

# ELEN0019-1 Audio Signal Processing

## Principles and Experiments

Julien OSMALSKYJ

University of Liège

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# Contents

- 1 Course organization
- 2 Introduction
- 3 DSP Architecture
- 4 Input / Output
- 5 Software : Code Composer Studio
- 6 Project example

# Contents

- 1 Course organization
- 2 Introduction
- 3 DSP Architecture
- 4 Input / Output
- 5 Software : Code Composer Studio
- 6 Project example

# General information

- Julien Osmalskyj
- Office R28
- `www.montefiore.ulg.ac.be/~josmalskyj/dsp.php`
- **Email** : `josmalsky@ulg.ac.be`

# Course organization

Practical course organized in 12 lab sessions of 4 hours.  
Groups of 2 - 3 students.

- 6 sessions for the course
- 6 sessions for the final project

No theoretical session except this first one.

## Evaluation

Evaluation is based only on the students final project. No evaluation of the lab sessions.

Everybody has to attend each lab session in order to succeed the course.

# Students projects examples

- Guitar Tuner
- Equalizer
- Tap-delay
- Chorus - Flanger
- Compressor
- Loudspeaker frequency correction
- Signal analyzer
- etc.

# Contents

- 1 Course organization
- 2 Introduction
- 3 DSP Architecture
- 4 Input / Output
- 5 Software : Code Composer Studio
- 6 Project example

# Introduction

Digital signal processors are used in many areas such as

- sound
- video
- computer vision
- music analysis
- etc.

They are found in

- Cellular phones
- Disk drives
- MP3 players
- etc.



# Introduction

## Principle

A DSP digitizes an analog signal, manipulates it using mathematical and logical operations and converts the result back to an analog wave form.



FIGURE : General DSP system

# Introduction

## Real-time constraint

DSP are processors specialized for real-time processing. Audio samples arrive at a constant rate (e.g. every  $1/48000$  seconds) and must be processed before the next samples arrive.

DSP have a specific architecture and specialized instructions optimized to minimize the number of CPU clock cycles.

- MAC instruction (Multiply ACcumulate) for fast convolutions
- Fast memory access
- Harvard or modified Harvard architecture
- Programmable in Assembly or C language

# Contents

- 1 Course organization
- 2 Introduction
- 3 DSP Architecture**
- 4 Input / Output
- 5 Software : Code Composer Studio
- 6 Project example

# Texas Instrument C6748 Processor

In this course, we use a Texas Instrument OMAP-L138 processor which is a single chip containing a C6748 DSP core and an ARM9 processor.

- Based on Texas Instrument very long instruction word (VLIW)
- Clock rate of 375 MHz
- Fetches eight 32-bit instructions every clock cycle
- Both floating-point and fixed-point architecture

# Interrupts

4 types of interrupt on the CPU :

- Reset
- Maskable
- Non-maskable (NMI)
- Exception

Reset and Non-maskable interrupts have the highest priority. 12 maskable interrupts (INT4 - INT15) can be associated with external devices, on-chip peripherals or software control.

## Receive samples

In this course, events corresponding to a new input sample (events #61 or #8) are associated with INT4 interrupt.

# Interrupt selector (IS)

128 systems events are available in the DSP. The IS allows to select one event and route it to the appropriate CPU interrupt.

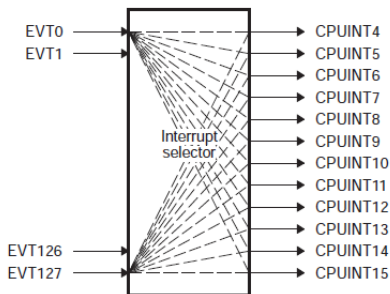


FIGURE : Interrupt selector

# Interrupt Service Table (IST)

The code executed when an interrupt occurs is determined by the content of the IST. The IST contains one Interrupt Service Fetch Packet (ISFP) associated with each maskable interrupt. The ISFP associated with INT4 contains a branch instruction to a function `interrupt4()` which must be defined in the C program.

|           |            |
|-----------|------------|
| xxxx 000h | RESET ISFP |
| xxxx 020h | NMI ISFP   |
| xxxx 040h | Reserved   |
| xxxx 060h | Reserved   |
| xxxx 080h | INT4 ISFP  |
| xxxx 0A0h | INT5 ISFP  |
| xxxx 0C0h | INT6 ISFP  |
| xxxx 0E0h | INT7 ISFP  |
| xxxx 100h | INT8 ISFP  |
| xxxx 120h | INT9 ISFP  |
| xxxx 140h | INT10 ISFP |
| xxxx 160h | INT11 ISFP |
| xxxx 180h | INT12 ISFP |
| xxxx 1A0h | INT13 ISFP |
| xxxx 1C0h | INT14 ISFP |
| xxxx 1E0h | INT15 ISFP |

Program memory

FIGURE : Interrupt Service Table





# OMAP-L138 eXperimenter

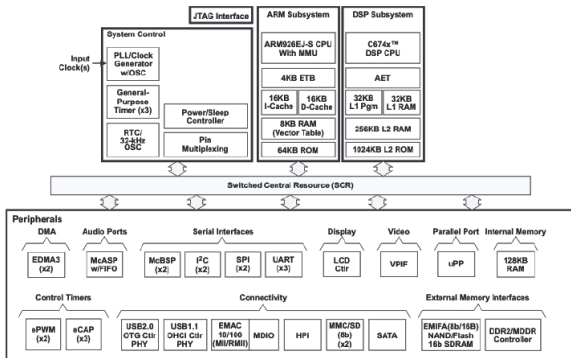


FIGURE : OMAP-L138 eXperimenter

# Contents

- 1 Course organization
- 2 Introduction
- 3 DSP Architecture
- 4 Input / Output**
- 5 Software : Code Composer Studio
- 6 Project example

# Input / Output

## Library

A basic library for accessing DSP inputs and outputs is used for this course.

3 ways of reading input samples and writing output samples.

- Polling
- Interrupts
- Direct Memory Access (DMA)

# Library functions

Functions are available for reading and writing samples in each mode.

- `int32_t input_sample()`
- `int16_t input_left_sample()`
- `int16_t input_right_sample()`
- `void output_sample(int32_t out_data)`
- `void output_left_sample(int16_t out_data)`
- `void output_right_sample(int16_t out_data)`

# Data format

AIC3106 codec converts samples of both channels to 16-bit signed integers. Channels are combined to form a 32-bit sample.

```
1 typedef union {  
2     uint32_t uint;  
3     short channel[2];  
4 } AIC31_data_type;
```

FIGURE : Union structure to store samples

```
1 AIC31_data_type codec_data;  
2 codec_data.uint = input_sample();  
3 short left_sample = codec_data.channel[LEFT];  
4 short right_sample = codec_data.channel[RIGHT];
```

FIGURE : Read both left and right channels

# Polling scheme

## Principle

Processor queries the codec when the processing is finished.  
The input and output functions wait for the codec to be ready.

Initialization function :

```
L138_initialise_poll(FS_48000_HZ, ADC_GAIN_0DB,  
DAC_ATTEN_0DB)
```

- FS\_48000\_HZ : **Sampling frequency set to 48000 Hz**
- ADC\_GAIN\_0DB : **Gain at the input set to 0 dB**
- DAC\_ATTEN\_0DB : **Attenuation at the output set to 0 dB**

# Polling code example

```
1 #include "L138_aic3106_init.h"
2
3 void main(void) {
4     uint32_t sample;
5
6     L138_initialise_poll(FS_48000_HZ, ADC_GAIN_0DB, DAC_ATTEN_0DB);
7     while (1) {
8         sample = input_sample();
9         output_sample(sample);
10    }
11 }
```

FIGURE : Input / Output using polling

# Interrupt-based scheme

## Principle

INT4 is triggered when a sample arrives at the input of the codec and the code of function `interrupt4()` is executed.

Initialization function :

```
L138_initialise_intr(FS_48000_HZ, ADC_GAIN_0DB,  
DAC_ATTEN_0DB)
```



# Interrupt-based code example

```
1 #include "L138_aic3106_init.h"
2
3 interrupt void interrupt4(void) { // interrupt routine
4     uint32_t sample;
5     sample = input_sample();
6     output_sample(sample);
7     return;
8 }
9
10 void main(void) {
11     L138_initialise_intr(FS_48000_HZ, ADC_GAIN_0DB, DAC_ATTEN_0DB);
12     while (1) ;
13 }
```

FIGURE : Input / Output using interrupts

# Direct Memory Access (DMA)

## Principle

EDMA3 controller transfers **blocks of  $N$  samples** between the codec and the memory without intervention of the CPU. An interruption is triggered when all  $N$  samples have been transferred.

Initialization function :

```
L138_initialise_edma(FS_48000_HZ, ADC_GAIN_0DB,  
DAC_ATTEN_0DB)
```

# DMA code example I

```
1 interrupt void interrupt4(void) { // interrupt routine
2   switch(EDMA_3CC_IPR) {
3     case 1:                // TCC = 0
4       procBuffer = PING;   // process ping
5       EDMA_3CC_ICR = 0x0001; // clear EDMA3 IPR bit TCC
6       break;
7
8     case 2:                // TCC = 1
9       procBuffer = PONG;   // process pong
10      EDMA_3CC_ICR = 0x0002; // clear EDMA3 IPR bit TCC
11      break;
12
13     default:              // may have missed an interrupt
14       EDMA_3CC_ICR = 0x0003; // clear EDMA3 IPR bits 0 and 1
15       break;
16   }
17   EVTCLR0 = 0x00000100;
18   buffer_full = 1;        // flag EDMA3 transfer
19   return;
20 }
```

FIGURE : DMA interrupt routine

# DMA code example II

```
1 int main(void) {  
2     L138_initialise_edma (FS_48000_HZ , ADC_GAIN_0DB , DAC_ATTEN_0DB);  
3     zero_buffers ();  
4  
5     while (1) {  
6         while (!is_buffer_full ());  
7         process_buffer ();  
8     }  
9 }
```

**FIGURE :** DMA scheme main function

The `process_buffer ()` function implements the processing of the block of samples.

# Contents

- 1 Course organization
- 2 Introduction
- 3 DSP Architecture
- 4 Input / Output
- 5 Software : Code Composer Studio**
- 6 Project example

# Code Composer Studio

Code Composer Studio (CCS) is the integrated development environment provided by Texas Instrument.

It is based on the Eclipse framework and offers a convenient editor and a debugger.

## Important note

By default, CCS compiles the code in *debug* mode. To achieve better performances, configure CCS to compile the code in *release* mode.

# Code Composer Studio : Editor

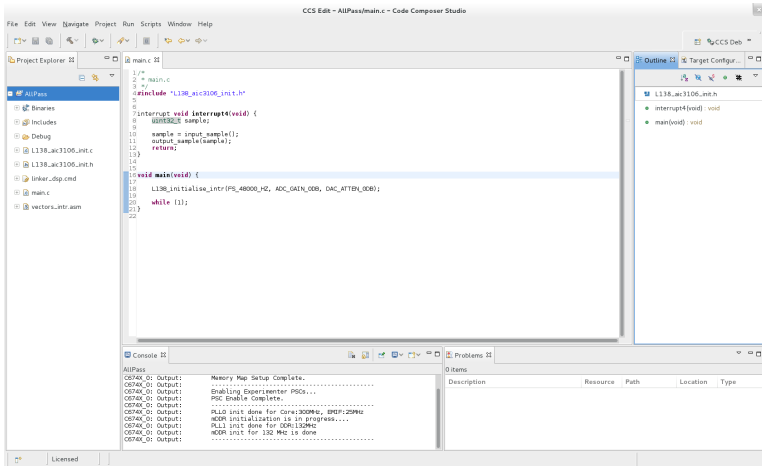


FIGURE : CCS Editor

# Code Composer Studio : Debugger

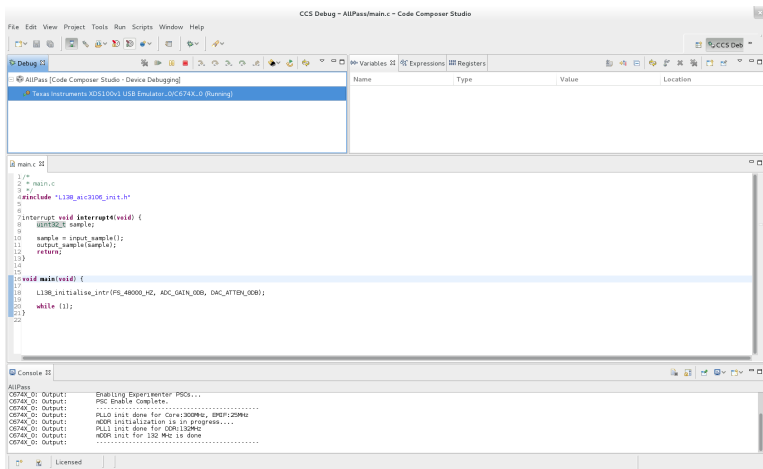


FIGURE : CCS Debugger



# Contents

- 1 Course organization
- 2 Introduction
- 3 DSP Architecture
- 4 Input / Output
- 5 Software : Code Composer Studio
- 6 Project example

# Project Example

Let us create a simple project which switches the left and right channels of an input signal.

The procedure to program the DSP is the following.

- Create new CCS project
- Configure include paths and linker file search path
- Add necessary files to the projects
- Code the program
- Compile in *debug* or *release* mode
- Run the program by starting the debugger

Detailed procedure for creating a new project in CCS is explained in Chapter 2 of the course notes.

# Interrupt service routine

```
1 #include "L138_aic3106_init.h"
2
3 AIC31_data_type codec_data;
4 int channel = LEFT;
5
6 interrupt void interrupt4(void) {
7     codec_data.uint = input_sample();
8     short left_sample = codec_data.channel[LEFT];
9     short right_sample = codec_data.channel[RIGHT];
10
11     if (channel == LEFT) {
12         codec_data.channel[LEFT] = left_sample;
13         codec_data.channel[RIGHT] = right_sample;
14         channel = RIGHT;
15     }
16     else {
17         codec_data.channel[LEFT] = right_sample;
18         codec_data.channel[RIGHT] = left_sample;
19         channel = LEFT;
20     }
21     output_sample(codec_data.uint);
22     return;
23 }
```

# Interrupt service routine

The main function only initializes the DSP using interrupt-based scheme and starts the main loop. At each sampling instant

$$T = 1/48000s$$

the program will be interrupted.

```
1 void main(void) {  
2  
3     L138_initialise_intr(FS_48000_HZ, ADC_GAIN_0DB, DAC_ATTEN_0DB);  
4  
5     while (1);  
6 }
```

FIGURE : Main function