STM32L4 Cube HAL

PDM protocol, audio recording

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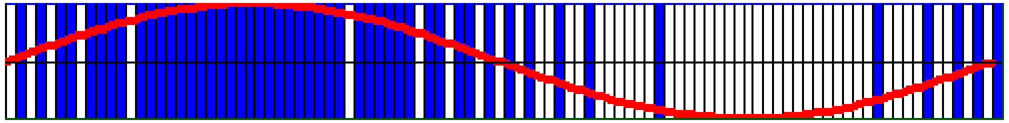
Pulse-density modulation



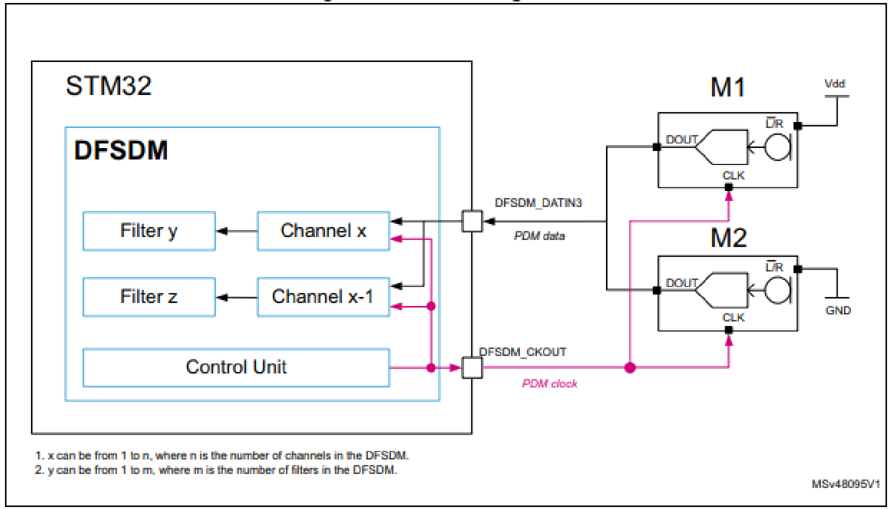
- Is a digital binary signal used to represent an analog signal.
- As the number of logic “1” grows, means the voltage level is higher.
- Based on your input clock source, it generates the output => synchronous.
- You need a filter to convert the stream of ‘0’ and ‘1’ into a meaningful signal.

Pulse-density modulation

- You can use a single data line to get the two microphone outputs (stereo).
- At each edge of clock, one microphone puts the data on the data line.
- What is the source of clock of the DFSDM module?
- And, what is frequency of clock generated by this block?
- At the output of MICs, we have a stream of ‘0’ and ‘1’,

given to the DFSDM block.

Figure 21. Stereo configuration

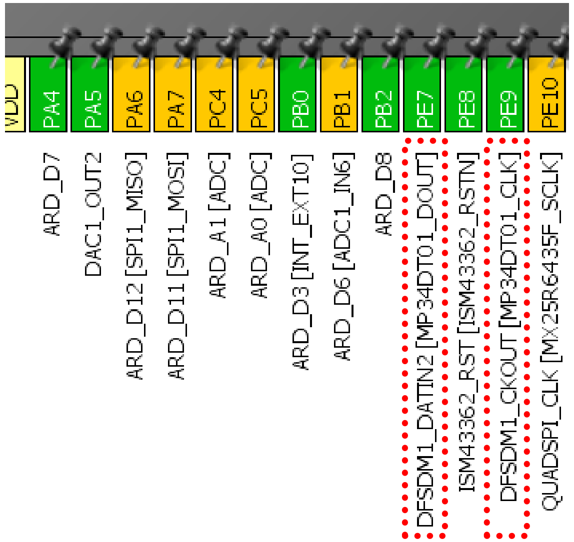


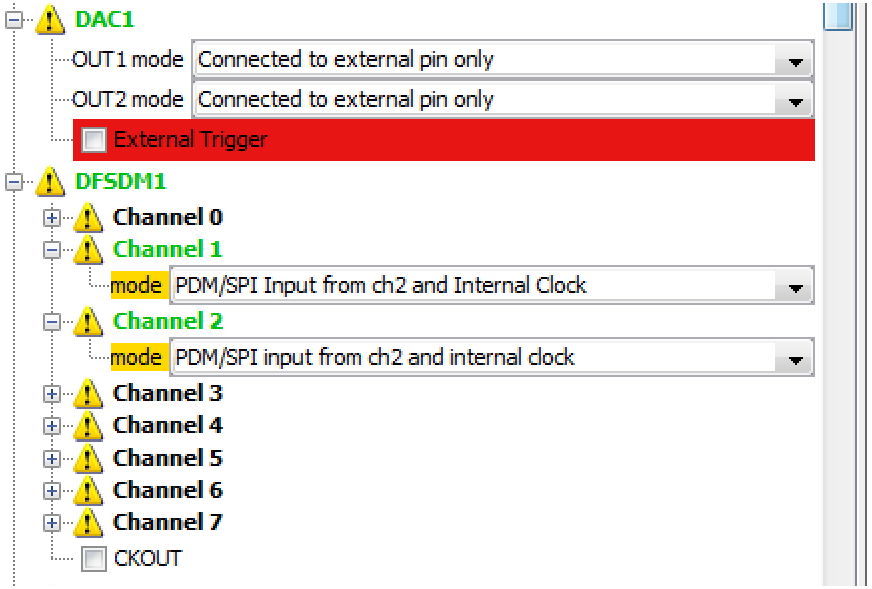
Pulse-density modulation

- Once the data is converted, it would be an analog signal but in digital format.
- An ‘Offset’ value will be automatically subtracted from the data result. You must tune it with reasons!
- The output data generated by this block is big > 30bits. Downgrade it into your desired size by ‘Right shifting’. You must choose the amount of shift !
- Then, feed your DAC by this value. Ensure the variation is within the expected range of your DAC.
- A higher produced sound quality => a higher grade.

How to use CubeMx

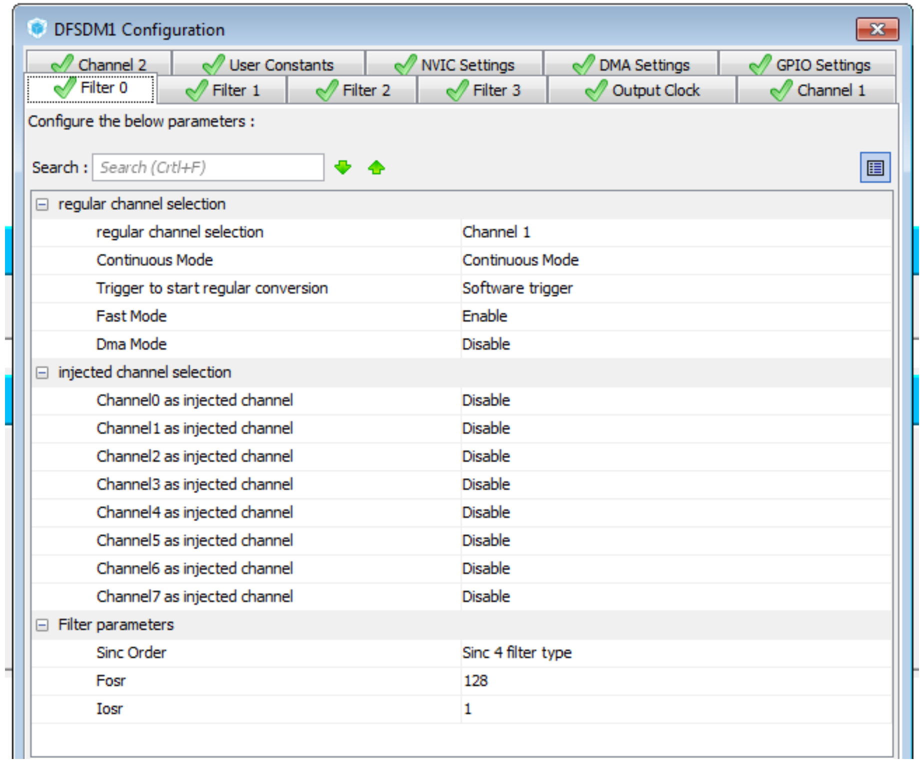
- Enable the block by selecting the CH2 and Internal clock.





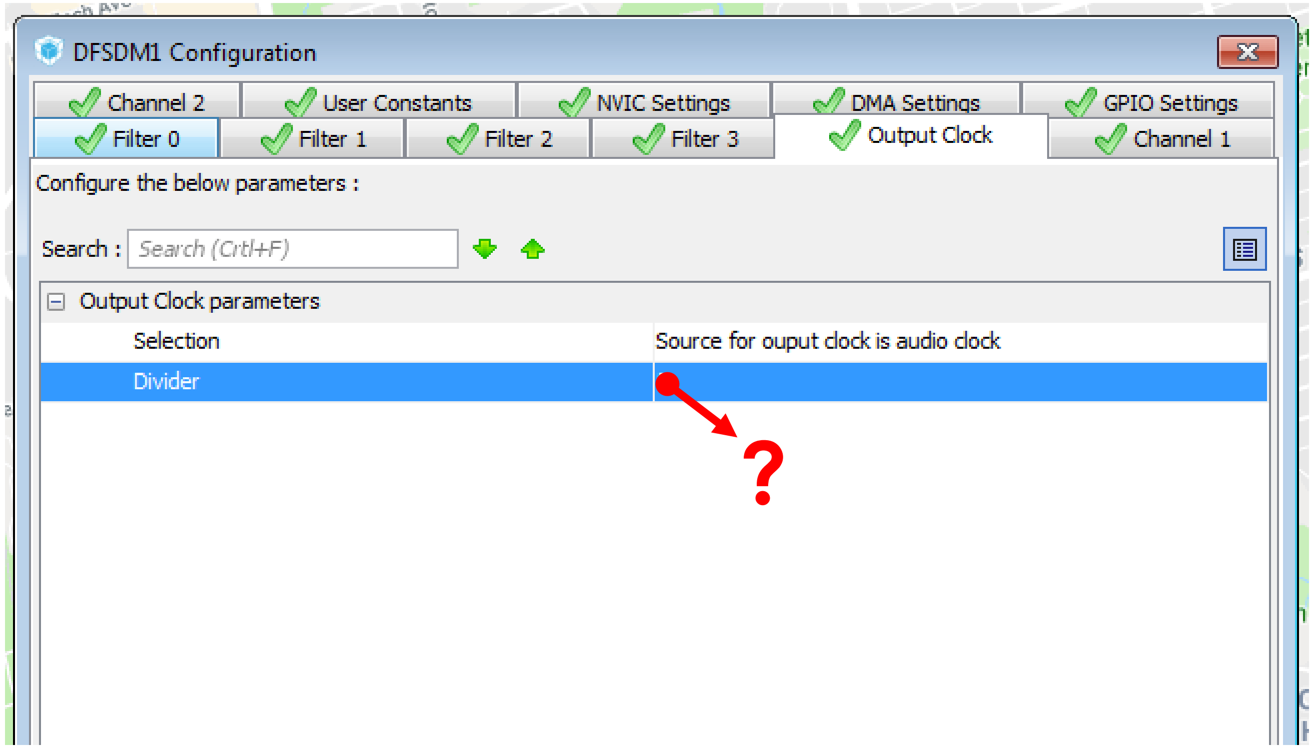
How to use CubeMx

- Enable two filters as below, one for channel 1 and another for channel 2:



How to use CubeMx

- Enable two filters as below, one for channel 1 and another for channel 2:



How to use CubeMx

- Now, you need to start the filter in regular mode. Then wait to GetValue to give you HAL_OK and the data.
- Or, if you enabled the interrupt, call the start function in interrupt mode and get the data in the interrupt handler.
- The base project is based on the polling mode. You are not obliged to use IT or DMA modes.

dos and don'ts

- You have two options:
  - 1- Recording >10 Sec data once the BTN pressed, and then play all when the BTN is pressed again.
    - Write ‘0’ to your DAC when you finish playing the audio.
    - You have more freedom (time) to apply any processing you want, but remember you have a big size of data.
  - 2- Play immediately what you sample => live operation, as long as the BTN is pressed.
    - Once you finish playing, the DAC output should be set to 0 volt.

- In both options, you are required to analyze the range of audio digital values and try to make best output ranges by changing the offset, bit resolution and sampling.
- Part of the grading (almost 2 points) would be based on the sound quality and efforts have been done for this goal. Groups will be sorted from best to worst (competition style).