Introduction

The PDM2PCM library converts a PDM bit stream from a MEMS microphone into a PCM audio stream.

This user manual describes the PDM2PCM library, which is part of the STM32Cube firmware package. It provides details about the interface parameters and the configuration of the library. It also shows how to integrate this library into a main program.

This document is applicable to the microcontrollers that allow the user to connect a digital PDM microphone, namely those of STM32F4, STM32F7 and STM32H7 Series.
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1 Module overview

1.1 Algorithm functionality

The PDM2PCM library has the function to decimate and filter out a Pulse Density Modulated (PDM) stream from a digital microphone, to convert it to a Pulse Code Modulated (PCM) signal output stream.

The PCM output stream is implemented with 16-bit resolution. The sampling rate is not specified in the interface but it is agreed in this document that the PCM sampling rate used is 16 kHz. Various decimation factors can be configured, to adapt to various PDM clocks.

A configurable high-pass filter and a digital volume are also proposed.

1.2 Module configuration

PDM2PCM library takes as input a PDM signal (768 kHz to 2.048 MHz) stream of 1-bit digital samples. This signal is acquired in blocks of 8 samples by using a synchronous serial port (SPI or I2S) of the STM32 microcontroller, based on Arm® cores(a).

Different versions of the module are available, depending upon the core and the used tool chain.

a. Arm is a registered trademark of Arm Limited (or its subsidiaries) in the US and/or elsewhere.
1.3 Summary of resources

*Table 1* contains the requirements for memories and frequency.

The footprints are measured on board, using IAR Embedded Workbench® for Arm® v7.40 (IAR Embedded Workbench® common components v7.2).

<table>
<thead>
<tr>
<th>PDM clock</th>
<th>Flash code .text (bytes)</th>
<th>Flash data .rodata (bytes)</th>
<th>Stack (bytes)</th>
<th>RAM (bytes)</th>
<th>Frequency (MHz)</th>
</tr>
</thead>
<tbody>
<tr>
<td>2.048 MHz (decimation = 128)</td>
<td>7020</td>
<td>789</td>
<td>50</td>
<td>1028</td>
<td>4.9</td>
</tr>
<tr>
<td>1.280 MHz (decimation = 80)</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>3.4</td>
</tr>
<tr>
<td>1.024 MHz (decimation = 64)</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>2.7</td>
</tr>
<tr>
<td>768 kHz (decimation = 48)</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>2.3</td>
</tr>
<tr>
<td>512 kHz (decimation = 32)</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1.8</td>
</tr>
</tbody>
</table>
## Module interfaces

Two files are needed to integrate the PDM2PCM library, the pdm2pcm_glo.h header file and the right library file (according to target and tool chain). They contain all definitions and structures to be exported to the software integration framework.

*Note:* The audio_fw_glo.h file is a generic header file common to all audio modules and must be included in the audio framework.

### 2.1 APIs

Five functions have a software interface to the main program:

- `PDM_FilterInit`
- `PDM_Filter_setConfig`
- `PDM_Filter_getConfig`
- `PDM_Filter_deInterleave`
- `PDM_Filter`

#### 2.1.1 `PDM_FilterInit` function

This procedure initializes the static memory, sets default values and initializes lookup tables of the PDM2PCM library.

```c
uint32_t PDM_FilterInit(PDM_Filter_Handler_t *pHandler);
```

<table>
<thead>
<tr>
<th>I/O</th>
<th>Name</th>
<th>Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Input</td>
<td>pHandler</td>
<td>PDM_Filter_Handler_t *</td>
<td>Pointer to internal static memory</td>
</tr>
<tr>
<td>Returned value</td>
<td>-</td>
<td>uint32_t</td>
<td>Error value</td>
</tr>
</tbody>
</table>

This routine must be called at least once at initialization time, when the real time processing has not started yet.

#### 2.1.2 `PDM_Filter_setConfig` function

This procedure sets module dynamic parameters from the main framework to the module internal memory. It can be called at any time during processing.

```c
uint32_t PDM_Filter_setConfig(PDM_Filter_Handler_t *pHandler, PDM_Filter_Config_t *pConfig);
```
2.1.3  
**PDM_Filter_getConfig** function  
This procedure gets module dynamic parameters from internal static memory to the main framework. It can be called at any time during processing.

```c
uint32_t PDM_Filter_getConfig(PDM_Filter_Handler_t *pHandler,  
PDM_Filter_Config_t *pConfig);
```

<table>
<thead>
<tr>
<th>I/O</th>
<th>Name</th>
<th>Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Input</td>
<td>pHandler</td>
<td>PDM_Filter_Handler_t *</td>
<td>Pointer to internal static memory</td>
</tr>
<tr>
<td>Input</td>
<td>pConfig</td>
<td>PDM_Filter_Config_t *</td>
<td>Pointer to dynamic parameters structure</td>
</tr>
<tr>
<td>Returned value</td>
<td>-</td>
<td>uint32_t</td>
<td>Error value</td>
</tr>
</tbody>
</table>

2.1.4  
**PDM_Filter_deInterleave** function  
Not yet implemented.

2.1.5  
**PDM_Filter** function  
This procedure decodes an input PDM stream to an output PCM stream. It has to be called to process each frame.

```c
uint32_t PDM_Filter(void *pDataIn, void *pDataOut, PDM_Filter_Handler_t *pHandler);
```

<table>
<thead>
<tr>
<th>I/O</th>
<th>Name</th>
<th>Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Input</td>
<td>pDataIn</td>
<td>void *</td>
<td>Pointer to PDM input data</td>
</tr>
<tr>
<td>Output</td>
<td>pDataOut</td>
<td>void *</td>
<td>Pointer to PCM output data</td>
</tr>
<tr>
<td>Input</td>
<td>pHandler</td>
<td>PDM_Filter_Handler_t*</td>
<td>Pointer to internal static memory</td>
</tr>
<tr>
<td>Returned value</td>
<td>-</td>
<td>uint32_t</td>
<td>Error value</td>
</tr>
</tbody>
</table>
2.2 External definitions

2.2.1 Returned error values

*Table 6* lists the possible returned error values. Each error sets a dedicated bit to 1, so more than one error code can be accumulated.

<table>
<thead>
<tr>
<th>Definition</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>PDM_FILTER_NO_ERROR</td>
<td>0x0000</td>
<td>No error</td>
</tr>
<tr>
<td>PDM_FILTER_ENDIANNESS_ERROR</td>
<td>0x0001</td>
<td>Unsupported endianness</td>
</tr>
<tr>
<td>PDM_FILTER_BIT_ORDER_ERROR</td>
<td>0x0002</td>
<td>Unsupported bit order</td>
</tr>
<tr>
<td>PDM_FILTER_CRC_LOCK_ERROR</td>
<td>0x0004</td>
<td>Target is not STM32</td>
</tr>
<tr>
<td>PDM_FILTER_DECIMATION_ERROR</td>
<td>0x0008</td>
<td>Unsupported decimation factor</td>
</tr>
<tr>
<td>PDM_FILTER_INIT_ERROR</td>
<td>0x0010</td>
<td>-</td>
</tr>
<tr>
<td>PDM_FILTER_CONFIG_ERROR</td>
<td>0x0020</td>
<td>-</td>
</tr>
<tr>
<td>PDM_FILTER_GAIN_ERROR</td>
<td>0x0040</td>
<td>Unsupported microphone gain</td>
</tr>
<tr>
<td>PDM_FILTER_SAMPLES_NUMBER_ERROR</td>
<td>0x0080</td>
<td>Unsupported number of samples</td>
</tr>
</tbody>
</table>

2.3 Static parameters structure

The PDM2PCM initial parameters are set using the corresponding static parameter structure before calling the `PDM_Filter_setConfig()` function.

```c
typedef struct {
    uint16_t bit_order;
    uint16_t endianness;
    uint32_t high_pass_tap;
    uint16_t in_ptr_channels;
    uint16_t out_ptr_channels;
    uint32_t pInternalMemory[INTERNAL_MEMORY_SIZE];
} PDM_Filter_Handler_t;
```

*Table 7. Static parameters*

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
<th>Comment</th>
</tr>
</thead>
<tbody>
<tr>
<td>bit_order</td>
<td>Specifies the bit order for input (MSB or LSB)</td>
<td>PDM_FILTER_BIT_ORDER_LSB (0x0000)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>PDM_FILTER_BIT_ORDER_MSB (0x0001)</td>
</tr>
<tr>
<td>endianness</td>
<td>Specifies if byte inversion is required</td>
<td>PDM_FILTER_ENDIANNESS_LE (0x0000)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>PDM_FILTER_ENDIANNESS_BE (0x0001)</td>
</tr>
<tr>
<td>high_pass_tap</td>
<td>Specifies the HP filter tap value</td>
<td>Coefficient value in Q31 format of the high-pass filter. If 0 the filter is not used.</td>
</tr>
</tbody>
</table>
Dynamic parameters structure

It is possible to change the PDM2PCM configuration by setting new values in the dynamic parameter structure before calling the `PDM_Filter_setConfig()` function.

```c
typedef struct {
    uint16_t decimation_factor;
    uint16_t output_samples_number;
    int16_t  mic_gain;
}PDM_Filter_Config_t;
```

### Table 7. Static parameters (continued)

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
<th>Comment</th>
</tr>
</thead>
</table>
| in_ptr_channels| Specifies the number of channels in the input PDM stream | INTEGER NUMBER > 0  
|                |                                                  | in_ptr_channels = 1 when used with one microphone.                     |
| out_ptr_channels| Specifies the number of channels in the output PCM stream. | INTEGER NUMBER > 0.  
|                |                                                  | out_ptr_channels=1 when used with one microphone.                      |
| plInternalMemory | Internal memory                              | Pointer to an array.                                                   |

### Table 8. Dynamic parameters

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
<th>Comment</th>
</tr>
</thead>
</table>
| decimation_factor  | Specifies the decimation factor.                 | PDM_FILTER_DEC_FACTOR_16 (0x0005)  
PDM_FILTER_DEC_FACTOR_24 (0x0006)  
PDM_FILTER_DEC_FACTOR_32 (0x0007)  
PDM_FILTER_DEC_FACTOR_48 (0x0001)  
PDM_FILTER_DEC_FACTOR_64 (0x0002)  
PDM_FILTER_DEC_FACTOR_80 (0x0003)  
PDM_FILTER_DEC_FACTOR_128 (0x0004) |
| output_samples_number| Specifies the number of PCM samples to be generated at each call of `PDM_Filter()` function | INTEGER NUMBER > 0 |
| mic_gain           | Specifies the microphone gain in dB              | Gain is in the interval [-12 dB: +51 dB], with 1 dB steps. |
3 Algorithm description

3.1 Processing steps

As shown in Figure 1, a MEMS microphone outputs a PDM stream, which is a high frequency stream of 1-bit digital samples. The library expects a stream made of 8-sample blocks (one byte), which will be acquired using a synchronous serial port (SPI or I2S) of the STM32 microcontroller. The microphone PDM output is synchronous with its input clock, therefore the used STM32 serial port generates a clock signal for the microphone.

![Figure 1. Block diagram](image)

The PDM data from the microphone are packed in 8-bit blocks, and then filtered and decimated. The frequency of the obtained PCM signal depends on the decimation factor configured before the library initialization.

The decimation factors have been defined to get a PCM stream of the desired sampling frequency, depending on the PDM clock value. Examples are given in Table 9.

<table>
<thead>
<tr>
<th>Decimation factor</th>
<th>PDM clock frequency</th>
<th>PCM sample rate</th>
</tr>
</thead>
<tbody>
<tr>
<td>128</td>
<td>1.024 MHz</td>
<td>8 kHz</td>
</tr>
<tr>
<td></td>
<td>2.048 MHz</td>
<td>16 kHz</td>
</tr>
<tr>
<td></td>
<td>3.072 MHz</td>
<td>24 kHz</td>
</tr>
<tr>
<td>80</td>
<td>1.280 MHz</td>
<td>16 kHz</td>
</tr>
<tr>
<td>64</td>
<td>1.024 MHz</td>
<td>16 kHz</td>
</tr>
<tr>
<td></td>
<td>2.048 MHz</td>
<td>32 kHz</td>
</tr>
<tr>
<td></td>
<td>3.072 MHz</td>
<td>48 kHz</td>
</tr>
<tr>
<td>48</td>
<td>768 kHz</td>
<td>16 kHz</td>
</tr>
<tr>
<td>32</td>
<td>512 kHz</td>
<td>16 kHz</td>
</tr>
<tr>
<td>24</td>
<td>384 kHz</td>
<td>16 kHz</td>
</tr>
<tr>
<td>16</td>
<td>256 kHz</td>
<td>16 kHz</td>
</tr>
</tbody>
</table>
The digital signal resulting from the filter and decimator pipeline is then processed by a high-pass filter, to remove DC offset, and by a digital gain, to attenuate or amplify the PCM samples.

### 3.1.1 High-pass filter

The high-pass filter is a one-pole recursive filter. The cut-off frequency is configured by modifying the parameter `high_pass_tap` from the `PDM_Filter_Handler_t`. This coefficient value must be in the range `[0 : 1]`. The format used is Q0.31, meaning that 1 corresponds to the maximum integer value obtainable with 31-bit resolution. For example, configuring the `high_pass_tap` parameter to 0.98 corresponds to $0.98 \times (2^{31} - 1) = 2104533974$.

*Figure 2* is a plot of the frequency response of this filter for three different values of the `high_pass_tap` parameter, for a PCM sampling rate of 16 kHz.

![Figure 2. Frequency response of the high-pass filter](image)

When the coefficient is set to 0, the high-pass filter is bypassed.
3.1.2 Digital volume

The digital volume attenuates or amplifies the samples before saturating them to a signed 16-bit value. The \textit{mic\_gain} parameter is the gain value (in dB) to apply to the PCM stream. The minimum value is -12 dB, the maximum is 51 dB, with 1 dB steps.

3.2 Data formats

The input of PDM2PCM library is expected to be a PDM stream, byte-packed, at the MEMS microphone clock frequency. It can be a single or double data stream. The output is a PCM stream.

3.3 Results of measurements

All measurements have been made using an STM32F469 board, with a MEMS microphone MP34DT01 mounted on it. They have been made in an anechoic environment, using a professional monitoring system as acoustic source.

3.3.1 Distortion measurements

The distortion measurements are made with a test signal at a nominal acoustic level of about 90 dB SPL at the source point. The microphone is placed at a distance of 10 cm, and \textit{mic\_gain} is equal to 0 dB.

These data take into account all system noise, including noise floors brought back by PCB and power supplies.

Figure 3. Distortion measurements at 300 Hz
### Table 10. Distortion measurements at 300 Hz

<table>
<thead>
<tr>
<th>PDM clock</th>
<th>SNR at -31 dBFS</th>
<th>SNR at 0 dBFS (extrapolation)</th>
</tr>
</thead>
<tbody>
<tr>
<td>2048 kHz</td>
<td>42.8</td>
<td>73.8</td>
</tr>
<tr>
<td>1280 kHz</td>
<td>42.7</td>
<td>73.7</td>
</tr>
<tr>
<td>1024 kHz</td>
<td>42.0</td>
<td>73.0</td>
</tr>
<tr>
<td>768 kHz</td>
<td>39.8</td>
<td>72.8</td>
</tr>
<tr>
<td>512 kHz</td>
<td>32.1</td>
<td>63.1</td>
</tr>
</tbody>
</table>

### Table 11. Distortion measurements at 500 Hz

<table>
<thead>
<tr>
<th>PDM clock</th>
<th>SNR at -31 dBFS</th>
<th>SNR at 0 dBFS (extrapolation)</th>
</tr>
</thead>
<tbody>
<tr>
<td>2048 kHz</td>
<td>43.1</td>
<td>74.1</td>
</tr>
<tr>
<td>1280 kHz</td>
<td>42.1</td>
<td>73.1</td>
</tr>
<tr>
<td>1024 kHz</td>
<td>42.1</td>
<td>73.1</td>
</tr>
<tr>
<td>768 kHz</td>
<td>40.1</td>
<td>71.1</td>
</tr>
<tr>
<td>512 kHz</td>
<td>29.8</td>
<td>60.8</td>
</tr>
</tbody>
</table>
Figure 5. Distortion measurements at 1000 Hz

Table 12. Distortion measurements at 1000 Hz

<table>
<thead>
<tr>
<th>PDM clock</th>
<th>SNR at -31 dBFS</th>
<th>SNR at 0 dBFS (extrapolation)</th>
</tr>
</thead>
<tbody>
<tr>
<td>2048 kHz</td>
<td>38.7</td>
<td>69.7</td>
</tr>
<tr>
<td>1280 kHz</td>
<td>38.5</td>
<td>69.5</td>
</tr>
<tr>
<td>1024 kHz</td>
<td>38.4</td>
<td>69.4</td>
</tr>
<tr>
<td>768 kHz</td>
<td>37.1</td>
<td>68.1</td>
</tr>
<tr>
<td>512 kHz</td>
<td>31.2</td>
<td>62.2</td>
</tr>
</tbody>
</table>
3.3.2 Speech signal

The test signal is a speech sequence, played at a nominal level of 90 dB SPL. The MEMS microphone is placed at 30 cm from the acoustic source. The digital signal captured at the output of the PDM2PCM library for different \textit{mic\_gain} values is shown in Figure 6.

**Figure 6. Comparison of speech levels**

![Comparison of speech levels](image)

<table>
<thead>
<tr>
<th>\textit{mic_gain} parameter value</th>
<th>Digital speech level</th>
</tr>
</thead>
<tbody>
<tr>
<td>0 dB</td>
<td>-54 dBFS</td>
</tr>
<tr>
<td>12 dB</td>
<td>-42 dBFS</td>
</tr>
<tr>
<td>24 dB</td>
<td>-30 dBFS</td>
</tr>
<tr>
<td>36 dB</td>
<td>-18 dBFS</td>
</tr>
<tr>
<td>48 dB</td>
<td>-8 dBFS</td>
</tr>
</tbody>
</table>

Table 13. Average speech signal level at PDM2PCM library output
4 Application description

4.1 Module integration example

4.1.1 Library initialization

Once the memory is allocated, some routines must be called to initialize the PDM2PCM library static memory:

- **PDM_Filter_Init()** has to be called each time the processing in the audio is stopped and started.
- **PDM_Filter_setConfig()** has to be called at least once before processing start, to set configurable parameter

Furthermore, as the PDM2PCM library runs on STM32 devices, CRC HW block must be enabled and reset.

The static and dynamic parameters structures must be allocated. Their types are defined in pdm2pcm_glo.h header. Example of allocation:

```c
/* Enables and resets CRC-32 from STM32 HW */
__HAL_RCC_CRC_CLK_ENABLE();
CRC->CR = CRC_CR_RESET;

PDM_Filter_Handler_t PDM1_filter_handler;
PDM_Filter_Config_t   PDM1_filter_config;

/* Initialize PDM Filter structure */
PDM1_filter_handler.bit_order = PDM_FILTER_BIT_ORDER_LSB;
PDM1_filter_handler.endianness = PDM_FILTER_ENDIANNESS_BE;
PDM1_filter_handler.high_pass_tap = 2122358088;
PDM1_filter_handler.out_ptr_channels = 1;
PDM1_filter_handler.in_ptr_channels  = 1;
PDM_Filter_Init((PDM_Filter_Handler_t *)(&PDM1_filter_handler));

PDM1_filter_config.output_samples_number = 16;
PDM1_filter_config.mic_gain = 24;
PDM1_filter_config.decimation_factor = PDM_FILTER_DEC_FACTOR_64;
PDM_Filter_setConfig((PDM_Filter_Handler_t *)&PDM1_filter_handler, &PDM1_filter_config);
```

4.1.2 Module execution

The run time process can start when the hardware is configured and the PDM2PCM library is initialized and configured.

At each new interrupt, when enough bits have been buffered, the PDM2PCM filter routine can be called. Between two consecutive calls to this filter routine, the dynamic parameters can be changed.
do
{
    /* process current frame */
    PDM_Filter(&pdm_buffer[0], &pcm_buffer[0], &PDM1_filter_handler);

    /* change volume setting */
    PDM1_filter_config.mic_gain = 12;
    PDM_Filter_setConfig((PDM_Filter_Handler_t *)&PDM1_filter_handler,
                        &PDM1_filter_config);
}

4.1.3 Module APIs calls

The flow is shown in Figure 7, and the steps listed below:

1. As explained above, PDM2PCM handler and configuration structures have to be allocated, as well as PDM input and PCM output buffers. CRC must be enabled in order to unlock the library.
2. The PDM filter handler parameters must now be set to the desired values, and PDM.Filter_Init() function called.
3. The PDM filter configuration parameters must now be set to the desired values, and PDM.Filter_setConfig() function called.
4. The PDM input bit stream is read from the proper interface, packed byte by byte.
5. Call to PDM.Filter() function will execute the PDM2PCM algorithm.
6. The PCM output audio stream can now be written in the proper interface.
7. If needed, the user can change configuration parameters and call the PDM.Filter_setConfig() function to update the library configuration.
8. If the application and the PDM input stream are still running, the processing loop goes back to step 5, otherwise it ends.
9. Once the processing loop is over, allocated memory has to be freed.
Figure 7. Module flow-chart

1. Memory allocation
   CRC enable and reset

2. Filter handler parameter initialization
   \texttt{PDM\_Filter\_Init()}

3. Filter configuration parameter initialization
   \texttt{PDM\_Filter\_setConfig()}

4. PDM stream acquisition

5. \texttt{PDM\_Filter}

6. PDM stream write

7. New configuration needed?
   \begin{tabular}{ll}
   Yes & 8 \texttt{PDM\_Filter\_setConfig()}
   
   No & 8 Sample to proceed?
   \end{tabular}

8. Sample to proceed?
   \begin{tabular}{ll}
   Yes & 9 Memory freeing
   
   No & 8 \texttt{PDM\_Filter\_setConfig()}
   \end{tabular}

9. Memory freeing
5 Revision history

Table 14. Document revision history

<table>
<thead>
<tr>
<th>Date</th>
<th>Revision</th>
<th>Description of changes</th>
</tr>
</thead>
<tbody>
<tr>
<td>06-Jul-2018</td>
<td>1</td>
<td>Initial release</td>
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