present a new approach to develop the digital class-D audio amplifier. Then I will describe one of the alternate implemetations for generating the optimal pulse width modulation (PWM) of odd bi-level waveform.

The optimal PWM algorithm is used to solve problem of generating the PWM output switching waveform when the input are well-known frequency spectrums. The main task of this problem is to determine switching times, where the PWM signal changes its state. We have also discovered that with using this algorithm we can determine all n numbers of switching times in $\mathcal{O}(n \log^2 n)$ times.

Further, the algorithm has been implemented off-line in programming language C with using ARM GNU software/tools, where the coding algorithm is performed by software routines that wrote to a file switching times instants. Input data are then fed to a very simple hardware - a microcontroller that generates a PWM-like output signal to directly drive a high-efficiency switching audio amplifier. The final result of the implementation shows the hardware architecture is used to implement the algorithm, it requires a hight frequency speed to generate precisely PWM output corresponding to switching times.

Contents

List of Figures	vii
List of Tables	ix
1 Introduction	1
1.1 Objectives of the Thesis	2
1.2 Methods	2
1.3 Outline of the Thesis	3
2 Requirements analysis and system architecture	4
2.1 Overview of digital class-D audio amplifiers	4
2.1.1 History	4
2.1.2 Basic principles	6
2.2 New strategy to implement the digital class-D audio amplifier	8
2.2.1 Frequency spectrum of odd signal	10
2.2.1.1 Fast Fourier Transform of odd signal	10
2.2.1.2 Efficient Fast Fourier Transform (FFT) algorithm	13
2.2.2 Optimal PWM modulation problem of odd bi-level waveform	14
2.2.2.1 Algorithm of odd bi-level PWM waveform	17
2.3 System architecture	20
2.3.1 Architectural design	20
2.3.1.1 Hardware components	20
2.3.1.2 Software components	21
3 Hardware Support	23
3.1 Hardware components	23
3.1.1 AT91SAM9G20 processor	23
3.1.2 Embedded modules	26

3.1.2.1	The module OC8-S	26
3.1.2.2	The OC8-H header board	28
3.2	Software issues for OC8-S	28
4	Starting with an embedded Linux	30
4.1	Overview	30
4.2	Configuring the software environment	32
4.2.1	Hosting Target Boards	33
4.3	The GNU Toolchain	35
4.4	Bootloader	37
4.4.1	A Universal Bootloader: Das U-Boot	38
4.4.2	Building U-boot	39
4.4.3	Downloading the U-Boot onto OC8-S	40
4.4.4	Important routines	41
4.5	Linux distribution	44
4.5.1	Getting an embedded Linux	44
4.5.2	Adding new drivers and application	45
5	Software	48
5.1	USART device driver	48
5.1.1	Overview	48
5.1.2	Implementation of USART driver	49
5.2	Fast Fourier Transform	57
5.3	The optimal PWM	60
5.4	Compiling driver and application	62
6	Testing and final work	63
6.1	Software testing	63
6.1.1	The optimal PWM algorithm	63
6.2	Embedded PWM generator	67
6.3	Frequent error scenarios	71
6.4	Final work	71
7	Conclusion	73

A Algorithms/Mathematical Background	I
A.1 QR decomposition	I
A.2 A pseudocode algorithm for finding real and complex roots of real polynomials with multiple roots	II
B Content of the Attached CD	III

List of Figures

1.1	General scheme for the digital class-D audio amplifier.	2
2.1	The classification of audio amplifiers class-D.	5
2.2	A basic audio amplifier class-D with PWM comparator, FET output stage, and second-order LC output filter	6
2.3	The PWM process as performed by a differential comparator	7
2.4	Scheme of h-bridge for audio amplifiers class-D	7
2.5	Familiar versions of filters for h-bridge output	8
2.6	Block diagram of the proposed class-D audio amplifier	8
2.7	Main tasks of the block Audio signal/Optimal PWM	9
2.8	Block diagram of the digital class-D audio amplifier implemented by the optimal PWM algorithm	9
2.9	The principle of audio amplifier class-D using the optimal PWM algorithm of odd signal.	10
2.10	The tree of input vectors to the iterative calls of the FFT procedure.	13
2.11	(a) Frequency spectrum of a separated base-band signal. The base-band can be recovered by an LPF. (b) Principal scheme for optimal PWM problem.	15
2.12	Odd bi-level PWM waveform.	16
3.1	AT91SAM9G20 Block Diagram	25
3.2	The embedded processor module Linux systems OC8-S	26
3.3	Hardware architecture of the module OC8-S	27
3.4	The OC8-H header board	28
4.1	Cross-development setup	33
5.1	The block diagram of USART	49
5.2	Principle of <code>gpio_pwm_write()</code>	54

5.3	A circular buffer	54
5.4	Bit reversal process in FFT	58
6.1	Odd bi-level PWM waveform for $n = 20$	65
6.2	Spectrums of odd bi-level PWM waveform for $n = 20$	65
6.3	PWM output signal for $n = 5$	67
6.4	Simulated odd bi-level PWM waveform for $n = 5$	67
6.5	The first period of the real PWM output signal for $n = 5$	68
6.6	The measured and computed PWM output signal fo $n = 5$	68
6.7	PWM output signal for $n = 4$	69
6.8	Simulated odd bi-level PWM waveform for $n = 4$	69
6.9	The first period of the real PWM output signal for $n = 4$	70
6.10	The real and computed PWM output signal for $n = 4$	70
7.1	The prototype of the digital class-D audio amplifier.	74

List of Tables

4.1	Display all serial ports	33
4.2	Detailed info about serial devices on the USB ports	34
4.3	Minicom terminal	34
4.4	TFTP Configuration	35
4.5	NFS Server Configuration	35
4.6	Unpacking toolchain	37
4.7	Configuration address for CodeSoucery in workstation.	37
4.8	Create default configuration OC8-S	39
4.9	Configuration U-Boot.	40
4.10	Cross-compiling.	40
4.11	Das U-Boot	41
4.12	U-Boot's network settings	41
4.13	U-Boot's network settings	42
4.14	Erasing the flash memory	42
4.15	Writting uImage to the Flash memory	42
4.16	Downloading the root file system	43
4.17	Writting the root file system to the Flash memory	43
4.18	Reading of the uImage to the RAM	43
4.19	The boot parameter	43
4.20	Boot uImage	44
4.21	Getting the kernel source	45
4.22	Configuration of the Linux kernel	45
4.23	Creating the Linux kernel	45
4.24	Set PATH environtmnet for compiling	46
4.25	Creating the Linux kernel	46
4.26	Loading the device driver to kernel	46
4.27	Creating a user application	47

4.28	Loading application onto OC8-S	47
5.1	Define module parameters	50
5.2	Define pinout	50
5.3	Peripheral initialization	51
5.4	File_operations structure	51
5.5	File structure	52
5.6	Using mutex	53
5.7	Interrupt handler	56
5.8	Bit-reversed function	58
5.9	Fast Fourier Transform	59
6.1	The partial results for case $n = 20, n_C = 5, n_E = 15, A = 10, T = 0.01$ and $(b_{f_1}, b_{f_2}, b_{f_3}, b_{f_4}, b_{f_5}) = (4, -5, 3, 1, -2)$	64
6.2	The partial results for case $n = 5, n_C = 1, n_E = 4, A = 15, T = 0.0002$ and $(b_{f_1}) = (15)$	66
6.3	Execute optimalpwm	67
6.4	The partial results for case $n = 4, n_C = 1, n_E = 3, A = 6, T = 0.001$ and $(b_{f_1}) = (3)$	69
6.5	Show the status of modules in the Linux Kernel	71

Chapter 1

Introduction

In practice, audio amplifiers class-D use pulse width modulation (PWM) as the preferred modulation technique to generate the output switching waveform. A controller converts analog or digital audio to a PWM signal then it is amplified by the maturing of metal-oxidesemiconductor field-effect transistors(MOSFETs). In fact, PWM presents a signal into a few discrete levels, with the information represented in pulse duty ratios. The digital audio signal is perfect for making PWM signal when all digital audio has a finite resolution. This resolution can be quickly translated into a set pulse width, effectively eliminating a A/D then D/A conversion which can cause errors in the signal due to re-sampling. These varying pulse widths can be used to drive the h-bridge circuits present in the class-D amplifiers to the correct positive or negative state for its given length of time. Of course we have to account for switching losses and discontinuous states, but audio amlifiers class-D with a good control algorithm can easily reach 90% efficiency.

Various simplified works have been used to implement a digital audio amplifier that can be divided into three groups: the first one is the derivation of digital PWM techniques, the second one is the design of digital controller for audio amplifier class-D and the last one is the design of digital audio interface. A principle of interpolation methods for sampled data conversion and noise shaping techniques for improving the spectral distortion are cause why audio amplifiers class-D have a high reliability. Audio amplifiers class-D use different kind of h-bridge or full-bridge topologies to reach the hight-efficiency goal [1].

1.1 Objectives of the Thesis

The primary objective of this thesis is to propose a new strategy to implement a digital class-D audio amplifier. In this diploma thesis, the optimal PWM algorithm of odd bi-level waveform is used to implement. We will try determine switching times of PWM output signal when the input are well-known frequency spectrums.

The secondary objective is to propose, develop and implement the algorithm of the optimal PWM odd bi-level waveform in programming language C for an embedded module with using ARM GNU software/tools. Furthermore, the validity of the implemented algorithm will have to be verified by a selected hardware prototype with platform ARM9.

1.2 Methods

Objectives of this thesis are accomplished via the algorithm of the optimal PWM odd bi-level waveform from detailed background study of the topic and to develop a general scheme like Figure 1.1, that uses this algorithm to implement with a prototype of the digital class-D audio amplifier. To do this, it requires a PC to pass values from well-known frequency spectrums to the embedded module OC8-S based on the ARM9 ATMEL AT91SAM9G20, in that is already implemeted the algorithm of optimal PWM odd bi-level waveform. This module will take these values into a buffer and then release these bits of information at sampling rate and it will pass the PWM control signal to the h-bridge power converter. The h-bridge output will be filtered and delivered to a speaker load. Rather a simple microcontroller will be used, because it is a simple and universal hardware with a building embedded linux system for development.

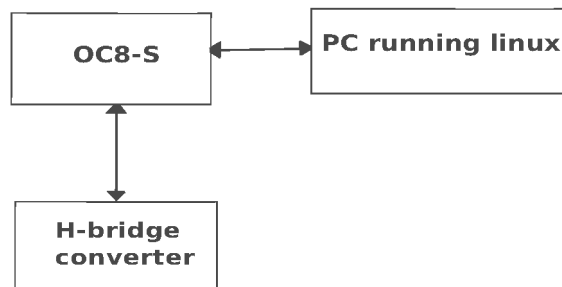


Figure 1.1: General scheme for the digital class-D audio amplifier.

1.3 Outline of the Thesis

This section will give an overview about the work that was done. The chapters this document is divided into correspond to phases of development.

- Chapter 1 – **Introduction** presents the studied topic and proposed goals of the thesis. It gives insight to a implementation of the optimal PWM of odd bi-level waveform.
- Chapter 2 – **Requirements analysis and system architecture** contains a detailed requirements analysis and system architecture for the digital class-D audio amplifiers. Basic inputs and outputs are determined and a list of required hardware and software components is made.
- Chapter 3 – **Hardware Support** presents the embedded hardware Linux supports to develop the digital class-D audio amplifiers. The hardware components are introduced and configured. The selected hardware builds upon a ARM9 processor from ATMEL. Chapter ends with software issues for OC8-S.
- Chapter 4 – **Starting with the embedded Linux** introduces the embedded Linux operating system for the ARM9 processor. In this chapter, the exercised of configuring the software environment will be described and components that are required for a complete embedded Linux operating system will be specified.
- Chapter 5 – **Software** is the main chapter of this document because it describes the software development phase. Device driver is implemented first: For USART(Universal Synchronous Asynchronous Receiver Transceiver) one has to be implemented for an embedded Linux kernel. Thereafter, two user space applications are developed: The Fast Fourier Transform and the optimal PWM algorithm.
- Chapter 6 – **Testing and final work** deals with testing and final work. Accurate testing is conducted with the main application. Finally, the work is completed by deploying the software onto the target.
- Chapter 7 – **Conclusion** summarizes goals and obtained results of the thesis. Problems are pointed out and a list of imaginable future enhancements of the digital class-D audio amplifiers is presented

Chapter 2

Requirements analysis and system architecture

This chapter presents various sections about the planning phase of a prototype audio amplifier class-D. The requirements elicitation process starts with basic principles of audio amplifiers class-D where a general idea about the capabilities of the system is found. Afterwards, a new strategy to implement a prototype audio amplifier class-D is described. Furthermore, the hardware/software component analysis is detailed described in section system architecture .

2.1 Overview of digital class-D audio amplifiers

2.1.1 History

The task of a power audio amplifier is to reproduce input audio signals at sound producing output elements, with desired volume and power levels faithfully, efficiently, and at low distortion. Audio frequencies range from about 20 Hz to 20 kHz, so the amplifier must have good frequency response over this range (less when driving a band-limited speaker, such as a woofer or a tweeter). Power capabilities vary widely depending on the application, from milliwatts in headphones, to a few watts in TV or PC audio, to tens of watts for "mini" home stereos and automotive audio, to hundreds of watts and beyond for more powerful home and commercial sound systems and to fill theaters or auditoriums with sound.

While principles of audio amplifiers class-D cited in 1947, it is regarded as was invented in 1950 in th UK by Dr. A. H. Reeves, father of Pulse Code Modulation(PCM). Audio amplifiers class-D have divided into different cathegories, depending on its topologies, numbers of audio connector and types of modulator [2].

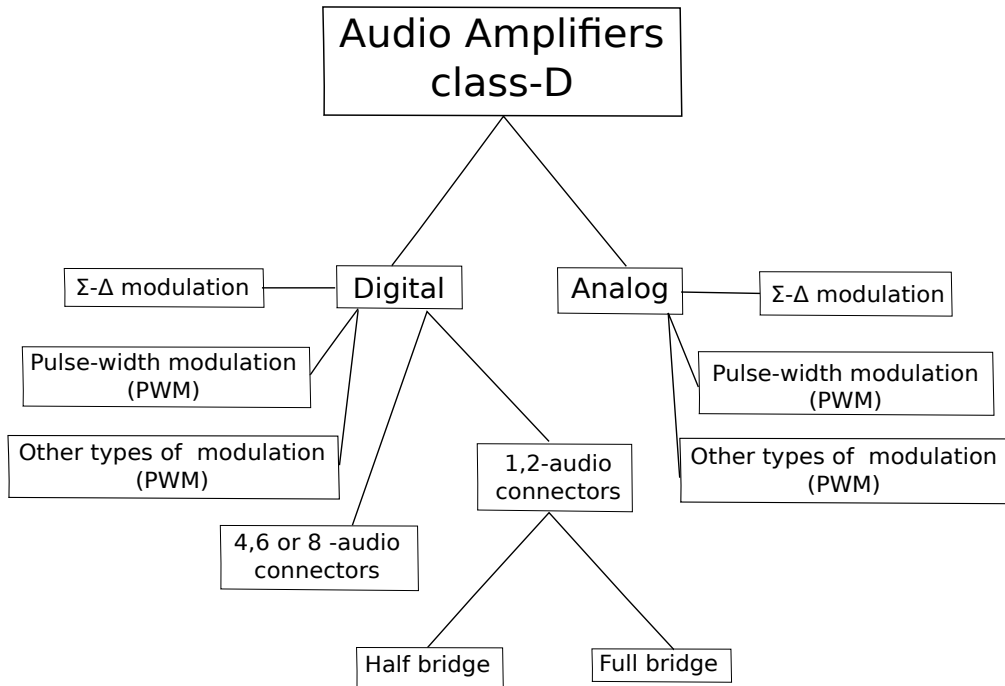


Figure 2.1: The classification of audio amplifiers class-D.

As can be seen in Figure 2.1, the audio amplifiers class-D can be divided into two basic groups. The first one are analog audio amplifiers class-D that switch amplifiers with an analog input signal and an analog control system. Usually there is present some degree of feedback error correction. The second are full digital audio amplifiers class-D that provide to work directly with a digital input signal. Amplifiers with a digitally generated control that switch as power stage. No error control is present. Those that do have an error control can be show to be topologically equivalent to an analog - control class-D with a DAC convert.

Both groups use switching power stages. While real operating efficiency in class-AB amplifiers provide around 20%, class-D can be easily reach 90% efficiency without significant effort. Higher efficiencies are possible depending on details of the design with higher power (around 100W or more) amplifiers actually attaining higher efficiencies than their low power relatives. There are the largest advantages of audio amplifiers class-D.

2.1.2 Basic principles

Audio amplifiers class-D differ radically from the more familiar classes of A, B and G. In class-D there are no output devices operating in the linear mode. Instead they are switched on and off at an ultrasonic frequency, the output being connected alternately to each supply rail. When the mark-space ratio of the input signal is varied, the average output voltage varies with it, the averaging being done by a low-pass output filter, or by the loudspeaker inductance alone. Note that the output is also directly proportional to the supply voltage; there is no inherent supply rejection at all with this sort of output stage, unlike the class-B output stage. The use of negative feedback helps with this. The switching frequencies used range from 50 kHz to 1 MHz. A higher frequency makes the output filter simpler and smaller, but tends to increase switching losses and distortion. The classic method of generating the drive signal is to use a differential comparator. One input is driven by the incoming audio signal, and the other by a sawtooth waveform at the required switching frequency [3].

Basic audio amplifiers class-D are shown in Figure 2.2,

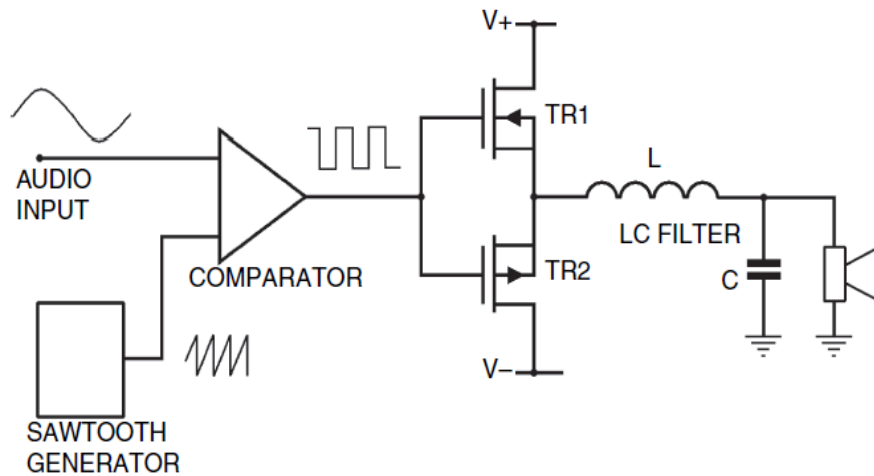


Figure 2.2: A basic audio amplifier class-D with PWM comparator, FET output stage, and second-order LC output filter

The PWM process is illustrated in Figure 2.3 . Clearly the sawtooth needs to be linear (i.e., with constant slope) to prevent distortion being introduced at this stage. There are other ways to create the required waveform, such as a sigma-delta modulator.

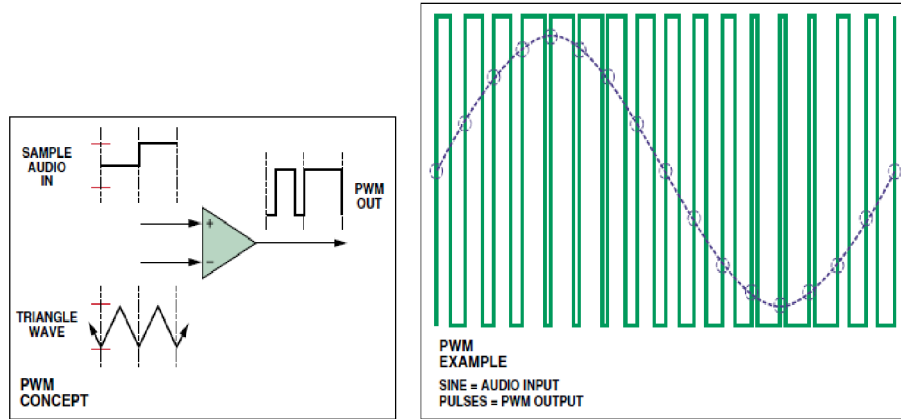


Figure 2.3: The PWM process as performed by a differential comparator

When the aim is to produce as much audio power as possible from a low voltage supply such as 5 V, the h-bridge configuration is employed, as shown in Figure 2.4 (for more see [4]). It allows twice the voltage-swing across the load, and therefore theoretically four times the output power, and also permits the amplifier to run from one supply rail without the need for bulky output capacitors of doubtful linearity. This method is also called the Bridge-Tied Load, or BTL [4].

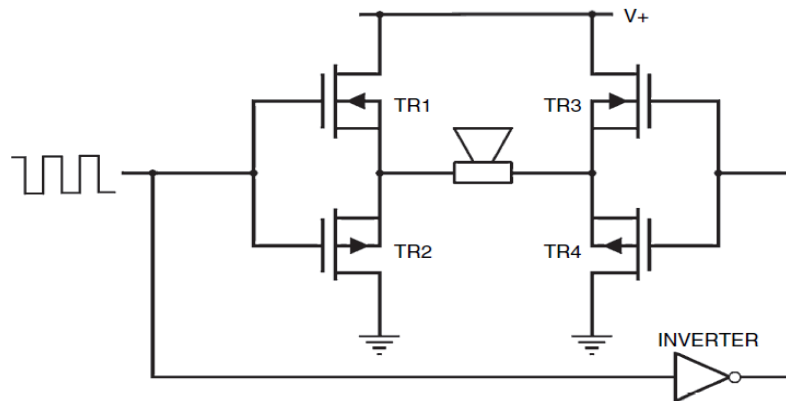


Figure 2.4: Scheme of h-bridge for audio amplifiers class-D

Familiar versions of filters for h-bridge for audio amplifiers working in class-D are shown in Figure 2.5. Filter in Figure 2.5a is simplest but allows a common-mode signal on the speaker cabling; filter in Figure 2.5b and Figure 2.5c are most usual version; in Figure 2.5d is a 4-pole filter.

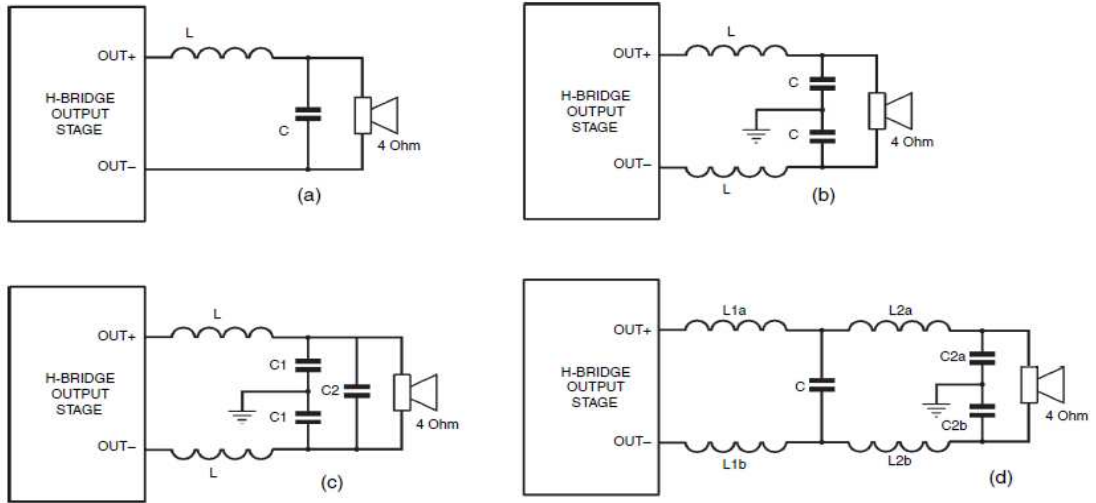


Figure 2.5: Familiar versions of filters for h-bridge output

2.2 New strategy to implement the digital class-D audio amplifier

In the previous section it was introduced to new strategy to implement the digital audio amplifier working in class-D. It is a brand-new approach to generate a drive signal without using a differential comparator. The basic principle of this method is shown in Figure 2.6. This method was developed by Ing. Petr Kujan, Ph.D in his dissertation thesis [5], namely the optimal PWM algorithm for odd single-phase multilevel problem. In this project, the algorithm is just applied and implemented for odd bi-level waveform.

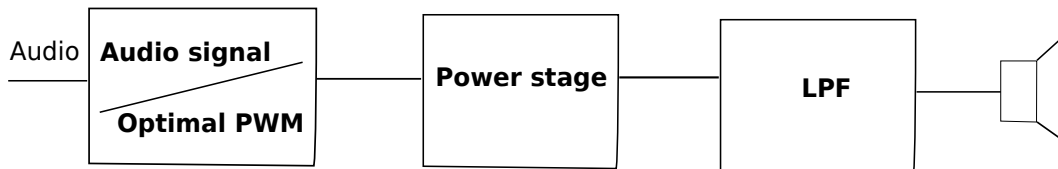


Figure 2.6: Block diagram of the proposed class-D audio amplifier

Input audio signals are typically introduced into the block **Audio signal/Optimal PWM**. Here it is possible to realize with using a microcontroller. And the output PWM will be generated by FPGA or another microcontroller. The principle is shown in Figure 2.7.

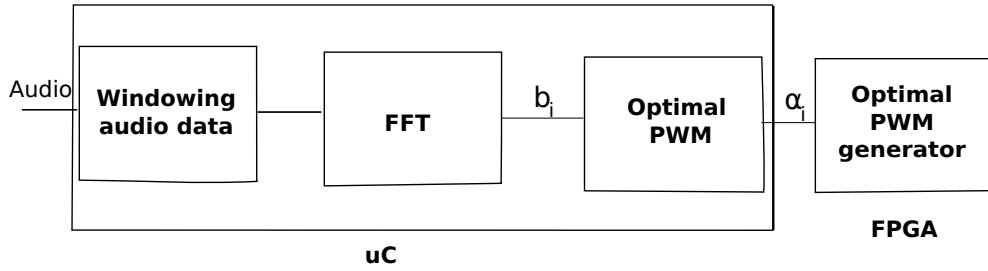


Figure 2.7: Main tasks of the block Audio signal/Optimal PWM

In this project, the algorithm has been implemented off-line in programming language C with using ARM GNU software/tools on an PC workstation, where the coding algorithm is performed by software routines that wrote to a file of switching times instants. Input data are then fed to a very simple hardware - a microcontroller that generates a PWM-like output signal to directly drive a high-efficiency switching audio amplifier.

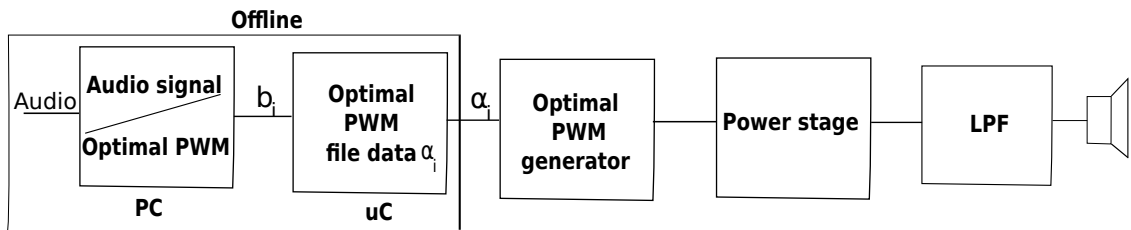


Figure 2.8: Block diagram of the digital class-D audio amplifier implemented by the optimal PWM algorithm

Since the digital audio amplifier class-D using the optimal PWM algorithm per se implies nothing than being able to play audio data, in this phase additional features are found and a basic concept of device is created. Two questions are important during this phase:

1. What features should be supported by the system?
2. What are the inputs and outputs of system?

The features of the digital class-D audio amplifier implemented by the optimal PWM algorithm:

- The audio amplifier class-D should be worked similar to a traditional amplifier.
- Audio playback should also be possible from some mass storage.

- The playback volume should be adjustable.

Figure 2.7 points out the main input and output components that will be needed, based on the following input/ouput analysis of the system:

Inputs:

- Frequency spectrums from a mass storage device.

Ouputs:

- PWM audio controlled signal directly do h-bridge.

The principle of the digital class-D audio amplifiers implemented by the optimal PWM algorithm can be illustrated in Figure 2.9

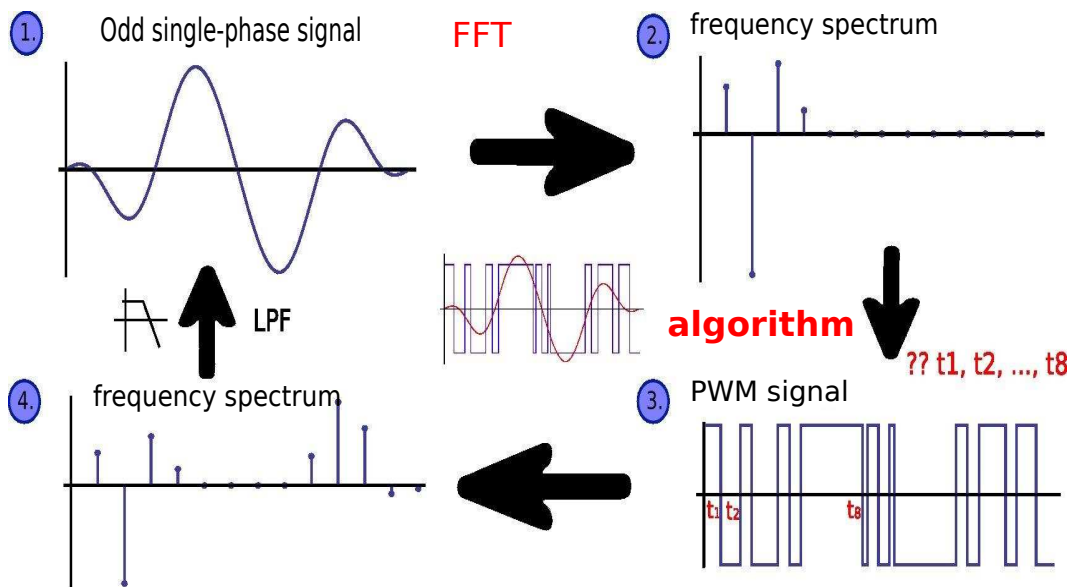


Figure 2.9: The principle of audio amplifier class-D using the optimal PWM algorithm of odd signal.

2.2.1 Frequency spectrum of odd signal

2.2.1.1 Fast Fourier Transform of odd signal

Without loss of generality, we consider the digital sequence x_k consisting of 2^m samples, where m is positive integer - the number of samples of digital sequence x_k is power of 2,

$N_x = 2, 4, 8, 16$, etc. We begin with the definition of Discrete Fourier Transform (DFT):

$$X_k = \sum_{n=0}^{N_x-1} x_n W_{N_x}^{kn}, \quad \text{for } k = 0, 1, \dots, N_x - 1, \quad (2.1)$$

where $W_{N_x} = e^{-j\frac{2\pi}{N_x}}$ is the twiddle factor, and $N_x = 2, 4, 8, 16, \dots$

Using the Euler's formula of complex analysis $e^{i\phi} = \cos \phi + i \sin \phi$, it follows that

$$X_k = \sum_{n=0}^{N_x-1} x_n \cos\left(\frac{-2\pi kn}{N_x}\right) + j \sum_{n=0}^{N_x-1} x_n \sin\left(\frac{-2\pi kn}{N_x}\right), \quad (2.2)$$

for $k = 0, 1, \dots, N_x - 1$.

Equation (2.1) can be expanded as

$$X_k = x_0 + x_1 W_{N_x}^k + \dots + x_{N_x-1} W_{N_x}^{k(N_x-1)}, \quad (2.3)$$

Again, if we split Equation (2.3) in to

$$X_k = x_0 + x_1 W_{N_x}^k + \dots + x_{N_x/2-1} W_{N_x}^{k(N_x/2-1)} \\ + x_{N_x/2} W_{N_x}^{kN_x/2} + \dots + x_{N_x-1} W_{N_x}^{kn}, \quad (2.4)$$

then we can rewrite it as a sum of following two parts

$$X_k = \sum_{n=0}^{N_x/2-1} x_n W_{N_x}^{kn} + \sum_{n=N_x/2}^{N_x-1} x_n W_{N_x}^{kn} \quad (2.5)$$

Now we consider the digital sequence y_k consisting of odd signal x_k . For this digital sequence, we get :

$$y_k = -y_k(N_y - k), \quad (2.6)$$

where N_y is the number of samples of digital sequence y_k .

Similar to Equation (2.1), the frequency spectrum is given by

$$Y_k = \sum_{n=0}^{N_y-1} y_n W_{N_y}^{kn}, \quad \text{for } k = 0, 1, \dots, N_y - 1, \quad (2.7)$$

where $W_{N_y} = e^{-j\frac{2\pi}{N_y}}$ is the twiddle factor, and $N_y = 2, 4, 8, 16, \dots$

Similarly, if we split Equation (2.7) in two parts as

$$Y_k = \sum_{n=0}^{N_y/2-1} y_n W_{N_y}^{kn} + \sum_{n=N_y/2}^{N_y-1} y_n W_{N_y}^{kn} \quad (2.8)$$

thus, we obtain

$$\begin{aligned} Y_k &= \sum_{n=0}^{N_y/2-1} y_n \cos\left(\frac{-2\pi kn}{N_y}\right) + j \sum_{n=0}^{N_y/2-1} y_n \sin\left(\frac{-2\pi kn}{N_y}\right) \\ &+ \sum_{n=N_y/2}^{N_y-1} y_n \cos\left(\frac{-2\pi kn}{N_y}\right) + j \sum_{n=N_y/2}^{N_y-1} y_n \sin\left(\frac{-2\pi kn}{N_y}\right), \end{aligned} \quad (2.9)$$

for $k = 0, 1, \dots, N_y - 1$.

Now, we have the following equations

$$\cos(k\pi + \varphi) = \cos(k\pi - \varphi); \quad (2.10)$$

$$\sin(k\pi + \varphi) = -\sin(k\pi - \varphi); \quad (2.11)$$

for $k \in \mathcal{Z}$.

It follows that

$$\cos(k\pi(1+q)) = \cos(k\pi(1-q)); \quad (2.12)$$

$$\sin(k\pi(1+q)) = -\sin(k\pi(1-q)); \quad (2.13)$$

for $k \in \mathcal{Z}$, $q \in \mathcal{R}$.

Equation (2.10) presents the symmetry property of sine and cosine function about $k\pi$. Thus, Equation (2.9) becomes

$$\begin{aligned} Y_k &= \sum_{n=0}^{N_y/2-1} y_n \cos\left(\frac{-2\pi kn}{N_y}\right) + j \sum_{n=0}^{N_y/2-1} y_n \sin\left(\frac{-2\pi kn}{N_y}\right) \\ &- \sum_{n=0}^{N_y/2-1} y_n \cos\left(\frac{-2\pi kn}{N_y}\right) + j \sum_{n=0}^{N_y/2-1} y_n \sin\left(\frac{-2\pi kn}{N_y}\right), \end{aligned} \quad (2.14)$$

for $k = 0, 1, \dots, N_y - 1$.

From Equation (2.14), because real parts reduce to zero, the final result is given by

$$Y_k = 2j \left(\sum_{n=0}^{N_y/2-1} y_n \sin\left(\frac{-2\pi kn}{N_y}\right) \right) \quad (2.15)$$

for $k = 0, 1, \dots, N_y - 1$,

where Y_k is the frequency spectrum of odd signal y_k .

2.2.1.2 Efficient Fast Fourier Transform (FFT) algorithm

By using the Fast Fourier Transform (FFT), which takes advantage of the special properties of the complex roots of unity, we can compute $DFT_n(a)$ in time $\mathcal{O}(n \log n)$, as opposed to the $\mathcal{O}(n^2)$ times of the straightforward method. In practice, we can compute the DFT with recursive or iterative FFT algorithm. In this project, the iterative FFT algorithm is used to implement.

We now show how to make the FFT algorithm iterative. In Figure 2.10 we have arranged the input vectors $A[0 \dots n - 1]$ in an iterative invocation to the iterative calls in a tree structure, where the initial call is for $n = 8$. The tree has one node for each call of the procedure, labeled by the corresponding input vector. Each iterative invocation makes two iterative calls, unless it has received a 1-element vector.

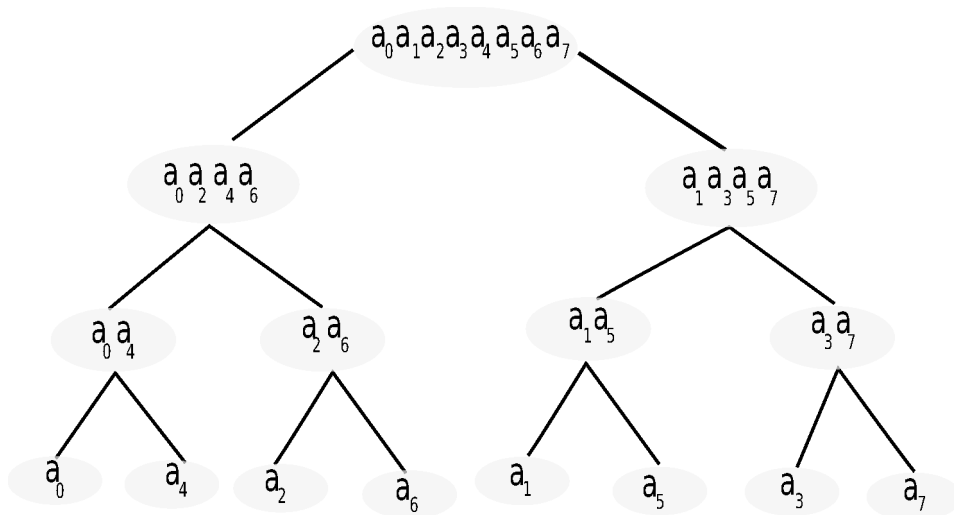


Figure 2.10: The tree of input vectors to the iterative calls of the FFT procedure.

The pseudocode of FFT algorithm is shown in Algorithm 2.2.1. The code first calls the auxiliary procedure BIT-REVERSE-COPY (a, A) to copy vector into array A in the initial order in which we need the values. The twiddle factor w_n used in each butterfly operation depends on the value of s , it is a power of w_m , where $m = 2^s$.

Algorithm 2.2.1: ITERATIVE-FFT(a)

```

BIT-REVERSE-COPY(a,A)
 $n \leftarrow \text{length}[a]$   ▷  $n$  is a power of 2
for  $s \leftarrow 1$  to  $\log n$ 
  do  $m \leftarrow 2^s$ 
   $w_m \leftarrow e^{2\pi i/m}$ 
  comment: Here begins the Danielson-Lanczos Lemma section of the routine
  for  $k \leftarrow 0$  to  $n - 1$  by  $m$ 
    do  $w \leftarrow 1$ 
    for  $j \leftarrow 0$  to  $m/2 - 1$ 
      do  $t \leftarrow wA[k + j + m/2]$ 
     $u \leftarrow A[k + j]$ 
     $A[k + j] \leftarrow u + t$ 
     $A[k + j + m/2] \leftarrow u - t$ 
   $w \leftarrow w \cdot w_m$ 

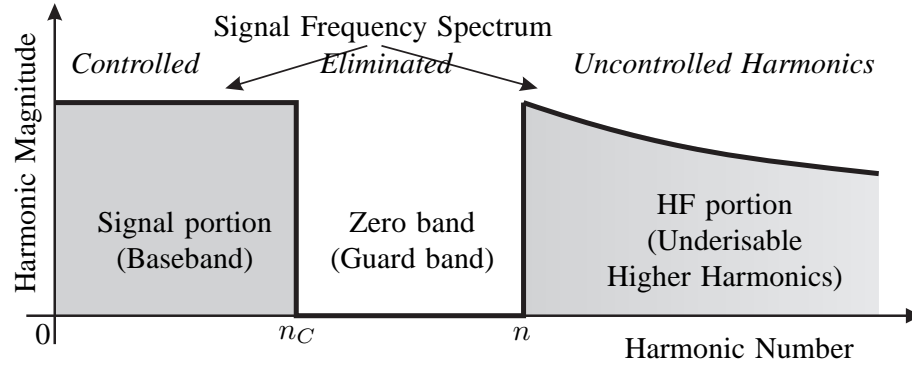
```

The iterative FFT implementation runs in time $\mathcal{O}(n \log n)$. The call to BIT-REVERSE-COPY(a , A) certainly runs in $\mathcal{O}(n \log n)$ times, since we iterate n times and can reverse an integer between 0 and $n - 1$, with $\log n$ bits, in $\mathcal{O}(\log n)$ times.

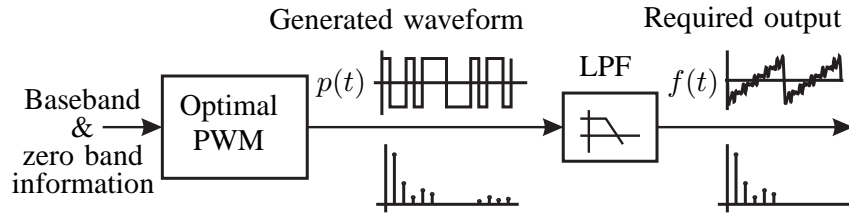
2.2.2 Optimal PWM modulation problem of odd bi-level waveform

In the preceding section 2.2.1 the optimal PWM modulation is introduced as a new approach to determine a sequence of switching times $\bar{\alpha}$. In this section it will be described detailed and the algorithm to solve it will be shown at the end.

Key issue of the optimal PWM problem is to determine the switching times (angles) so as to produce the signal portion (base-band) and not generate specific higher order harmonics (guard band or zero band). This spectral gap separates the base-band which has to be identical to the required output waveform, from an uncontrolled higher frequency portion. The required output signal can be recovered by means of an analog low-pass filter (LPF) with cutoff frequency in the guard band. The procedure is depicted on the Figure 2.11 [5].



(a)



(b)

Figure 2.11: (a) Frequency spectrum of a separated base-band signal. The base-band can be recovered by an LPF. (b) Principal scheme for optimal PWM problem.

Methods described in this section are based on exploiting appropriate trigonometric transcendental equations that define the harmonic content of the generated periodic PWM waveform $p(t)$ which is equal to required finite frequency spectrum of $f(t)$. The main problem lies in solving these systems of equations.

The solution of the optimal PWM problem is a sequence of switching times $\bar{\alpha}^* = (\alpha_1, \dots, \alpha_n)$. This sequence is obtained from the solution of the system of equations

$$\begin{aligned}
 & a_{p_0}(\bar{\alpha}) = a_{f_0} , \\
 & \left. \begin{aligned} a_{p_k}(\bar{\alpha}) &= a_{f_k} \\ b_{p_k}(\bar{\alpha}) &= b_{f_k} \end{aligned} \right\} \text{ for all } k \in \mathcal{H}_C, \\
 & \left. \begin{aligned} a_{p_k}(\bar{\alpha}) &= 0 \\ b_{p_k}(\bar{\alpha}) &= 0 \end{aligned} \right\} \text{ for all } k \in \mathcal{H}_E, \\
 & \text{subject to } \quad 0 < \alpha_i < T,
 \end{aligned}$$

where $\bar{\alpha} = (\alpha_1, \dots, \alpha_n)$ are unknown variables, a_{p_0} and a_{p_k} , b_{p_k} are zeroth and k -th

cosine, respectively sine Fourier coefficients of the generated waveform $p(t)$, a_{f_0} and a_{f_k} , b_{f_k} are zeroth and k -th cosine, sine Fourier coefficients of the required output waveform $f(t)$. The \mathcal{H}_C is the set of controlled harmonics and the number of elements is n_C . The \mathcal{H}_E is the set of eliminated harmonics and the number of elements is n_E . The number of equations is $n = 1 + 2(n_C + n_E)$.

Without loss of generality, we consider the Fourier series of T periodic odd bi-level PWM waveform $p(t)$ like Figure 2.12 with amplitude A is sine,

$$p(t) \sim \sum_{k=1}^{\infty} b_k \sin wkt \quad (2.17)$$

where

$$b_k = \frac{4A}{k\pi} \left(o_{n+k} + \sum_1^n (-1)^i \cos wk\alpha_i \right) \quad (2.18)$$

for $k = 1, 2, 3, \dots$

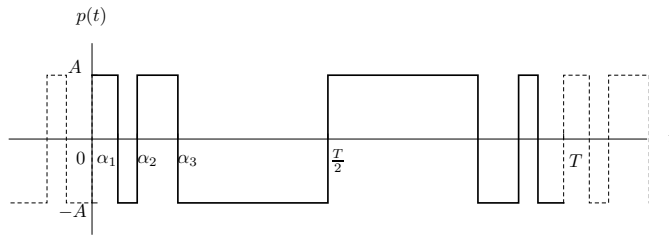


Figure 2.12: Odd bi-level PWM waveform.

The unknown switching times $\bar{\alpha} = (\alpha_1, \dots, \alpha_n)$ are subject to $0 < \alpha_1 < \alpha_2 < \dots < \alpha_n < T/2$ and $\omega = 2\pi/T$ is angular frequency. The integer n is number of switching times in the half period. The parameters are :

$$A_k = \frac{4A}{k\pi}, \quad B_k = o_{n+k}, \quad C_k = 1. \quad (2.19)$$

2.2.2.1 Algorithm of odd bi-level PWM waveform

Algorithm 2.2.2: OPTIMALPWM: COMPUTE OPTIMAL PWM PROBLEM($\alpha_1, \dots, \alpha_n$)

Input:

- n ... the number of switching times (it is equal to number of controlled harmonics n_C plus number of zero harmonics n_E),
 $(b_{f_1}, \dots, b_{f_{n_C}})$... the sequence of controlled harmonics,
 ω ... frequency,
 A ... amplitude of PWM waveform.

Output:

- $(\alpha_1, \dots, \alpha_n)$... the optimal switching times.

1. Compute the RHS of composite sum of powers p_i , $i = 1, \dots, n$, using

$$p_{2i} = o_n + 2^{-2i+1} \frac{\pi}{A} \sum_{j=1}^{\underline{K}} \binom{2i}{i-j} j b_{f_{2j}}, \quad (2.20a)$$

$$\underline{K} := \begin{cases} i & \dots i < \lfloor n_c/2 \rfloor, \\ \lfloor n_c/2 \rfloor & \dots i \geq \lfloor n_c/2 \rfloor, \end{cases} \quad (2.20b)$$

$$i = 1, 2, \dots, \lfloor n/2 \rfloor,$$

$$p_{2i-1} = -o_n + 2^{-2i+1} \frac{\pi}{A} \sum_{j=1}^{\overline{K}} \binom{2i-1}{i-j} (2j-1) b_{f_{2j-1}}, \quad (2.20c)$$

$$\overline{K} := \begin{cases} i & \dots i < \lceil n_c/2 \rceil, \\ \lceil n_c/2 \rceil & \dots i \geq \lceil n_c/2 \rceil, \end{cases} \quad (2.20d)$$

$$i = 1, 2, \dots, \lceil n/2 \rceil.$$

where o_n is the odd parity test:

$$o_n = \frac{1 - (-1)^n}{2} = \begin{cases} 0 & \text{for even } n, \\ 1 & \text{for odd } n. \end{cases} \quad (2.21)$$

2. Compute composite sum of powers

$$y_1^i + \dots + y_{\lfloor n/2 \rfloor}^i - y_{\lfloor n/2 \rfloor + 1}^i \dots - y_n^i = p_i, \quad i = 1, \dots, n. \quad (2.22)$$

using Algorithms PadeCSop 2.2.3.

Set $(\bar{y}^+, \bar{y}^-) = XXXCSOP(p_1, \dots, p_n)$.

3. **if** $\bar{y}^+ \in \mathcal{R}^{(-1,1)} \wedge \bar{y}^- \in \mathcal{R}^{(-1,1)}$ **then** continue **else** exit- no exact solution

4. **end if**

5. Set $\bar{y}^+ = (y_{s1}^+, y_{s2}^+, \dots, y_s \lceil n/2 \rceil) = \text{sort} > \bar{y}^+$ where $y_{s1}^+ > y_{s2}^+ > \dots$

Set $\bar{y}^- = (y_{s1}^-, y_{s2}^-, \dots, y_s \lfloor n/2 \rfloor) = \text{sort} > \bar{y}^-$ where $y_{s1}^- > y_{s2}^- > \dots$

6. Set $\bar{x} = (x_1, \dots, x_n) = \text{riffle}(\bar{y}_{s1}^+, \bar{y}_{s2}^-) = (y_{s1}^+, y_{s1}^-, y_{s2}^+, y_{s2}^-, \dots)$.

7. Return $\bar{\alpha}^* = (\alpha_1^*, \alpha_2^*, \dots, \alpha_n^*) = \frac{1}{\omega} \arccos \bar{x}$

Algorithm 2.2.3: PADÉ METHOD(\bar{y}^+, \bar{y}^-)

Input:

$p_1, \dots, p_n \dots$ the right hand side of composite sum of powers, solved according to (2.20).

Output:

$(\bar{y}^+, \bar{y}^-) = ((y_1, y_2, \dots, y_{\lfloor n/2 \rfloor}), (y_{\lfloor n/2 \rfloor + 1}, y_{\lfloor n/2 \rfloor + 2}, \dots, y_n))$
 \dots the solution of composite sum of powers (2.22).

1. Compute the moments $\mu_k, k = 1, \dots, n$ according to

$$\mu_0 = 1, \quad \mu_k = -\frac{1}{k} \sum_{j=1}^k p_j \mu_{k-j} \quad (2.23)$$

2. Set $p = (-1)^n p$ (* for condition $k \leq \lfloor n/2 \rfloor$ *)

3. Set $k = \lfloor n/2 \rfloor$

4. **if** n is odd integer

then

5. Solve linear Hankel system for

$$\begin{bmatrix} \mu_0 & \cdots & \mu_k \\ \vdots & \ddots & \vdots \\ \mu_k & \cdots & \mu_{2k} \end{bmatrix} \cdot \begin{bmatrix} w_{k+1,0} \\ \vdots \\ w_{k+1,k} \end{bmatrix} = - \begin{bmatrix} \mu_{k+1} \\ \vdots \\ \mu_{2k+1} \end{bmatrix}$$

6. Solve matrix equation with triangular hankel matrix

$$\begin{bmatrix} v_{k,k-1} \\ \vdots \\ v_{k,0} \end{bmatrix} = \begin{bmatrix} 0 & \cdots & \mu_0 \\ \vdots & \ddots & \vdots \\ \mu_0 & \cdots & \mu_{k-1} \end{bmatrix} \cdot \begin{bmatrix} w_{k+1,1} \\ \vdots \\ w_{k+1,k} \end{bmatrix} + \begin{bmatrix} \mu_1 \\ \vdots \\ \mu_k \end{bmatrix}$$

7. Set $W_{k+1}(y) = x^{k+1} + \sum_{i=0}^k w_{k+1,i} x^i$ and $V_k(y) = x^k + \sum_{i=0}^k v_{k,i} x^i$.

8. Return $(\bar{y}^+, \bar{y}^-) = (\text{roots}(W_{k+1}(y)), \text{roots}(V_k(y)))$

9. **else**

10. Solve linear Hankel system for

$$\begin{bmatrix} \mu_1 \cdots \mu_k \\ \vdots \cdots \vdots \\ \mu_k \cdots \mu_{2k-1} \end{bmatrix} \cdot \begin{bmatrix} w_{k,0} \\ \vdots \\ w_{k,k-1} \end{bmatrix} = - \begin{bmatrix} \mu_{k+1} \\ \vdots \\ \mu_{2k} \end{bmatrix}$$

11. Solve matrix equation with triangular hankel matrix

$$\begin{bmatrix} v_{k,k-1} \\ \vdots \\ v_{k,0} \end{bmatrix} = \begin{bmatrix} 0 \cdots \mu_0 \\ \vdots \cdots \vdots \\ \mu_0 \cdots \mu_{k-1} \end{bmatrix} \cdot \begin{bmatrix} w_{k,0} \\ \vdots \\ w_{k,k-1} \end{bmatrix} + \begin{bmatrix} \mu_1 \\ \vdots \\ \mu_k \end{bmatrix}$$

12. Set $W_k(y) = x^{k-1} + \sum_{i=0}^k w_{k,i}y^i$ and $V_k(y) = y^k + \sum_{i=0}^{k-1} v_{k,i}y^i$

13. Return $(\bar{y}^+, \bar{y}^-) = (\text{roots}(V_k(y)), \text{roots}(W_k(y)))$

14. **end if**

2.3 System architecture

2.3.1 Architectural design

Architectural design of the audio amplifier class-D is the description of a system in terms of its modules [6]. The system will now be examined from the design perspective. For the digital class-D audio amplifier device, this step is accomplished by a component-driven approach: Starting from the requirements defined in the previous section, a thorough analysis of required input/output components is done 2.9, splitted into hardware and software parts. The results of this phase are a document listing hardware and software components.

2.3.1.1 Hardware components

Embedded processor

The processor is the core of the whole system. The two most important aspects are: Execution speed and manifoldness of hardware interfaces. The required execution speed

primarily depends on the computation switching times but also influences the selection of software components, e.g. if it is possible to deploy an embedded operating system. Hardware interfaces must be available to connect all other hardware components, e.g. display, DAC, RAM, etc.

Memory

Two types of memory are required in the system:

1. RAM is mandatory for program execution on the processor and for buffering the audio stream. Again, the amount of available RAM influences selection of software components.
2. Non-volatile memory is required in the system - usually flash memory is used. This is generally important for stand-alone devices to store its firmware. Additionally, personal settings like web radio stations are stored here. Memory devices and processors are mostly interconnected through the data/address bus.

Ethernet controller

For connection to a local network, an IEEE 802.3 compatible Ethernet chip is required. It should support at least the 10BASE-T standard which allows a maximum transfer speed of 10 Mbit/s. Ethernet controllers are mostly connected to processors via the data/address bus

Power supply

The device needs a stable power supply according to the requirements of used hardware components.

2.3.1.2 Software components

Device drivers

Device drivers are needed for all hardware components that must be dealt with:

1. Embedded module driver
2. PWM generator device driver
3. Ethernet chip device driver, including a TCP/IP network stack

4. Flash memory device driver

Network protocol drivers

For the following network protocols software modules must exist or be implemented: TFTP and NFS protocols for accessing files over the network; DHCP protocol for dynamic network configuration.

Main software application

The optimal PWM algorithm is the main software part that has to be implemented. It computes switching times to generate the PWM output signal.

Chapter 3

Hardware Support

This chapter deals with the basics of embedded hardware Linux systems that is used to develop audio amplifiers class-D. Hardware decisions have to come at first here, but also software-related issues must be considered. Regarding the hardware side, it was developed on available components from the company ATMEL. Processors from AT91SAM9 based on the ARM9 architecture are often used, and a lot of hardware extension modules which were specifically designed for development and evaluation of embedded systems are available. These products are also commercially distributed by Opencontroller with the brand name "module OC8-S". More information, data sheets etc. can be obtained on the web page <http://opencontroller.com>.

Determination of hardware components is certainly not made without respect to available software for a specific processor and peripherals. The progress of this project will rely on already available software parts or programs to a rather large extent, so the more software can be relied on, the less has to be implemented from scratch.

3.1 Hardware components

3.1.1 AT91SAM9G20 processor

The former is achieved by a 32-bit Advanced RISC Machine (Reduced Instruction Set Computer) architecture, a basic MMU (Memory Mangment Unit) which allows memory protection, data and instructions caches, and support for variety of hardware peripherals.

The AT91SAM9G20 is based on the integration of an ARM926EJ-S processor with fast ROM and RAM memories and a wide range of peripherals. Figure 3.1 shows its

block diagram.

The AT91SAM9G20 embeds an Ethernet MAC, one USB Device Port, and a USB Host controller. It also integrates several standard peripherals, such as the USART, SPI, TWI, Timer Counters, Synchronous Serial Controller, ADC and MultiMedia Card Interface.

The AT91SAM9G20 is architected on a 6-layer matrix, allowing a maximum internal bandwidth of six 32-bit buses. It also features an External Bus Interface capable of interfacing with a wide range of memory devices.

The AT91SAM9G20 is an enhancement of the AT91SAM9260 with the same peripheral features. It is pin-to-pin compatible with the exception of power supply pins. Speed is increased to reach 400 MHz on the ARM core and 133 MHz on the system bus and EBI. More information can be found in the datasheet [7].

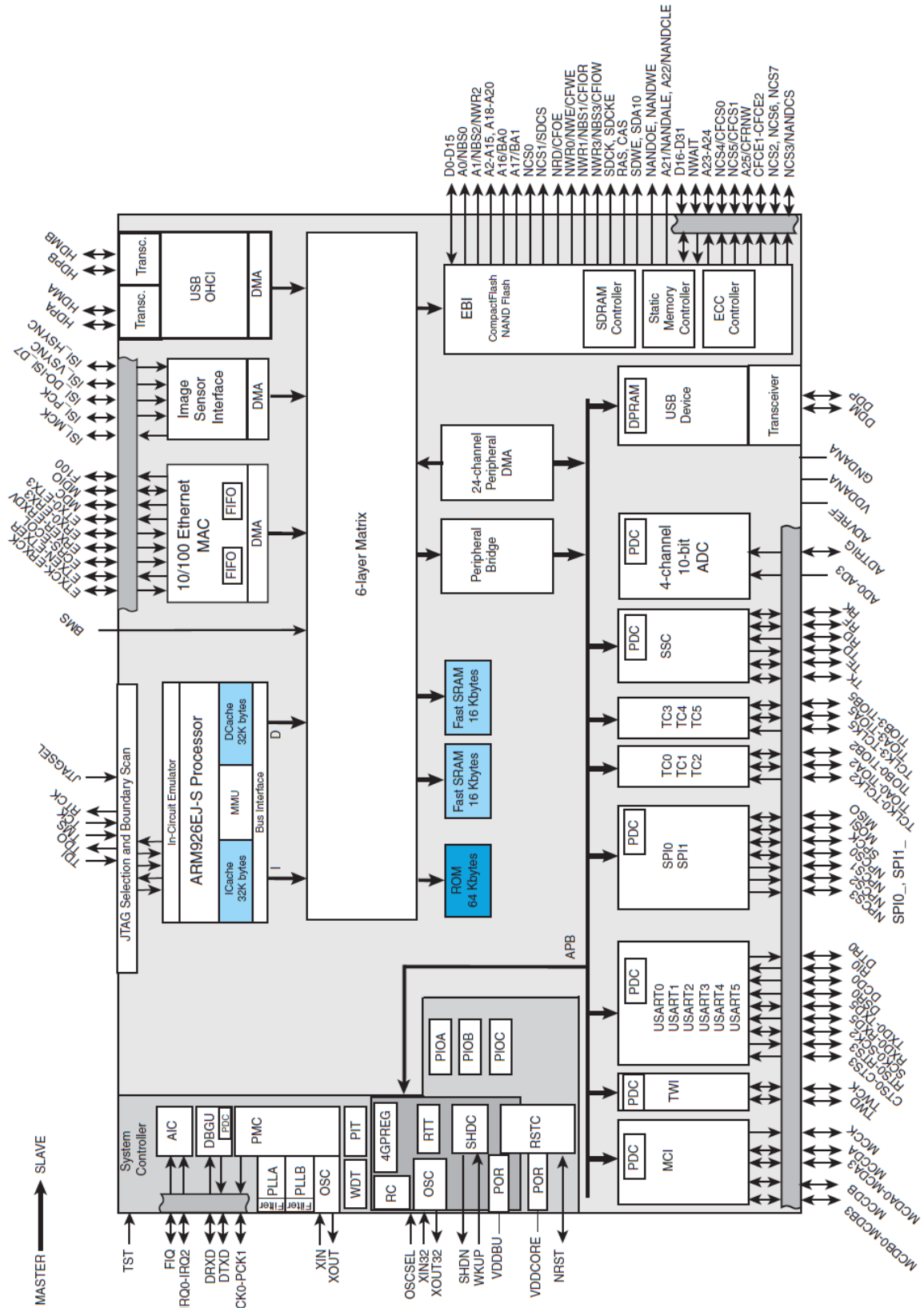


Figure 3.1: AT91SAM9G20 Block Diagram

3.1.2 Embedded modules

3.1.2.1 The module OC8-S

The processor introduced in 3.1.1 is plugged into the board OC8-S. With the evaluation board, it is able to set up a basic embedded environment which can be acted upon by a connection to a personal computer. Figure 3.2 shows the module OC8-S.



Figure 3.2: The embedded processor module Linux systems OC8-S

The evaluation board according Figure 3.3 comprises an RJ45 Ethernet plug, a JTAG plug. There is no stack-up connector for add-onboards, the hardware user manual and schematic are available from [8].

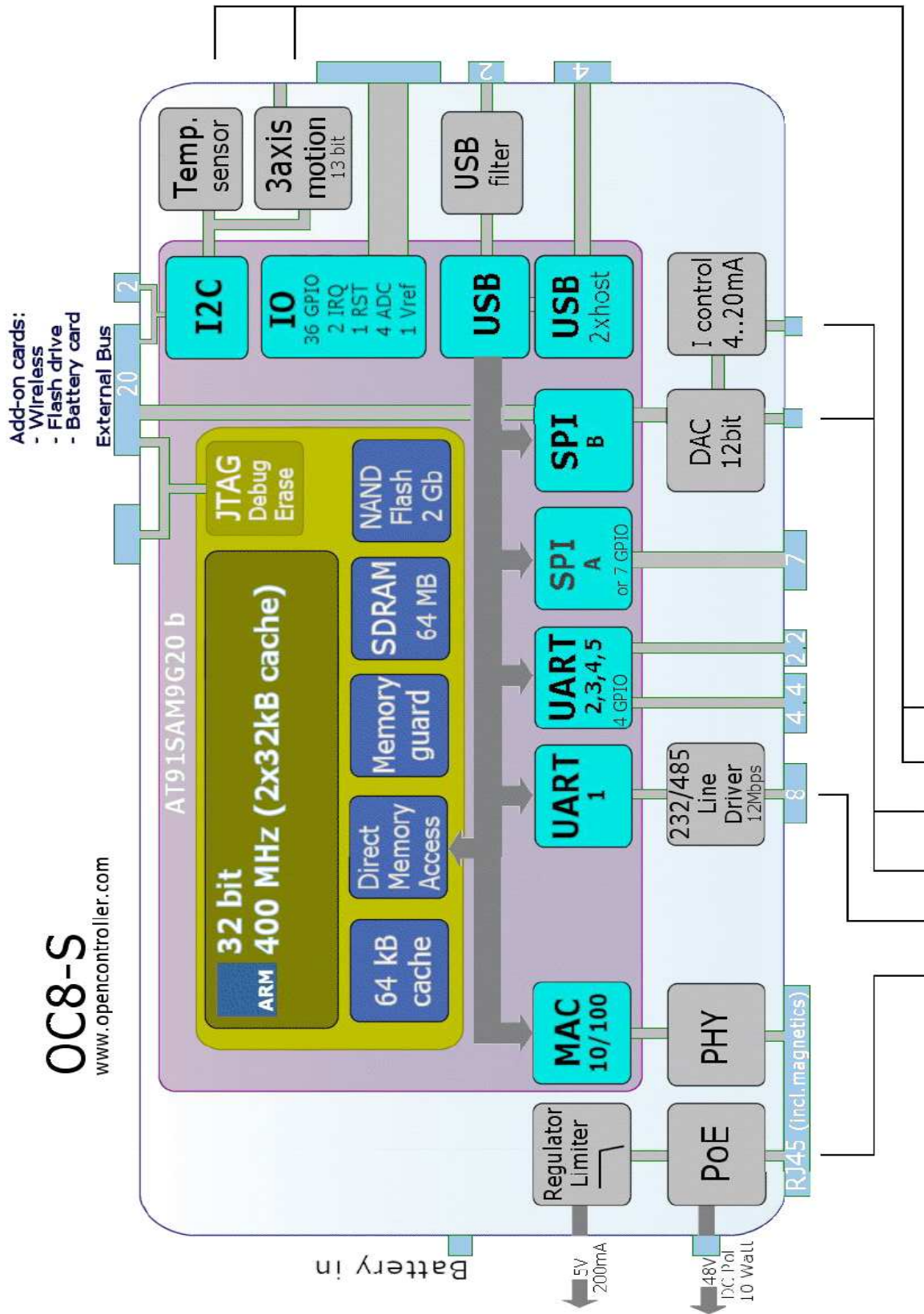


Figure 3.3: Hardware architecture of the module OC8-S

3.1.2.2 The OC8-H header board

The OC8-H is a header board and it is compatible to OC8-S. The USB FTDI-2232 device inside the OC8-H has 2 ports. One for JTAG and one for serial. Place the module OC8-S on the header board. Other products can have the USB-JTAG port integrated. The hardware user manual and schematic are available from [9]. Figure 3.4 shows a picture of the OC8-H.

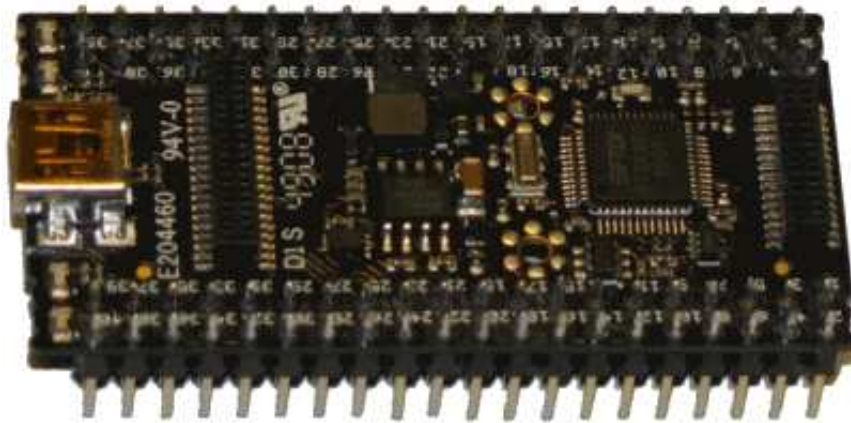


Figure 3.4: The OC8-H header board

3.2 Software issues for OC8-S

In the previous section a embedded processor was selected. This enables the deployment of a kernel or operating system in this project. Because the prototype audio class-D firmware contains several independent software components (USART driver, Fast Fourier Transform, Optimal PWM algorithm, etc.), it is necessary to build upon a kernel which offers basic multitasking functionality. To come to a decision the embedded Linux distribution is used here.

An embedded Linux distribution, which comprises the Linux kernel and GNU software/tools. An embedded Linux is a Linux derivative which is adapted to the needs of embedded microprocessors [10]. A port to the OC8-S architecture is available, including the GNU Compiler Collection (GCC) toolchain common in the Linux world. In this project, I use GNU Toolchain for ARM processors [11] as cross-compiler to compile software applications (Fast Fourier Transform, Optimal PWM algorithm) onto OC8-S.

The advantages of this solution are first that both an embedded Linux kernel and GNU software are open source software [12], and second that Linux is a familiar computing environment whose availability on embedded systems makes it easy to build an embedded application.

Chapter 4

Starting with an embedded Linux

This chapter introduces an embedded Linux operating system and covers how it is organized. Thereafter, this chapter presents the outline of embedded development environment and hosting target boards. In addition to the Linux, the components that are required for a complete embedded Linux operating system:

- GNU/Linux ARM cross-compiler toolchain for an embedded Linux and software applications.
- Bootloader ported to and configured for hardware platform.
- The Linux kernel source tree enabled for particular processor and board.

4.1 Overview

An embedded Linux is Linux operating system for embedded microtroller, short micro-controller Linux.

An embedded Linux distribution for embedded targets differs in several significant ways. First, the executable target binaries from an embedded distribution will not run on your PC, but are targeted to the architecture and processor of embedded system. A desktop Linux distribution tends to have many GUI tools aimed at the typical desktop user, such as fancy graphical clocks, calculators, personal time-management tools, email clients and more. An embedded Linux distribution typically omits these components in favor of specialized tools aimed at developers, such as memory analysis tools, remote debug facilities, and many more.

Another significant difference between desktop and an embedded Linux distributions is that an embedded distribution typically contains cross-tools, as opposed to native tools. For example, the GCC toolchain that ships with an embedded Linux distribution runs on the x86 desktop PC, but produces binary code that runs on the target boards. Many of the other tools in the toolchain are similarly configured: They run on the development host (usually an x86 PC) but operate on foreign architectures such as ARM or PowerPC [13].

Glibc

Another feature which makes Linux suitable for embedded devices is the use of the Glibc library. It is a C library for embedded Linux and Glibc is the one true C library in the GNU system, and in most newer systems with the Linux kernel. Glibc is a powerful set of shared libraries that is used on hundreds of thousands of computer systems all over the world. Like GCC, Glibc is a living testimonial to the power of open source software and the insight and philanthropy of its designers and contributors [14].

Linux distribution

Development of the Linux kernel for embedded devices tends to be split according to the processor architecture involved. For example, Russell King leads a group of developers who actively port Linux to ARM-based devices [15].

An embedded Linux includes not only the kernel itself, but a huge collection of GNU tools and programs commonly available in the Linux world. A rather complete list is available at [16]. The following is a briefly list of the main menu options available to all embedded Linux architectures:

- **Networking** : Many protocols are supported at client and/or server side: tftp (TFTP client), portmap (port to RPC3 program number mapper, used also for mounting of NFSs), ifconfig (network interface configuration), dhcpcd (DHCP client), etc
- **System tools**: In an embedded Linux, all of typical Linux commands (for file manipulation, kernel control, user management, etc.) are also accomplished with BusyBox. BusyBox is "The Swiss Army Knife of Embedded Linux." [17]. This is a fitting description, for BusyBox is a small and efficient replacement for a large collection of standard Linux command line utilities. It often serves as the foundation for a resource-limited embedded platform. BusyBox is modular and highly

configurable, and can be tailored to suit your particular requirements. The package includes a configuration utility similar to that used to configure the Linux kernel and will, therefore, seem quite familiar. For more information is available at [18].

4.2 Configuring the software environment

Getting ready for an embedded Linux project is a straightforward process. We need to collect the tools and install the necessary software components. A typical environment for an embedded Linux basically consists of the following elements :

- A development host: This is essentially a Linux box, where software for the target device is developed and cross-compiled. In this project, a PC with the Linux operating system is used. Naturally, it is equipped with an Ethernet card and some USB ports.
- The embedded target device. Here, development was started just with the evaluation board and the plugged-in core module. Further components were added as needed. Connections between development host and target device: They are used to load software onto the target boards and interact with programs running on the target boards.
- With this project, a serial connection is used over a USB cable with JTAG plug, and RJ45 Ethernet is used as well as an Ethernet connection via the local network.

Figure 4.1 shows the layout of a typical cross-development environment which was used for the Prototype Audio Amplifiers Class-D. A host PC is connected to the target board OC8-S via one or more physical connections. It is most convenient if both serial and Ethernet ports are available on the target.

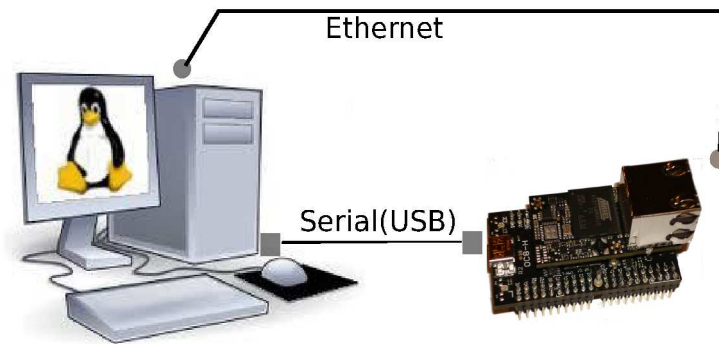


Figure 4.1: Cross-development setup

4.2.1 Hosting Target Boards

Linux terminal

Linux offers a terminal on the OC8-S's second UART which is the main communication facility at beginning of development (later, e.g. Telnet may be used). As mentioned above, a serial connection from the workstation to the AT91SAM9G20 processor is made via a USB cable. When the physical connection is made, a Linux workstation detects and creates the device file `/dev/ttyUSB1` or a similar one.

Display the available serial ports after connecting an OC8-H with a module on it:

```
$ ls -la /dev/ttyUSB*
```

```
crw-rw---- 1 root dialout 188, 0 Jul 26 10:35 /dev/ttyUSB0
crw-rw---- 1 root dialout 188, 1 Jul 26 10:35 /dev/ttyUSB1
```

Table 4.1: Display all serial ports

Get detailed info about the attached serial devices on the USB port:

```
$ sudo cat /proc/tty/driver/usbserial
```

```

usbserinfo:1.0 driver:2.0
0: module:ftdi_sio name:"FTDI USB Serial Device" vendor:0403 product
  :6010 num_ports:1
port:1 path:usb-0000:00:1a.1-2
1: module:ftdi_sio name:"FTDI USB Serial Device" vendor:0403 product
  :6010 num_ports:1
port:1 path:usb-0000:00:1a.1-2

```

Table 4.2: Detailed info about setial devices on the USB ports

The number in the first column resembles the `ttyUSB#`. The first serial USB port of each Header board is the JTAG port and the 2nd serial USB port is the debug serial port. Normally the debug port of the first board will be assigned to `ttyUSB1`, but this can change as modules get disconnected and reconnected on the USB port.

Table 4.3 shows the serial setup to connect to this serial debug port used `minicom` (terminal application) :

```
$ minicom
```

```

Serial Device          : /dev/ttyUSB1
Lockfile Location     : /var/lock
Callin Program        :
Callout Program       :
Bps/Par/Bits          : 115200 8N1
Hardware Flow Control : No
Software Flow Control : No

```

Table 4.3: Minicom terminal

TFTP Server

Table 4.4 contains a TFTP configuration from a Ubuntu development workstation to enable the TFTP service.

```
$ vi /etc/xinetd.d/tftp
```

```

service tftp {
    protocol = udp
    port      = 69
    socket_type = dgram
    wait      = yes
    user      = root
    server    = /usr/sbin/in.tftpd
    server_args = -s /tftpboot
    disable   = no
}

```

Table 4.4: TFTP Configuration

NFS Server

Table 4.5 contains illustrates the configuration options for NFS in the kernel.

\$ vi /etc/exports

```

/home/nhq    &192.168.8.8(rw, sync, no_subtree_check, no_root_squash)

```

Table 4.5: NFS Server Configuration

This denotes that the directory is exported to target board with IP address 192.168.8.8 and both read and write access is granted (**rw**).

4.3 The GNU Toolchain

The software for the target device, i.e. U-Boot and embedded Linux, will be compiled on the development workstation. Due to the fact that the workstation and the target have different processor architectures, software applications for the OC8-S must be cross-compiled. Therefore, a dedicated toolchain is required on the workstation. The freely available GNU toolchain is chosen here because it is provided with Linux and tightly integrated, and besides that it is the most common one in the Linux world.

The GNU toolchain is a group of related projects: a compiler, libraries, linker, utilities, and a debugger:

- **GCC:** The GCC (GNU Compiler Collection) is a set of several compilers for different programming languages (C/C++, etc.).
- **Binutils:** The GNU Binutils (binary utilities) are a collection of binary tools and provide low-level handling of binary files, such as linking, assembling, and parsing ELF files. The GCC compiler depends on these tools to create an executable, because it generates object files that binutils assemble into an executable image.
- **Debugger:** The GNU debugger gdb is a symbolic debugger and is the most important debugging tool for any Linux system

Cross-Compiler

A cross-compiler is a tool that transforms source code into object code that will run on a machine other than the one where the compilation was executed. When we are working with languages that execute on virtual machines (like Java), all compilation is cross-compilation: the machine where the compilation runs is always different than the machine running the code. The concept is simple in that when the compiler generates the machine code what will eventually be executed, that code won't run on the machine that's doing the generating.

Some basic terminology is used to describe the players in the process of building the compiler:

- **Build machine:** The computer used to compile the code.
- **Host machine:** The computer where the compiler runs.
- **Target machine:** The computer for which GCC produces code.

For more information at [10].

There are many possibilities to get cross-compiler toolchain. Two possibilities exist for installing the toolchain: First, a pre-compiled toolchain can be downloaded. This is the fastest method, since no compiling is necessary. Second, the source code of a toolchain can be downloaded and compiled by oneself. In this case, we need build the supporting binutils, then a cross-compiler suitable for compiling glibc, and then the final compiler. For the purpose of illustration, the steps are broken out into several sections. In a real project, all the steps are combined into a script that can be run without intervention.

Getting toolchain

CodeSourcery Sourcery G++ Lite toolchain for ARM GNU/Linux EABI processors is used for this project. The binary distribution for 2010q1 version is available at [11].

The downloaded file `arm-2010q1-202-arm-none-linux-gnueabi-i686-pc-linux-gnu.tar.bz2` has to be unpacked using the command:

```
tar -xvjf arm-2010q1-202-arm-none-linux-gnueabi-i686-pc-linux-gnu.tar
.bz2
```

Table 4.6: Unpacking toolchain

The file is extracted to directory `/embedded/arm/install/cross-arm/`. After unpacking of the CodeSourcery G++ Lite toolchain, the `PATH` environment variable on the workstation has to be modified to include the toolchain executables so that these can be found independently of the working directory. This is best done by appending the following line to the `.bashrc` file in the home directory:

```
$ vi /home/nhq/.bashrc
```

```
export PATH= $PATH:/embedded/arm/install/cross-arm/arm-gnueabi-gcc/arm
-2010q1/bin
```

Table 4.7: Configuration address for CodeSourcery in workstation.

4.4 Bootloader

A boot loader isn't unique to Linux or embedded systems. It's a program first run by a computer so that a more sophisticated program can be loaded next. The need for a bootloader is caused by the fact that most processors can only execute code from predetermined sources at startup, e.g. from memory. To enhance boot methods, a boot loader is needed that itself lives in the ROM (usually Flash) memory of the target and provides more sophisticated functionality [18].

4.4.1 A Universal Bootloader: Das U-Boot

The official name for this bootloader is Das U-Boot. It is maintained by Wolfgang Denk and hosted on SourceForge at [19]. U-Boot has support for multiple architectures and has a large following of embedded developers and hardware manufacturers who have adopted it for use in their projects and have contributed to its development.

The following is a briefly list of U-Boot's functionality:

- Command line : U-Boot provides a command line to the user. Many commands are available for booting, memory programming and examination, network configuration, etc. The command list can be retrieved by entering help at the command line.
- Loading files : Several loading commands allow for different retrieval of (image) files. With this project tftp (for loading a file from a TFTP server) and nfs-server (for making storage storage location (the so-called *export*) available to other hosts on the network) are most often used.
- Booting : Several commands support booting of different images. For example:
 - **bootm** is used to boot compressed an embedded Linux images (out of RAM or ROM), whereas bootelf boots uncompressed ELF images which are usually stored in RAM due to their size.
 - **bootp** command issues a request that is answered by the DHCP server. Using the DHCP server's answer, U-Boot contacts the TFTP server and obtains the Linux kernel image file, which it places at the configured load address in the target RAM.
- Networking : U-Boot contains drivers for network devices, among others for the on-chip Ethernet MAC of the AT91SAM9G20 processor. It supports common protocols like TFTP and DHCP. Configuration of the Ethernet MAC (media access control) address is also done via U-Boot.
- Flash programming : U-Boot is the first choice for writing application images to flash memor.
- Environment variables : These variables contain customizable information for the target hardware, like IP address, Ethernet MAC address, etc. We can use the commands:

- `printenv` to display all environment variables,
- `setenv` to add new variables,
- `saveenv` to save new variables to flash memory.

4.4.2 Building U-boot

Compiling the U-Boot is rather simple. Optionally, configuration of U-Boot can be customized prior by editing the file `u-boot-2009.11/include/configs/oc8s.h`. All default environment variables are defined therein and can be changed, but this is done more conveniently with the U-Boot command line. Important options can only be changed before compiling: `AT91_MAIN_CLOCK`, `CONFIG_SYS_HZ`, and `AT91_SPI_CLK`, which configure the master clock and the CPU clock. With the default values a CPU clock of 400 MHz and a master clock of 133 MHz are set. For more information [7].

Configuration

To configure the U-Boot source code for the OC8-S module, command:

```
make oc8s_config
```

Table 4.8: Create default configuration OC8-S

By default the build is performed locally and the objects are saved in the source directory. One of the two methods can be used to change this behavior and build U-Boot to some external directory:

```

1. Add O= to the make command line invocations:
    make O=/tmp/build distclean
    make O=/tmp/build NAME_config
    make O=/tmp/build all
2. Set environment variable BUILD_DIR to point to the desired
   location:
    export BUILD_DIR=/tmp/build
    make distclean
    make NAME_config
    make all
Note that the command line "O=" setting overrides the BUILD_DIR
environment variable and "NAME" is target board(OC8S)

```

Table 4.9: Configuration U-Boot.

Cross-compiling

```

$ make CROSS_COMPILE=/opt/codesourcery/bin/arm-none-linux-gnueabi-gcc
After a successful compilation the following binaries will be
available:
(BOARD)-u-boot-2009.11.bin      - U-Boot binary
e.g. oc8s-u-boot-2009.11.bin
(BOARD)-u-boot-env-2009.11.bin  - U-Boot environment image
e.g. oc8s-u-boot-env-2009.11.bin

```

Table 4.10: Cross-compiling.

4.4.3 Downloading the U-Boot onto OC8-S

The JTAG device was described in section 3.1.2. In this project, it is the only possibility to download U-Boot's image do OC8-S. Therefore, a JTAG device was connected to the evaluation board and it was used to download U-Boot's image into flash memory.

Hitting any key stops the autoboot, U-Boot displays information such as at 4.11

```
U-Boot 2009.11 (Jul 21 2010 - 13:44:04)
CPU: AT91SAM9G20
Crystal frequency:      20 MHz
CPU clock               :    400 MHz
Master clock           : 133.333 MHz
DRAM                   :   64 MB
NAND                   :  256 MiB
In                     :   serial
Out                    :   serial
Err                    :   serial
Net                    :   macb0
macb0                  : link up, 100Mbps full-duplex (lpa: 0xcde1)
Hit Enter to stop autoboot:  1
U-Boot >
```

Table 4.11: Das U-Boot

4.4.4 Important routines

Network settings

The first order of business in enabling most network services (the DHCP client being the major exception) is the correct configuration of network settings. At minimum, this includes the target IP address and routing table; if the target will use DNS, a domain name server IP address needs to be configured. The following commands are entered at the U-Boot command line:

```
U-Boot> setenv gatewayip=192.168.8.1
U-Boot> setenv netmask=255.255.255.0
U-Boot> setenv ipaddr=192.168.8.8
U-Boot> setenv serverip=192.168.8.1
```

Table 4.12: U-Boot's network settings

Then command `saveenv` to save all variables.

Loading a file onto the target

We can use TFTP server on the development host to load file to the OC8-S. Transfers are fast and simple . Before the loading, the file must exist in the `/tftpboot/` directory on the workstation :

```
U-Boot> tftp 0x22000000 uImage
```

Table 4.13: U-Boot's network settings

The uImage is downloaded to RAM address 0x22000000.

Writing a Linux image to the Flash memory

After downloading the uImage do RAM at address 0x22000000, I need to write it to the Flash memory. The reason is simple: During development, it makes no sense to write each compilation of an embedded Linux into flash memory. Usually, these images are transferred to the target over the network and bootet from RAM. However, this project's goal is to develop a stand-alone device, and hence the uImage gets programmed into flash memory eventually. Only compressed images of an embedded Linux (usually named uImage) are small enough to fit into flash memory.

Before the uImage file can be written to the Flash memory, affected sectors must be erased first:

```
Boot> nand erase 0x00200000 0x200000
```

Table 4.14: Erasing the flash memory

Now, we can write the uImage from RAM to the Flash memory at address 0x00200000.

```
Boot> nand write 0x22000000 0x00200000 0x200000
```

Table 4.15: Writting uImage to the Flash memory

The next step is to download the valid root file system image (.jffs2) to RAM at address 0x21000000.

```
Boot> tftp 0x21000000 myrmica-minimal-oc8.jffs2
```

Table 4.16: Downloading the root file system

Then such as a previous step, I must to write the root file system to the Flash memory:

```
Boot> nand write.jffs2 ${fileaddr} 0x00400000 ${filesize}
```

Table 4.17: Writting the root file system to the Flash memory

Now I need to change environment variables such that, the boot command will read the uImage from 0x00200000 into the ram and will boot.

```
Boot> setenv bootcmd = nand read 0x23d00000 0x00200000 0x200000  
Boot> savenv bootcmd
```

Table 4.18: Reading of the uImage to the RAM

The setenv command will set the boot command such that the uImage presents in the Flash memory a address 0x00200000 will be loaded into the RAM at address 0x23d00000.

Booting an embedded Linux image

The boot parameters for the Linux kernel can be set in U-Boot with the *bootargs* variable, *printenv* is used to print the urrent content of environment variables.

```
bootargs=mem=64M console=ttyS0,115200 root=/dev/mtdblock1 rw rootfstype  
=jffs2  
ip=192.168.8.8:192.168.8.1:f
```

Table 4.19: The boot parameter

To boot a compressed *ulmage*, use *bootm* :

```
U-Boot> bootm 23d00000
## Booting kernel from Legacy Image at 23d00000 ...
Image Name:   Linux-2.6.34
Image Type:   ARM Linux Kernel Image (uncompressed)
Data Size:    1581568 Bytes = 1.5 MB
Load Address: 20008000
Entry Point:  20008000
Verifying Checksum ... OK
Loading Kernel Image ... OK
OK

Starting kernel ...
```

Table 4.20: Boot uImage

Now, board will boot from the uImage at 0x23d00000.

4.5 Linux distribution

In the previous sections the toolchain was installed and the U-Boot bootloader was brought onto the target board. Now it is time to attend to an embedded Linux itself. Since an introduction was already provided in section 4.1, this one concentrates on practical aspects.

4.5.1 Getting an embedded Linux

In this project, the Linux version 2.6.34 is used to build an embedded Linux for OC8-S. We can get it from internet or at [20].

Getting the kernel source

If you install the full sources, put the kernel tarball in a directory where you have permissions (eg. your home directory) and unpack it:


```
gzip -cd linux-2.6.34.tar.gz | tar xvf -  
  
or  
  
bzip2 -dc linux-2.6.34.tar.bz2 | tar xvf -
```

Table 4.21: Getting the kernel source

Configuration

It is very simple to configure and compilation of the Linux kernel for OC8-S.

```
$ make oc8s_defconfig
```

Table 4.22: Configuration of the Linux kernel

In order to create kernel with U-Boot information (`ulmage`). First, it need to make sure if we have program `mkimage` installed. The program `mkimage` is compiled along with U-Boot bootloader and can be found in `u-boot/tools/mkimage` 4.4.2 Compile the Kernel image prepared for the U-Boot loader:

```
$ make uImage
```

Table 4.23: Creating the Linux kernel

The image will be compiled in: `arch/arm/boot/ulmage`. The next step is to copy one of these into the `/tftpboot/` directory to be accessible via TFTP.

Now the image can be loaded onto the target and booted with U-Boot. This happens automatically with the U-Boot setup from section 4.4.4 . So, after hitting the reset button on the evaluation board, an embedded Linux boots up and displays a lot of information in doing so.

4.5.2 Adding new drivers and application

Device drivers

To add a new device driver, three steps have to be accomplished:

1. The source code file(s) have to be created. This/These will be written in the C language. (In this example, `add_driver_file.c` is used). Set PATH environment variable to include the toolchain executables in the `/.bashrc` file in the home directory:

```
$ export LINUX_SOURCES=/embedded/arm/152-linux-2.6.34
```

Table 4.24: Set PATH environment for compiling

2. Start compilation `add_driver_file.c` with command:

```
$ make
```

Table 4.25: Creating the Linux kernel

3. Now we have binary file `add_driver_file.ko`, copy it to the target and load it to the kernel.

```
$ insmod add_driver_file.ko
```

Table 4.26: Loading the device driver to kernel

The kernel `add_driver_file.ko` will be loaded into the `/dev/` on the OC8-S.

Applications

Adding a user application involves the following steps:

1. The source code file(s) have to be created. This/These will be written in the C language. (In this example, `add_app_file.c` is used). The PATH environment variable to include the toolchain executables was set in the `/.bashrc` file in the home directory 4.3 and 4.7.

2. Start compilation `add_app_file.c` with command:

```
$ make
```

Table 4.27: Creating a user application

3. Now we have binary file `add_driver_file-arm`, it can be loaded onto OC8-S with using NFS server or TFTP client 4.2.1.

```
$ cp /home/nhq/ad_app_file-arm /opt/app/
```

Table 4.28: Loading application onto OC8-S

The `add_app_file-arm` will be loaded into the `/opt/app/` on the OC8-S.

Chapter 5

Software

It's almost time to begin programming. This is the main chapter of this document because it describes the software development phase. The following software components must be developed: The device driver for USART to transfer switching times α to pinout of OC8-H [9]; the latter is the FFT 2.2.1 and Optimal PWM algorithm 2.2.2. As usual with Linux development, the C programming language is used exclusively.

5.1 USART device driver

In this project, the time is the deciding variable to develop the ambedded audio amplifier class-D. Switching times α generated with Optimal PWM algorithm will be stransfered to pinout on the OC8-H to generate the PWM signal. Switching times are determined in microseconds, so it requires a hight frequency resolution. One of the solutions to solve this problem is using USART to receive switching times and transfer them to pinout on OC8-H.

5.1.1 Overview

The Universal Synchronous Asynchronous Receiver Transceiver (USART) provides one full duplex universal synchronous asynchronous serial link. Data frame format is widely programmable (data length, parity, number of stop bits) to support a maximum of standards. The receiver implements parity error, framing error and overrun error detection. The receiver time-out enables handling variable-length frames and the transmitter time-

guard facilitates communications with slow remote devices. Multidrop communications are also supported through address bit handling in reception and transmission. The USART features three test modes: remote loopback, local loopback and automatic echo

The USART supports the connection to the Peripheral DMA (Direct Memory Access) Controller, which enables data transfers to the transmitter and from the receiver. The PDC provides chained buffer management without any intervention of the processor [7].

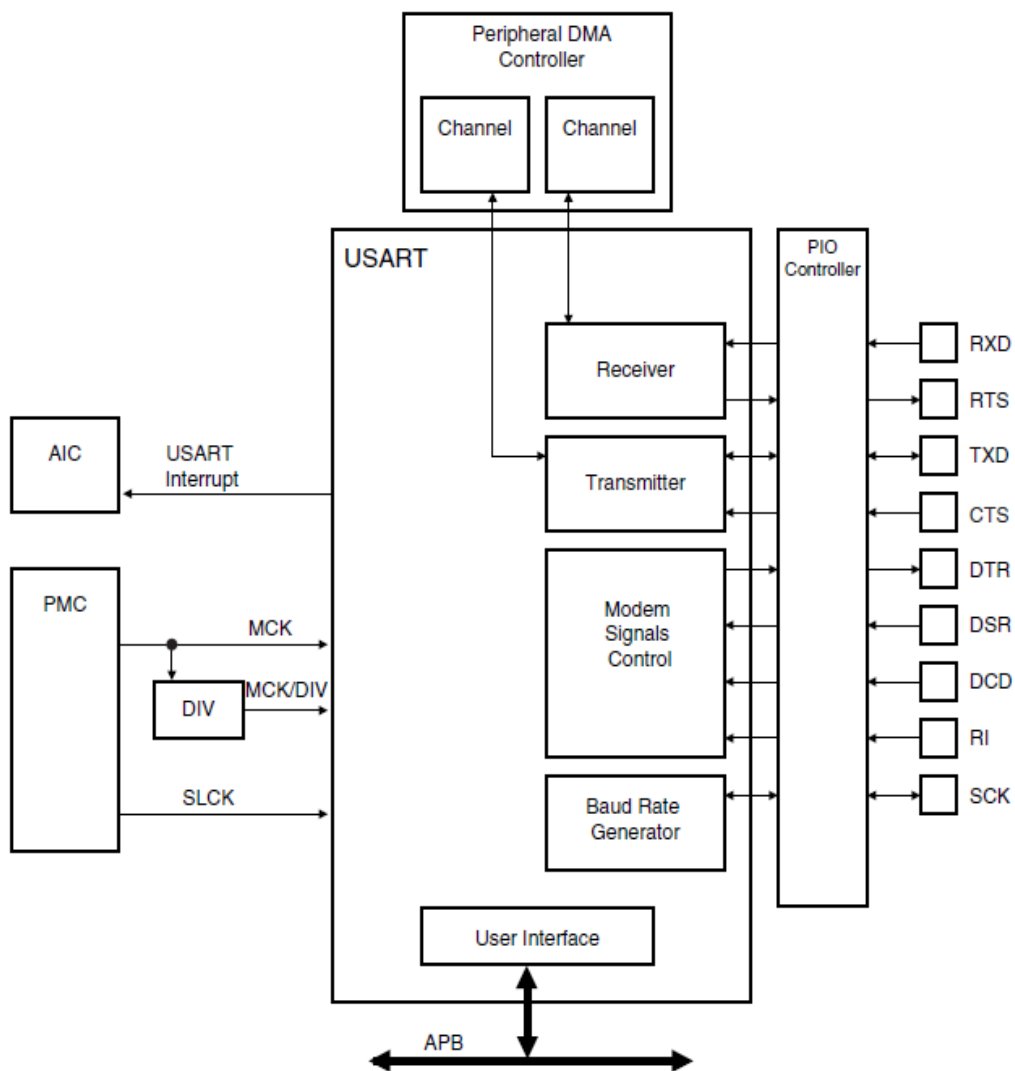


Figure 5.1: The block diagram of USART

5.1.2 Implementation of USART driver

In the source file of USART driver is main file `usart-pwm.c`.

usart-pwm.c

There are defined default module parameters:

```
static int xtal = 400000000 / 3; /* clock (Hz) */
static int baud = 1000000; /* baud rate */
static int parity = GPIO_PWM_PAR_NONE; /* parity */
static int data = 8; /* data length */
static int stop = GPIO_PWM_STOP_1; /* stop bit length */
static int type = GPIO_PWM_TYPE_RS485; /* RS-232/RS-485/RS-422 */
static int major = 0; /* char device major number */
static int minor = 0; /* char device minor number */
static int count = 1; /* char device amount *
```

Table 5.1: Define module parameters

The static variable `xtal` is maximum of the master clock (133.33MHz) [7]. Variable `baud` is the speed that data transmitted per second. In this case it is 1000000bit/s, that means it can transmit 1bit in $\doteq 1(\mu s)$. The next variable is `GPIO_PWM_TYPE_RS485`, it defines the type of serial interface. Global variables are declared as static, so are global within the file.

There I define the GPIO pinout on the OC8-H.

```
#define LED3 AT91_PIN_PB31
```

Table 5.2: Define pinout

This pinout is connected to *LED3* on the OC8-H, so the LED3 will show when the PWM signal is generated.

A program usually begins with a `main()` function, executes a bunch of instructions and terminates upon completion of those instructions. Kernel modules work a bit differently. Kernel modules defines: a "start" (initialization) function which is called `init_module()` when the module is insmoded into the kernel, and an "end" (cleanup) function which is called `cleanup_module()` just before it is rmmode. In this case, we specify two functions: `_init gpio_pwm_mod_init(void)` and `_exit gpio_pwm_mod_exit(void)`. There are called in the `module_init()` and `module_exit()` macros. These macros are defined in `<linux/init.h>`. The

only caveat is that `init` and `cleanup` functions must be defined before calling the macros, otherwise it'll get compilation errors.

Furthermore, for peripheral initialization the function `gpio_pwm_pin_init()` is called.

```
static void gpio_pwm_pin_init(struct gpio_pwm_priv *priv)
{
    at91_set_B_periph(AT91_PIN_PB4, 0); /* TXD0 */
    at91_set_B_periph(AT91_PIN_PB5, 0); /* RXD0 */
    at91_set_B_periph(AT91_PIN_PB26, 0); /* RTS0 */
    at91_set_B_periph(AT91_PIN_PB27, 0); /* CTS0 */
    at91_set_B_periph(AT91_PIN_PB22, 0); /* DSR0 */
    at91_set_B_periph(AT91_PIN_PB24, 0); /* DTR0 */
    if (type == GPIO_PWM_TYPE_RS485)
        at91_set_gpio_output(AT91_PIN_PB11, 1); /* RS-485 mode */
    else
        at91_set_gpio_output(AT91_PIN_PB11, 0); /* RS-232 mode */
}
```

Table 5.3: Peripheral initialization

Most of the fundamental driver operations involve three important kernel data structures, called `file_operations`, `file`, and `inode`. They are defined in `<linux/fs.h>`.

- `file_operations` structure: holds pointers to functions defined by the driver that perform various operations on the device. Each field of the structure corresponds to the address of some function defined by the driver to handle a requested operation.

In this case, the `file_operations` structure is initialized as follows:

```
static struct file_operations gpio_pwm_fops = {
    .read    = gpio_pwm_read,
    .write   = gpio_pwm_write,
    .open    = gpio_pwm_open,
    .release = gpio_pwm_close,
};
```

Table 5.4: File_operations structure

- `struct file`: is the second most important data structure used in device drivers. Note that a `file` has nothing to do with the `FILE` pointers of user-space programs. A

FILE is defined in the C library and never appears in kernel code. A file structure, on the other hand, is a kernel structure that never appears in user programs.

The file structure represents functions `gpio_pwm_open()` and `gpio_pwm_close()`.

```
static int gpio_pwm_open(struct inode *inode, struct file *file)
{
    try_module_get(THIS_MODULE);
}

static int gpio_pwm_close(struct inode *inode, struct file *file)
{
    module_put(THIS_MODULE);
}
```

Table 5.5: File structure

where,

- `try_module_get(THIS_MODULE)`: Increment the use count.
- `module_put(THIS_MODULE)`: Decrement the use count.

These manipulate the module usage count, to protect against removal. Before calling into module code, we should call `try_module_get()` on that module: if it fails, then the module is being removed and you should act as if it wasn't there. Otherwise, you can safely enter the module, and call `module_put()` when you're finished. The macro `THIS_MODULE` is defined in `<linux/module.h>`

Next two function are defined: `gpio_pwm_read()` and `gpio_pwm_write()`.

- `gpio_pwm_read()`: The function defines reading data from the serial device. It has four arguments: `struct file *file`, `char *buf`, `size_t len` and `loff_t *off`. First argument is pointer to `struct file` defined in `<linux/fs.h>`. It is the most important data structure used in device drivers. Argument `len` is the size of the requested data transfer. The `buf` argument points to the empty buffer where the newly read data should be placed. Finally, `off` is a pointer to a "long offset type" object that indicates the file position the user is accessing. The return value is a "static int type".

- `gpio_pwm_write()`: This function has arguments like as function `gpio_pwm_read()`, it gets requests from user space and put it into transmit serial data queue. it writes up to `len` bytes from the buffer starting at `buf` to the file pointer `file` at offset `off`.

```

static int gpio_pwm_write(struct file *file, const char *buf,
    size_t len, loff_t *off)
{
    mutex_lock(&gpio_pwm_priv->mtx);
    head = gpio_pwm_priv->pwm.head;
    tail = gpio_pwm_priv->pwm.tail;
    space = CIRC_SPACE(head, tail, BUFSIZE);
    if ((space) > 0 && (len <= space)) {
        space_to_end = CIRC_SPACE_TO_END(head, tail, BUFSIZE);
        /* non-wrapping copy */
        if (len <= space_to_end) {
            rv = copy_from_user(&gpio_pwm_priv->pwm.buf[head],
                buf, MIN(space, len));
            if (rv)
                goto out;
            /* wrapping copy */
        } else {
            rv = copy_from_user(&gpio_pwm_priv->pwm.buf[head],
                buf, space_to_end);
            if (rv)
                goto out;
            rv = copy_from_user(&gpio_pwm_priv->pwm.buf[0],
                &buf[space_to_end],
                len - space_to_end);
            if (rv)
                goto out;
        }
        gpio_pwm_priv->pwm.head = (head + len) & (BUFSIZE - 1);
        size = len;
    }
out:
    mutex_unlock(&gpio_pwm_priv->mtx);
}

```

Table 5.6: Using mutex

In this function, I used `mutex_lock` and `mutex_unlock` to control a queue of data to read/write to circular buffer. The principle of function `gpio_pwm_write()` is shown in Figure 5.2.

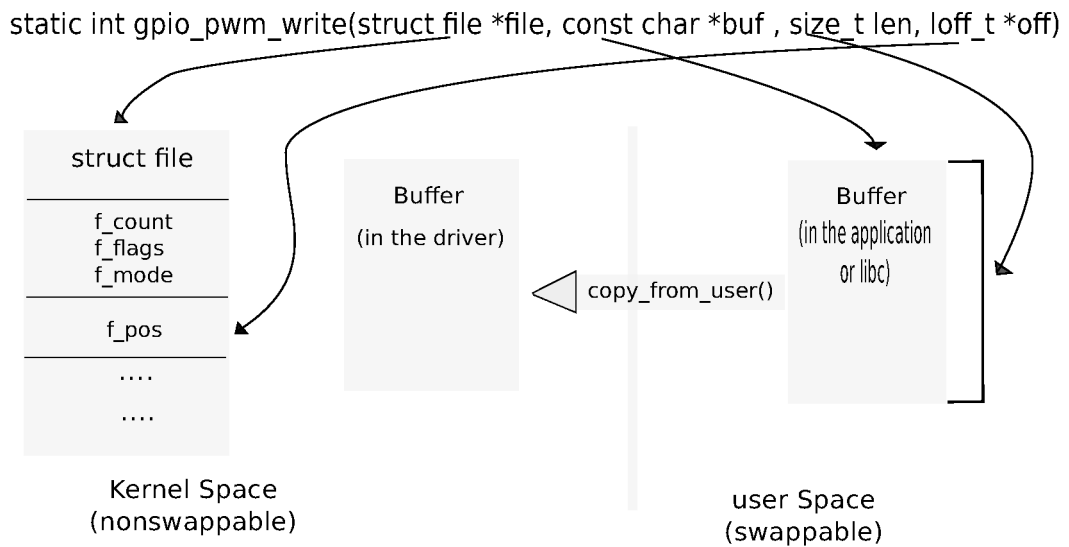


Figure 5.2: Principle of `gpio_pwm_write()`

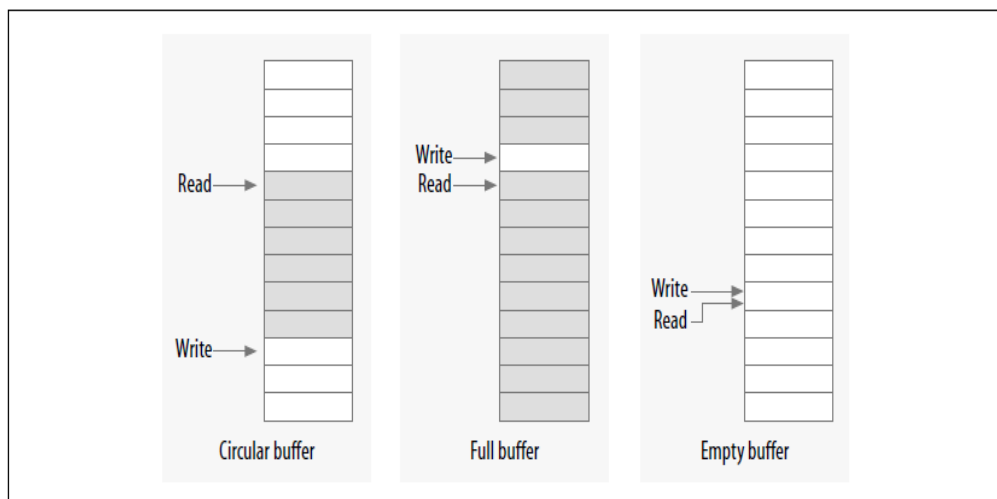


Figure 5.3: A circular buffer

The Figure 5.3 shows circular buffer in several states of fill. This buffer has been defined such that an empty condition is indicated by the read and write pointers be-

ing equal, while a full condition happens whenever the write pointer is immediately behind the read pointer.

- `irqreturn_t gpio_pwm_intr()`: This function handles the interrupt.

```

static irqreturn_t gpio_pwm_intr(int irq, void *data, struct pt_regs *regs)
{
    /* disable interrupts */
    gpio_pwm_outl(priv, ATMEL_US_IDR, ~0);
    . . .
    if (priv->toggle) {
        at91_set_gpio_value(LED3, 0);
    }
    else {
        at91_set_gpio_value(LED3, 1);
    }
    priv->toggle = !priv->toggle;
    . . .
    if (status & mask)
        goto int_pending;
    return IRQ_NONE;
int_pending:
    /* disable interrupts */
    gpio_pwm_outl(priv, ATMEL_US_IDR, ~0);
    . . .
    if (priv->toggle) {
        at91_set_gpio_value(LED3, 0);
    }
    else {
        at91_set_gpio_value(LED3, 1);
    }
    priv->toggle = !priv->toggle;
    /* TODO: use spin-lock */
    head = priv->pwm.head;
    tail = priv->pwm.tail;
    cnt = CIRC_CNT(head, tail, BUFSIZE);
    if (cnt > 0) {
        priv->pwm.tail = (tail + 1) & (BUFSIZE - 1);
        gpio_pwm_outl(gpio_pwm_priv, ATMEL_US_RTOR, priv->pwm.buf[tail]);
        gpio_pwm_outl(gpio_pwm_priv, ATMEL_US_IER, ATMEL_US_TIMEOUT);
        gpio_pwm_outl(gpio_pwm_priv, ATMEL_US_CR, ATMEL_US_RETTO);
    }
    /* TODO: use spin-unlock */
    return IRQ_HANDLED;
}

```

Table 5.7: Interrupt handler

When the data are copying from user space, the required user-space page may need to be swapped in from the disk before the copy can proceed, and that operation clearly requires a sleep. It takes long time when the process begins again. One of the solutions is to use the `spinlock` mechanism. Unlike semaphores or mutex, spinlocks are used in code that can not sleep, such as interrupt handlers. A spinlock is a mutual exclusion device that can have only two values: "locked" and "unlocked". Before taking a spinlock, interrupts were disabled by `gpio_pwm_outl`. While the lock is held, the device issues an interrupt, which causes the interrupt handler to run. The interrupt handler, before accessing the device, must also obtain the lock. Taking out a spinlock in an interrupt handler is a legitimate thing to do; that is one of the reasons that spinlock operations do not sleep. While the interrupt handler is spinning, the noninterrupt code will not be able to run to release the lock. That processor will spin forever. If there are data, the interrupt handler runs and the LED3 is disabled (0) else it is always enabled (1). Interrupt handlers should return a value indicating whether there was actually an interrupt to handle. If the handler found that its device did, indeed, need attention, it should return `IRQ_HANDLED`; otherwise the return value should be `IRQ_NONE`.

5.2 Fast Fourier Transform

In the section 2.2.1.2, the FFT was determined using the iterative implementation [21]. The source code for FFT is contained in the directory `/audio/FFT`. There one header file (`*.h`) and main file (`*.c`). In this project, the input for FFT are given and well-known discretely sampled data.

`fft.h`

This header file contains two functions:

- `bitrev()`: This function is used to compute bit-reversed permutation of the output arrays (see Figure 5.4). The corresponding source code is as follows:

```

unsigned int bitrev(unsigned int n, unsigned int bits)
{

for (n >>= 1; n; n >>= 1) {
    nrev <<= 1;
    nrev |= n & 1; /* give LSB of n to nrev */
    count--;
                }

}

```

Table 5.8: Bit-reversed function

This function has two arguments, the first one is a length of bits; the second argument is the number of bits that is used to compute bit-reversed.

Binary	index	1st split	2nd split	3rd split	Bit reversal
000	0	0	0	0	000
001	1	2	4	4	100
010	2	4	2	2	010
011	3	6	6	6	011
100	4	1	1	1	001
101	5	3	5	5	101
110	6	5	3	3	011
111	7	7	7	7	111

Figure 5.4: Bit reversal process in FFT

- `fft()` has two sections: The first section sorts the data into bit-reversed order. The second section has an outer loop that is executed $\log_2 N$ times and calculates, in turn, transforms of length 2, 4, 8, ..., N . For each stage of this process, two nested inner loops range over the subtransforms already computed and the elements of each transform, implementing the Danielson-Lanczos Lemma. The source code is defined as follow:

```

fft(double *b, double *a1, double *a2, int N, int sign)
{
    . . .
    /* reorder input and split input into real and complex parts */
    for (i=0; i<N; i++) {
        j = bitrev(i, log2n);
        *(a1+j) = *(b+2*i);
        *(a2+j) = *(b+2*i+1);
    }
    /* loop for FFT stages */
    for (s=1; s<=log2n; s++) {
        m = 1<<s; /* m = 2^s */
        wm1 = cos(sgn*2*PI/m); /* wm = exp(2*pi*i*k/m) */
        wm2 = sin(sgn*2*PI/m);
        w1 = 1;
        w2 = 0;
        for (j=0; j<m/2; j++) {
            for (k=j; k<N; k+=m) {
                /* t = w*a[k+m/2] */
                k2 = k + m/2;
                tmp1 = w1 * (*(a1+k2)) - w2 * (*(a2+k2));
                tmp2 = w1 * (*(a2+k2)) + w2 * (*(a1+k2));
                u1 = *(a1+k);
                u2 = *(a2+k);
                *(a1+k) = u1 + tmp1;
                *(a2+k) = u2 + tmp2;
                *(a1+k2) = u1 - tmp1;
                *(a2+k2) = u2 - tmp2;
            }
            /* w = w*wm */
            tmp1 = w1*wm1 - w2*wm2;
            w2 = w1*wm2 + w2*wm1;
            w1 = tmp1;
        }
    }
    . . .
}

```

Table 5.9: Fast Fourier Transform

The input quantities are the number of complex data points, it is an integer power of 2 ($N = 2^n$); *sign*, which should be set to either ± 1 , *sign* is 1 when we compute FFT and if *sign* is set to -1 when the routine thus calculates the inverse transform of FFT.

The real and imaginary parts of the frequency f are **a1* and **a2*. The frequency spectrum of FFT will return in **b*, where the length of **b* is 2 times **a1* (or **a2*). In other words, **(b + 1)* is the real part of F and **(b + 2)* is the imaginary part of F .

5.3 The optimal PWM

The implementation of the optimal PWM is according the optimal PWM algorithm of odd bi-level waveform 2.2.2.1. The source for the optimal PWM is contained in the directory `/audio/optimalpwm`. Main file is `optimalpwm.c` and there are five header files (`*.h`). Output of FFT are input for the optimal PWM algorithm, so they are frequency spectrums.

rhspowersum.h

This header file contains functions that compute the right hand side (RHS) of composite sum of powers p_i

- `binomic()`: This function computes the binomial coefficients using dynamic programming.
- `pEVEN()`: This function is used to compute the even i -th of p , p_{2i} .
- `pODD()`: This function is used to compute the odd i -th of p , p_{2i+1} .
- `pSUM()`: This function joins numbers of p_{2i} and p_{2i+1} .

moments.h

- `moment()`: This function computes the moments u_i , $i = 1, \dots, n$ according to the theory of Pade method; n are numbers of switching times. Input p_1, \dots, p_n are the right hand side of composite sum of powers.

pademethod.h

This header file is the "heart" of the optimal PWM algorithm. It contains the method to determine the switching times $\bar{\alpha} = (\alpha_1, \dots, \alpha_n)$.

- `hankel()`: This function defines hankel matrix created by moments u_i , where $i = 1, \dots, n$ are number of switching times
- `qrdec()` : This function compute QR decomposition. This is used to solve linear Hankel system. For more see A.1.
- `solvew()`: This is used to solve linear system $R * W = Q^T * B$, where $U = Q * R$, U is a upper matrix.
- `solvev()`: The function solves matrix equation with triangular hankel matrix.

rfpoly.h

This header file contains functions that are used Bairstow Method to find both the real and complex roots of a polynomial. Is is based on the idea of synthetic division of the given polynomial by a quadratic function. This approach can be used to find all the roots of a polynomial. For more information see [22] and its pseudocode in A.2

- `rootsfind()`: This function extracts individual real or complex roots from list of quadratic factors.
- `deflation()`: It is simply polynomial division. Suppose find a single root of an $n - th$ order polynomial. The effort of finding a root hopefully decreases in each step. But the mehod cannot convergen twice to the sam non multiple root.
- `quadfinder()`: Find quadratic factor using Bairstow's method (quadratic Newton method). A number of ad hoc safeguards are incorporated to prevent stalls due to common difficulties, such as zero slope at iteration point, and convergence problems.
- `diff_poly()`: Differentiate polynomial a returning result in **b**
- `recursive()`: This function reduces the order of original polynomial to lower order of all multiple roots by one, and has no other roots in common with it. If a root of the differentiated polynomial is a root of the original polynomial, there must be multiple roots at that location. The differentiated polynomial, however, has lower

order and is easier to solve. When the original polynomial exhibits convergence problems in the neighborhood of some potential root, a best guess is obtained and tried on the differentiated polynomial. The new best guess is applied recursively on continually differentiated polynomials until failure occurs. At this point, the previous polynomial is accepted as that with the least number of roots at this location, and its estimate is accepted as the root.

times.h

- `switch_time()`: This function compute switching times α_i according [7] in 2.2.2.

5.4 Compiling driver and application

In each directory is Makefile. To compile the driver is used toolchain 152-linux-2.6.34 that is set in PATH environment variable. The binary file will be `usart-pwm.ko`, we can copy it to the target and load it to Linux kernel according usage at 4.5.2. The `fft.c` and `optimalpwm.c` are applications, such as in section 4.1 so that must be compiled on the workstation by cross-compiler toolchain. And its binary files will be loaded onto the target board OC8-S. In this project, the cross-compiler toolchain `arm-gnueabi-gcc` is used to compile all applications 4.3. Output files are `fft-arm` and `optimalpwm-arm`, such as 4.28 we can load them onto the OC8-S.

Chapter 6

Testing and final work

In this chapter, several tests were conducted to ensure if the digital class-D audio amplifier has any problems. We are starting with software testing: check the correctness of the optimal PWM algorithm, calculate THD and the USART driver. The possible error scenarios that may happen in the environment and influence the device are run through next.

Finally, in the last section, the final embedded Linux will be configured and its image will be deployed on the target.

6.1 Software testing

6.1.1 The optimal PWM algorithm

The optimal PWM algorithm has been already tested on the PC workstation during the development of the digital class-D audio amplifier, no error was observed. The correctness of switching times is proven: Therefore input data are frequency spectrums b_{f_i} , output are switching times α_i . In table 6.1 are results of computation,

Table 6.1: The partial results for case $n = 20, n_C = 5, n_E = 15, A = 10, T = 0.01$ and $(b_{f_1}, b_{f_2}, b_{f_3}, b_{f_4}, b_{f_5}) = (4, -5, 3, 1, -2)$

i	p_{f_i}	μ_i	w_i	v_i	y_{f_i}	α_i
0	-	1	-	-	-	-
1	0.6858	-0.6858	-0.0005	-0.6991	-0.9905	0.0002266
2	0.3927	0.0388	-0.0007	-2.2299	0.9575	0.0004658
3	0.5877	-0.1150	0.0440	1.5471	-0.9167	0.0006929
4	0.3534	0.0283	0.0309	1.6195	0.8402	0.0009123
5	0.6318	-0.0773	-0.5062	-1.1226	0.6532	0.0011667
6	0.3093	0.0366	-0.1125	-0.3995	-0.6328	0.0013670
7	0.6822	-0.0665	1.7470	0.2848	0.4010	0.0016114
8	0.2749	0.0400	0.1004	0.0152	-0.3090	0.0018434
9	0.7239	-0.0600	-2.2778	-0.0151	0.1242	0.0020331
10	0.2485	0.0407	-0.0133	0.0002	-0.1138	0.0023018
11	0.7570	-0.0551	-	-	0.9899	0.0024801
12	0.2278	0.0403	-	-	-0.9713	0.0026815
13	0.7834	-0.0512	-	-	-0.8956	0.0029176
14	0.2111	0.0394	-	-	0.7431	0.0030000
15	0.8046	-0.0480	-	-	-0.6458	0.0036174
16	0.1974	0.0383	-	-	0.5298	0.0035904
17	0.8220	-0.0453	-	-	0.9067	0.0042663
18	0.1859	0.0372	-	-	-0.2594	0.0043457
19	0.8365	-0.0430	-	-	0.2892	0.0046177
20	0.1761	0.0361	-	-	0.0125	0.0047807

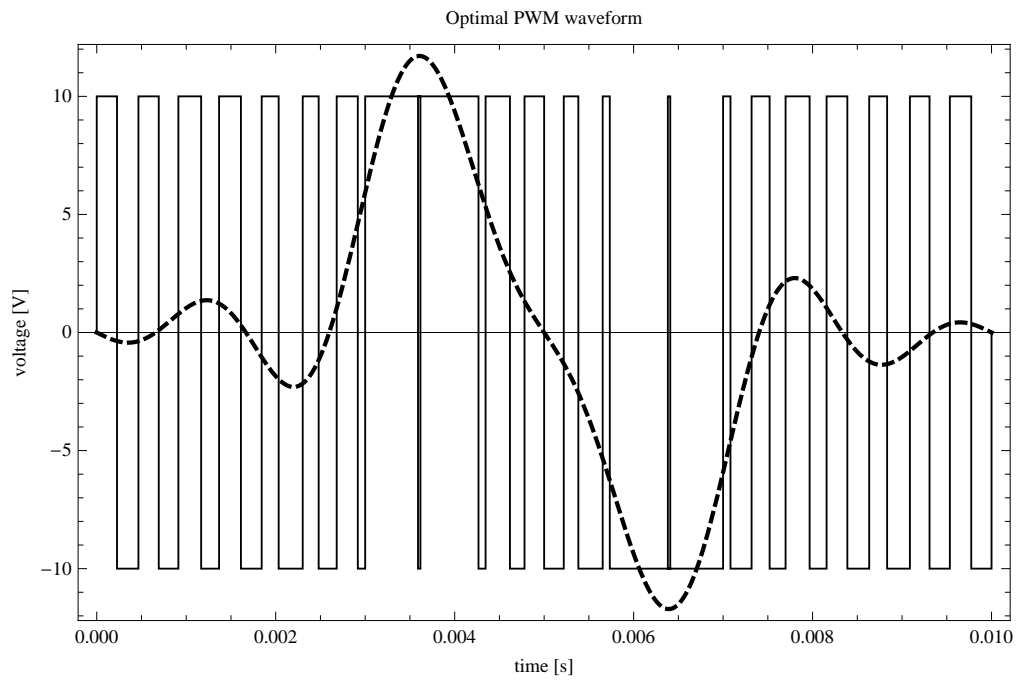
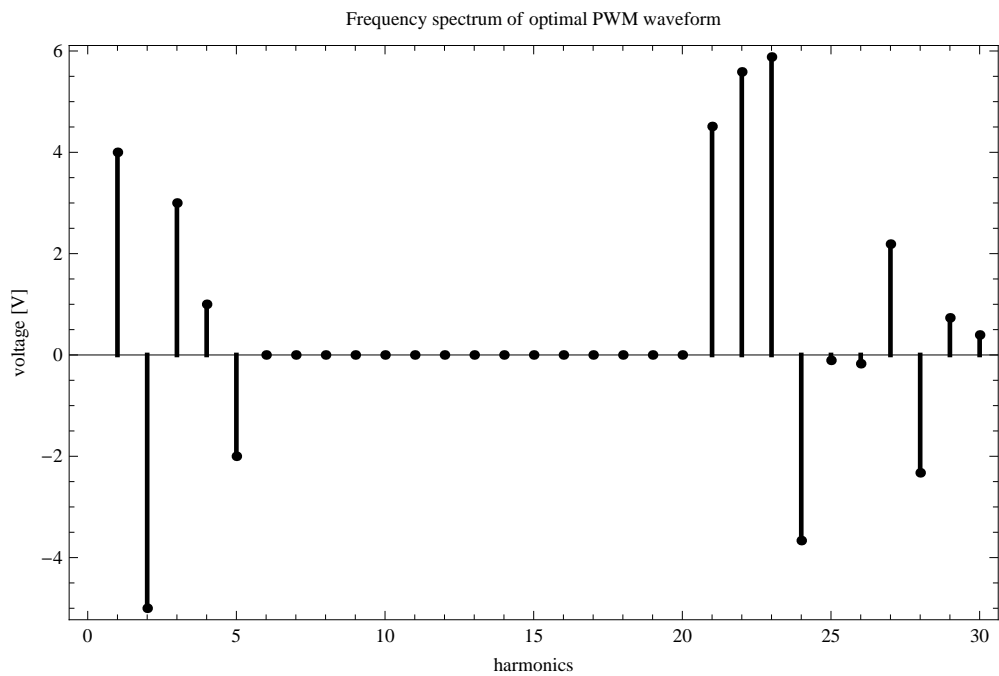
Figure 6.1: Odd bi-level PWM waveform for $n = 20$ Figure 6.2: Spectrums of odd bi-level PWM waveform for $n = 20$

Figure 6.1 and 6.2 show the input waveform of odd signal (the symmetry property of signal about $T/2$) and its computed PWM waveform for case $n_{even} = 20$. It really match

the input spectrums $(b_{f_1}, b_{f_2}, b_{f_3}, b_{f_4}, b_{f_5}) = (4, -5, 3, 1, -2)$ plus the cutoff frequency in the guard band and higher harmonics.

Setting of the values n and T has a big influence on a quality of signal. In table 6.2 are results for case $n_{odd} = 5$.

Table 6.2: The partial results for case $n = 5, n_C = 1, n_E = 4, A = 15, T = 0.0002$ and $(b_{f_1}) = (15)$

i	p_{f_i}	μ_i	w_i	v_i	y_{f_i}	α_i
0	-	1	-	-	-	-
1	-0.7854	-0.7854	-0.4003	-0.2446	-0.4792	0.00001550
2	1	0.8084	-0.8058	-0.4221	-0.9453	0.00002135
3	-0.5890	-0.6698	0.5408	-	0.8837	0.00006590
4	1	0.6993	-	-	0.7834	0.00006811
5	-0.4909	-0.5943	-	-	-0.5388	0.00008942

We have equation

$$THD(\bar{\alpha})[\%] = 100 \sqrt{\frac{\sum_{i=n_c+1}^{n+N} \left(\frac{a_{p_i}(\bar{\alpha}) + b_{p_i}(\bar{\alpha})}{i} \right)^2}{\sum_{i=1}^{n_c} \left(\frac{a_{p_i}(\bar{\alpha}) + b_{p_i}(\bar{\alpha})}{i} \right)^2}} \quad (6.1)$$

It determines the total harmonic distortion (THD). It is an amplifier specification that compares the output signal of amplifier with the input signal and measures the level differences in harmonic frequencies between the two. The difference is called total harmonic distortion. Total harmonic distortion is measured as a percentage, and it means that how the output of harmonic distortion is different than the input signal. Lower percentages are better.

The total harmonic distortion is influenced by setting of variables n : with decreasing n THD increases exponentially that increases the distortion of signal. From Equation 6.1, we achieve the minimal THD if the numenator

$$\sqrt{\sum_{i=n_c+1}^{n+N} \left(\frac{a_{p_i}(\bar{\alpha}) + b_{p_i}(\bar{\alpha})}{i} \right)^2} \quad (6.2)$$

is minimal. It means the uncontrolled harmonics must be minimal or zero.

According Equation 6.1 we obtain: for $n = 5$, $THD = [5.59\%]$ and $n = 20$, $THD = [4.55\%]$.

6.2 Embedded PWM generator

In this section, the stability of the embedded Linux will be checked. Each test is conducted with several input data (n, n_c, A, T, b_f) . With the command

```
./optimalpwm-arm
```

Table 6.3: Execute optimalpwm

the binary file of optimal PWM algorithm is executed. On the pinout $R37$ of OC8H, we can observe on the oscilloscope monitor

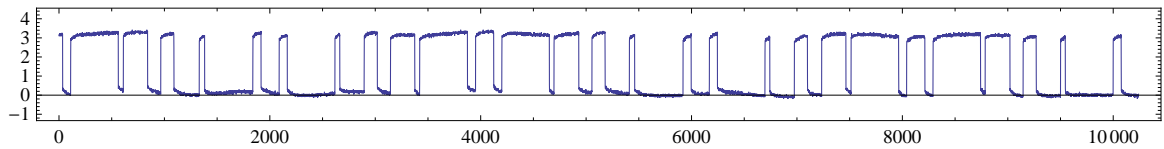


Figure 6.3: PWM output signal for $n = 5$

the PWM output signal with voltage range between 0 and 3.27V.

With using Mathematica to redraw measured data, we obtain the real odd bi-level PWM output with periodic generation for case $n = 5$ in the voltage range from $-3.27V$ to 3.27V.

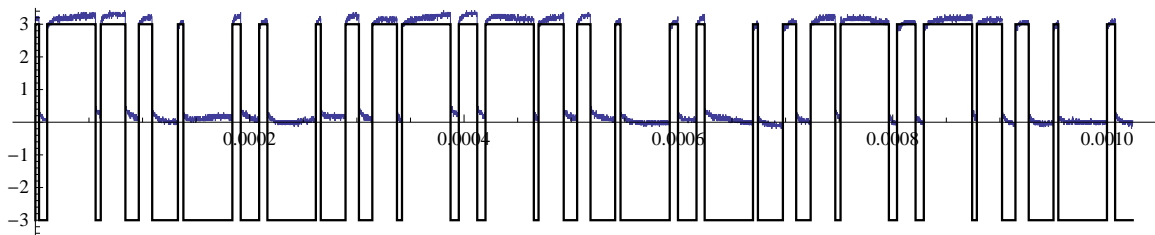


Figure 6.4: Simulated odd bi-level PWM waveform for $n = 5$

Figure 6.5 shows the first period of the generated odd bi-level PWM waveform for $n = 5$.

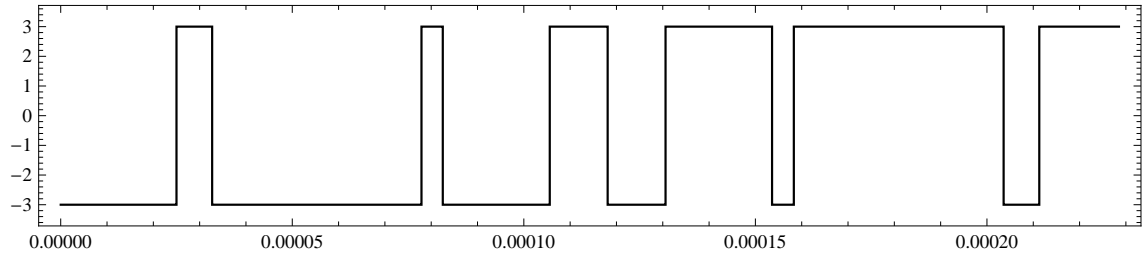


Figure 6.5: The first period of the real PWM output signal for $n = 5$

It really match the odd signal because it is symmetric about $T/2 = 0.0001$.

The results of the real and computed odd bi-level PWM waveform are portrayed in Figure 6.6.

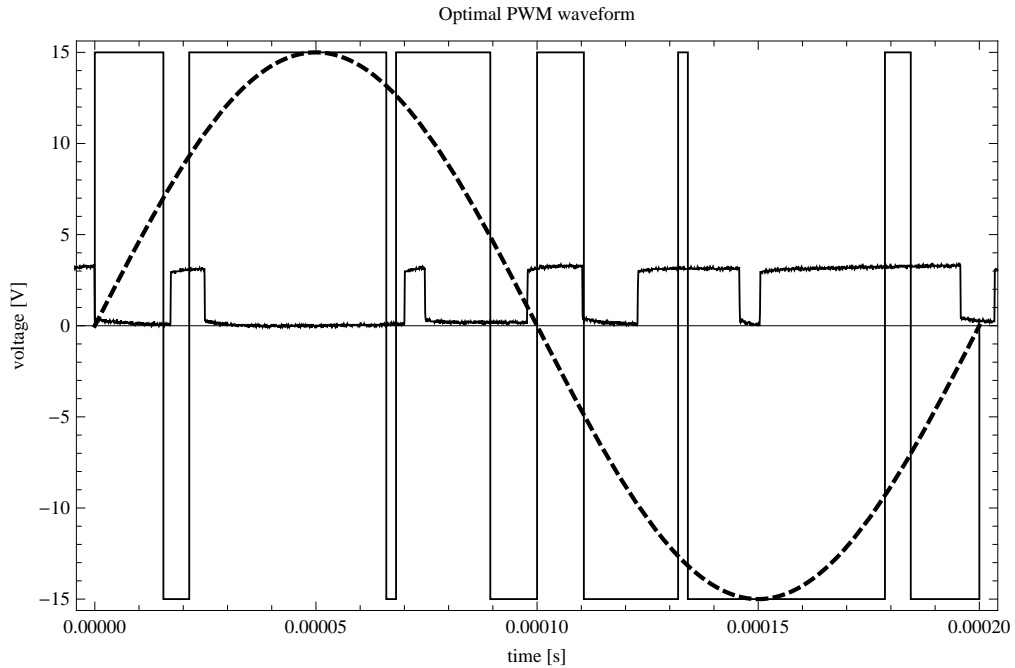


Figure 6.6: The measured and computed PWM output signal fo $n = 5$

From the Figure 6.6 above we could see how this PWM output is formed by the waveform generator base on value of switching times α_i . But the real PWM waveform isn't correctly corresponding to the computed PWM waveform. Every pulse of the real PWM are about $2(\mu s)$ delayed than the computed PWM waveform. The reason is explained that we used the pulse rate $5000(Hz)$ (period of $200(\mu s)$) for $n = 5$. So this frequency is higher than the frequency of OC8-S is used to generate the PWM output.

Similar, if we use the slower pulse rate $f = 1000(Hz)$ for case $n = 4, n_C = 1, n_E = 3, A = 6$ with $(b_{f_1}) = (3)$, we obtain

Table 6.4: The partial results for case $n = 4, n_C = 1, n_E = 3, A = 6, T = 0.001$ and $(b_{f_1}) = (3)$.

i	p_{f_i}	μ_i	w_i	v_i	y_{f_i}	α_i
0	-	1	-	-	-	-
1	0.6073	-0.6073	-0.3565	-0.3037	-0.7679	0.0001107
2	0	0.1844	0.3037	-0.3565	0.4642	0.0001732
3	0.7055	-0.2725	-	-	0.7679	0.0003268
4	0	0.1485	-	-	-0.4642	0.0003893

The PWM output with the periodic generation is formed by the waveform generator base on value of switching times α_i according to table 6.4 is shown in Figure 6.7

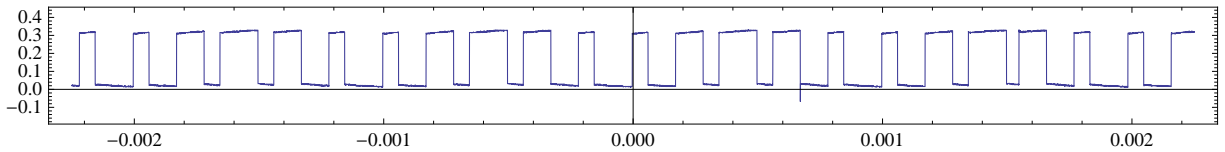


Figure 6.7: PWM output signal for $n = 4$

and Figure 6.8

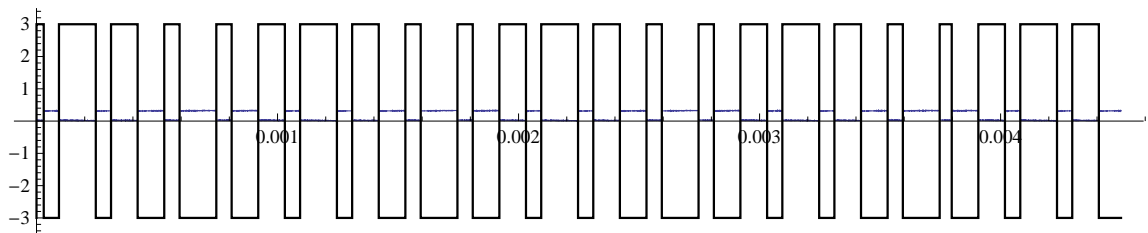


Figure 6.8: Simulated odd bi-level PWM waveform for $n = 4$

It is clear from Figure 6.9, it is really match the odd signal because it is symmetric about $T/2 = 0.0005$.

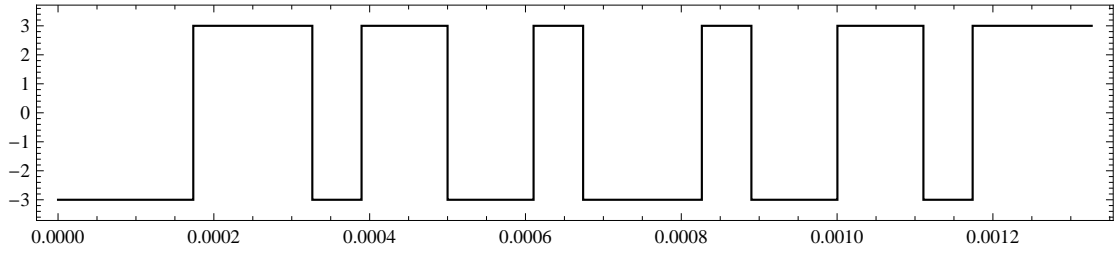


Figure 6.9: The first period of the real PWM output signal for $n = 4$

The results of the real and computed odd bi-level PWM waveform are portrayed in Figure 6.10

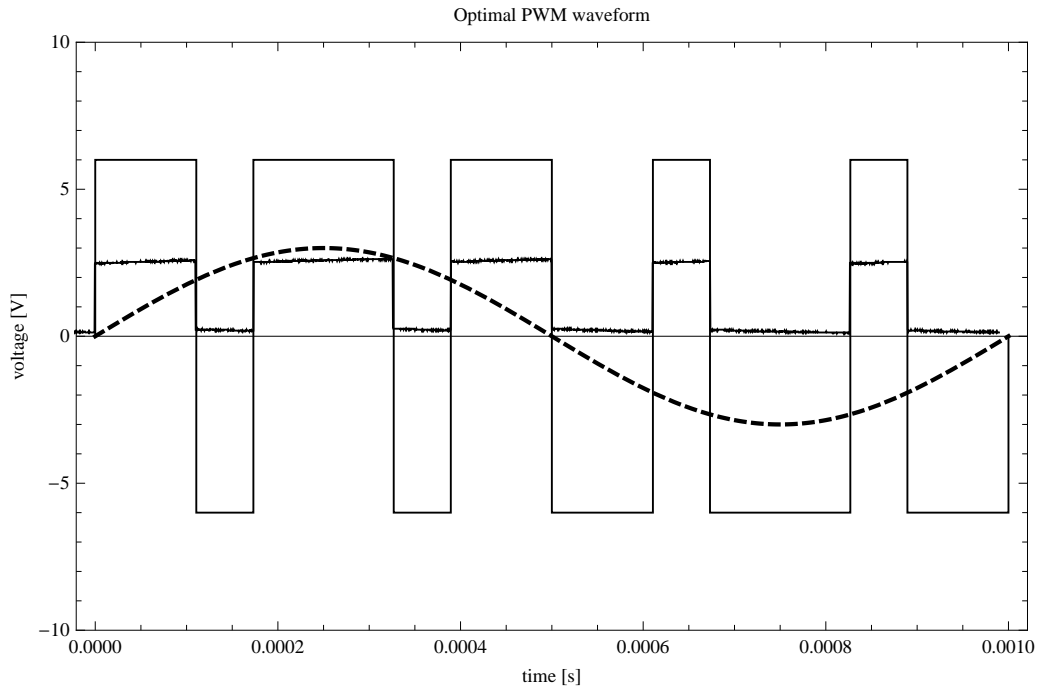


Figure 6.10: The real and computed PWM output signal for $n = 4$

In this case for the pulse rate $f = 1000(Hz)$, we could see that the real PWM output is corresponding to the computed PWM waveform.

From results for the case $n = 4, f = 1000(Hz)$, correctness of the optimal PWM algorithm is proven. Using this method, we are able to generate the driven PWM output signal. But it is limited by the power of selected hardware OC8-S. The PWM output is generated precisely only at slower frequency, it is about $f = 1000(Hz)$.

No failure has been observed during these tests. Switching times α_i were determined exactly and the PWM output was generated corresponding to the switching times.

6.3 Frequent error scenarios

In the previous section, we made several tests for software applications. The embedded Linux system that is used to develop our project isn't thorough, its less part has to be implemented from scratch, it has not any where verified. Thus, in the next phase of testing we continue analyze the system's behavior if an error occurs in the environment of the digital class-D audio amplifier that directly influences its operation. It is important that the system detects these errors and reacts to them by displaying an appropriate error message and bringing itself in a consistent state.

In the following, the frequent errors scenarios are explained and for each one the system's behavior is described.

Loss of an NFS connection

If the device providing the NFS network share respectively the NFS service running on it is shut down, the user has tried restart network configuration on the PC workstation and mount again the embedded module OC8-S.

Loading and copying USART driver to kernel

Another error is loading and copying USART driver to the kernel of the embedded Linux on OC8-S. The command

```
$ lsmod
```

Table 6.5: Show the status of modules in the Linux Kernel

shows the status of module in the Linux kernel. If the driver hasn't been loaded to the kernel, the user can load it into the kernel. If it already have loaded and it occurred error,remove it.

6.4 Final work

Before the embedded device can be used as audio amplifier class-D, some steps have to be accomplished: First, the embedded Linux configuration is cleaned up, i.e. all

applications that are not needed for proper operation are removed. This makes the audio amplifier image smaller that will be written to flash memory and thereby allows faster decompressing at boot and hence faster boot time.

All processes that are automatically started after boot system and they are not required for operation of the audio amplifier class-D, are terminated. It would even impose a large risk because it may be cause error of the embedded Linux system on the target.

Finally, this image has to be deployed onto the target. As soon as the image resides in flash memory, the U-Boot bootloader has to be configured to automatically boot from there. Both steps are described in section 4.4.4.

Chapter 7

Conclusion

As the final parts of this document, a summary of the work is presented in this chapter. The aim of the project is to try to develop the digital class-D audio amplifier with a new approach, using the optimal PWM algorithm to generating PWM like output signal. The module OC8-S based on ARM processor architecture AT91SAM9G20 was selected as a hardware prototype. The embedded Linux distribution that is available for the ARM architecture is used here because it was already included several software components inevitable for the project. And it is a familiar computing environment whose availability on embedded systems makes it easy to build an embedded application

Software development was the main phase of this project. The start was made by implementing the device driver for USART which provides to receive and transmit data. We use this feature of USART like PWM controller. The best way to generate PWM output signal is used the implemented PWM controller on the hardware. But in our case, the AT91SAM9G20 processor doesn't provide this PWM controller so we must use the alternative way to generate PWM output. It can receive input data and transmit them to pinout on the header board OC8-H. The advantages of this solution are: First that easy to handle the PWM output, the PWM module can run independently from the optimal PWM algorithm. We can compute switching times α_i in one program then we can send them directly to the module. Second that an PWM kernel is developed free as open source software. Everyone who are interesting in the the optimal PWM problem can have access to the complete source code.

Development continued with user space applications: The optimal PWM algorithm to determine switching times. Firstly, it was implemented and compiled by ARM GCC on the PC workstation, then it was loaded on to the target board.

The aim of this project as stated in the introductory chapter were clearly fulfilled. A

picture of the device during operation is shown in Figure 7.1

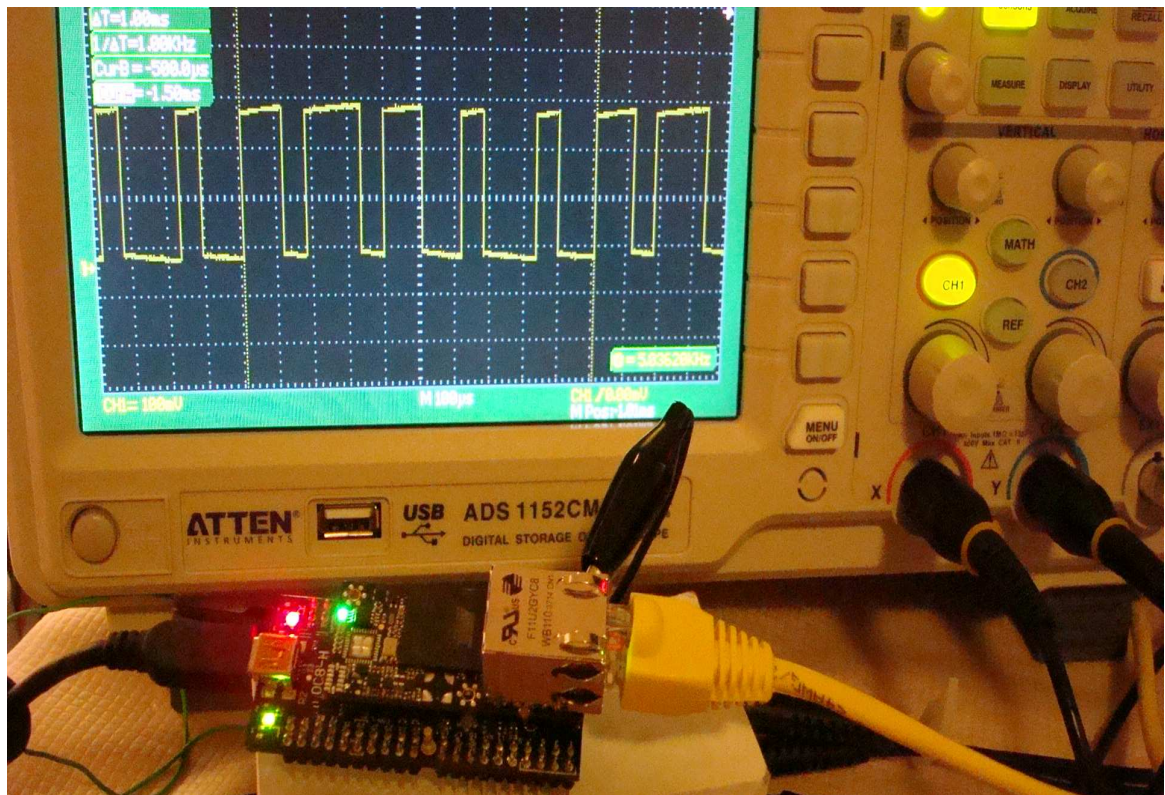


Figure 7.1: The prototype of the digital class-D audio amplifier.

The following conclusions of the work are :

- The advantage of Linux in general is that it is free of royalties because it is open source software. Everyone has access to the complete source code which is of high importance when developing embedded systems.
- The embedded Linux system is a fully functional operating system which makes software development for embedded systems much easier.
- A wide range of drivers, protocols and user space applications are available with the embedded Linux distribution.
- The version of the embedded Linux ported to the ARM processor that was used for the digital class-D audio amplifier was very stable, crashes of the kernel did never happen. However, a few problems were detected in important applications or tools. For example, the NFS server or DHCP network protocol suffered from a programming error that led to a failure.

- The conclusion thereof is that thoroughly testing of each application or kernel module is a must before it is loaded on to an embedded system.

In the future, the embedded Linux system for ARM processor including the applications is expected to become more and more stable. Development activity in the community is very high.

Finally, some ideas for future enhancements of the digital class-D audio amplifier are presented:

- The optimal PWM coding algorithm would be implemented in programming language more optimal.
- The USART driver would be more and more stable.
- We could use another hardware prototype to develop the audio amplifier class-D that provides more quality and accurate PWM signal than OC8-S. For example, FPGA.

Appendix A

Algorithms/Mathematical Background

A.1 QR decomposition

Algorithm A.1.1: MODIFIED GRAM SCHMIDT(R, Q)

```
for  $k = 1$  to  $n$  do
begin
   $s := 0$ ;
  for  $j := 1$  to  $m$  do
 $s := s + a_{jk}^2$ 
 $r_{kk} = \text{sqrt}(s)$ ;
  for  $j := 1$  to  $m$  do
 $q_{jk} = a_{jk}/r_{kk}$ ;
  for  $i := k + 1$  to  $n$  do
begin
   $s := 0$ ;
  for  $j := 1$  to  $m$  do  $s := s + a_{ji} * q_{jk}$ ;
   $r_{ki} := s$ ;
  for  $j := 1$  to  $m$  do  $a_{ji} := a_{ji} - r_{ki} * q_{jk}$ ;
end;
end
```


A.2 A pseudocode algorithm for finding real and complex roots of real polynomials with multiple roots

1. Call Bairstow's method with small initial estimate.
2. If convergence was satisfactory, goto [10].
3. Let $D = P'$.
4. If the constant term is zero, shift all coefficients down reducing order by [1].
5. If the order of D is 2, estimate is D : goto [10].
6. Call Bairstow's method on D using estimate.
7. If convergence was satisfactory, goto [10].
8. If wrong root, restore D and previous estimate, goto [10].
9. Goto [3].
10. Deflate polynomial using estimate and save quadratic.
11. Replace original polynomial with reduced polynomial.
12. If the order of the reduced polynomial is greater than 2, goto [1].
13. Get remaining root(s) and return to caller

Appendix B

Content of the Attached CD

1. Directory 1: The sources code of application in *.tar.gz archive files.

- **U-Boot for module OC8-S**

uboot.tar.gz

- **uImage for module OC8-S**

uimage.tar.gz

- **USART-PWM**

usartpwm.tar.gz

- **optimalpwm for x86**

optimalwpmx86.tar.gz

- **optimalpwm for oc8s**

optimalpwmoc8s.tar.gz

- **FFT**

fft.tar.gz

2. Directory 2: Thesis

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