(EasyDMA has finished transferring as well). Attempting to restart before receiving the STOPPED event may result in unpredictable behavior.

8.14.3 Decimation filter

In order to convert the incoming data stream into PCM audio samples, a decimation filter is included in the PDM interface module.

The input of the filter is the two-channel PDM serial stream (with left channel on clock high, right channel on clock low). Its output is 2 × 16-bit PCM samples at a sample rate lower than the PDM clock rate at a ratio depending on the RATIO register.

The filter stage of each channel is followed by a digital volume control, to attenuate or amplify the output samples in a range of -20 dB to +20 dB around the default (reset) setting, defined by $G_{PDM,default}$. The gain is controlled by the GAINL and GAINR registers.

As an example, if the goal is to achieve 2500 RMS output samples (16-bit) with a 1 kHz 90 dBA signal into a -26 dBFS sensitivity PDM microphone, do the following:

- Sum the PDM module's default gain ($G_{PDM,default}$) and the gain introduced by the microphone and acoustic path of his implementation (an attenuation would translate into a negative gain)
- Adjust GAINL and GAINR by the above summed amount. Assuming that only the PDM module influences the gain, GAINL and GAINR must be set to $-G_{PDM,default}$ dB to achieve the requirement.

With $G_{PDM,default}$ = 3.2 dB, and as GAINL and GAINR are expressed in 0.5 dB steps, the closest value to program would be 3.0 dB, which can be calculated as:

```
GAINL = GAINR = (DefaultGain - (2 * 3))
```

Remember to check that the resulting values programmed into GAINL and GAINR fall within MinGain and MaxGain.

8.14.4 EasyDMA

Samples will be written directly to RAM, and EasyDMA must be configured accordingly.

The address pointer for the EasyDMA channel is set in SAMPLE.PTR register. If the destination address set in SAMPLE.PTR is not pointing to the Data RAM region, an EasyDMA transfer may result in a HardFault or RAM corruption. See Memory on page 13 for more information about the different memory regions.

Note: The value programmed in SAMPLE.PTR register should be 32bit aligned

The DMA transfer supports Stereo (left and right 16-bit samples) and Mono (left only) data transfer as configured in the OPERATION field of the MODE register. The samples are stored little endian.

MODE.OPERATION	Bits per sample	Result stored per RAM word	Physical RAM allocated (32-bit words)	Result boundary indexes in RAM	Note
Stereo	32 (2x16)	L+R	ceil(SAMPLE.MAXCNT/2)	R0=[31:16]; L0=[15:0]	Default
Mono	16	2xL	ceil(SAMPLE.MAXCNT/2)	L1=[31:16]; L0=[15:0]	

Table 47: DMA sample storage

The destination buffer in RAM consists of one block, the size of which is set in SAMPLE.MAXCNT register. Format is number of bytes used by the samples that will be stored in the memory. The physical RAM allocated is always:

```
(RAM allocation, in bytes) = SAMPLE.MAXCNT * 2;
```

(but the mapping of the samples depends on MODE.OPERATION.