AN12590

i.MXRT600 PDM MEMS Microphone Audio Path Optimal Settings

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Application Note

by: NXP Semiconductors

Contents

1 Introduction

1.1 Goal

- Find the optimal setting for the PDM-DECIMATION audio path. The embedded decimator is designed to be 24-bit, so it is recommended to use the 24-bit mode. For the 16-bit mode, utilize the right shifter and the saturation configurations to protect from rollover while maintaining 16 bits by downshifting by 8.
- The goal is to align the 94 dB SPL acoustic reference level with the digital microphone sensitivity in the digital domain, at the decimator output.
    - — RT600 DMIC specifications summary (see Knowles microphone specifications).
- AOP is 120 dB SPL.
- SNR is 64 dB.
- The sensitivity at 94 dB SPL is -26 dBFS.
- The dynamic range is 90 dB.

The performances in the digital domain must be good enough to match the microphone dynamic range:

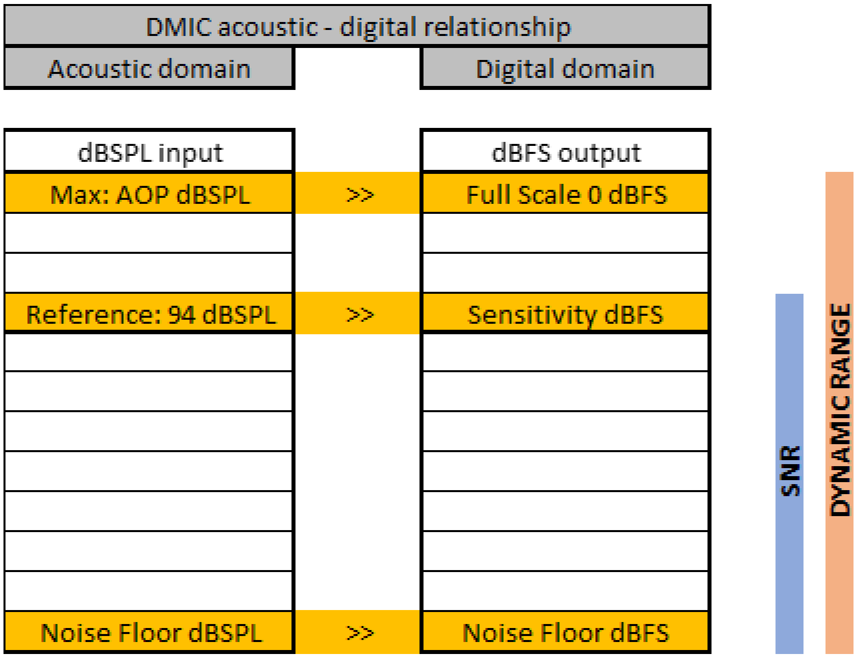


Figure 1. Acoustic/digital domain relationship

1.2 PDM-to-PCM conversion

1.2.1 PDM modulation

The Pulse-Density Modulation (PDM) is a data format used at the output of MEMS digital microphones (for example, Knowles SPH0641LM4H-1). Such microphones are used in mobile devices, because they output digital data, are immune to interferences, allow for flexible topology and board layout, and they have low noise level and cost.

The density of PDM pulses corresponds to the amplitude of the analog signal at the microphone input. The PDM 1-bit stream is encoded at a very high sampling rate. The PDM modulation uses noise-shaping techniques to lower the noise in the audio band and reject it out of band. The out-of-band noise is filtered by a low-pass filter to avoid noise aliasing in the audio band.

1.2.2 PDM decimation

The Pulse-Coded Modulation (PCM) represents the data in signed integer values in a multi-bit format, at audio sampling rates. The PCM format is used for signal-processing operations on audio streams.

The PDM-to-PCM conversion can be split into different operations:

- The PDM data must be downsampled by the OSR factor to match the expected audio sampling rate fs.
- The out-of-band noise generated by the PDM modulation must be removed to avoid noise aliasing in the audio base band.

Implementation: the digital decimator consists of a cascade of several filters:

- CIC filter with programmable GAIN and OSR
- Decimator stages sub-sampling down to fs
- Anti-aliasing low-pass filter

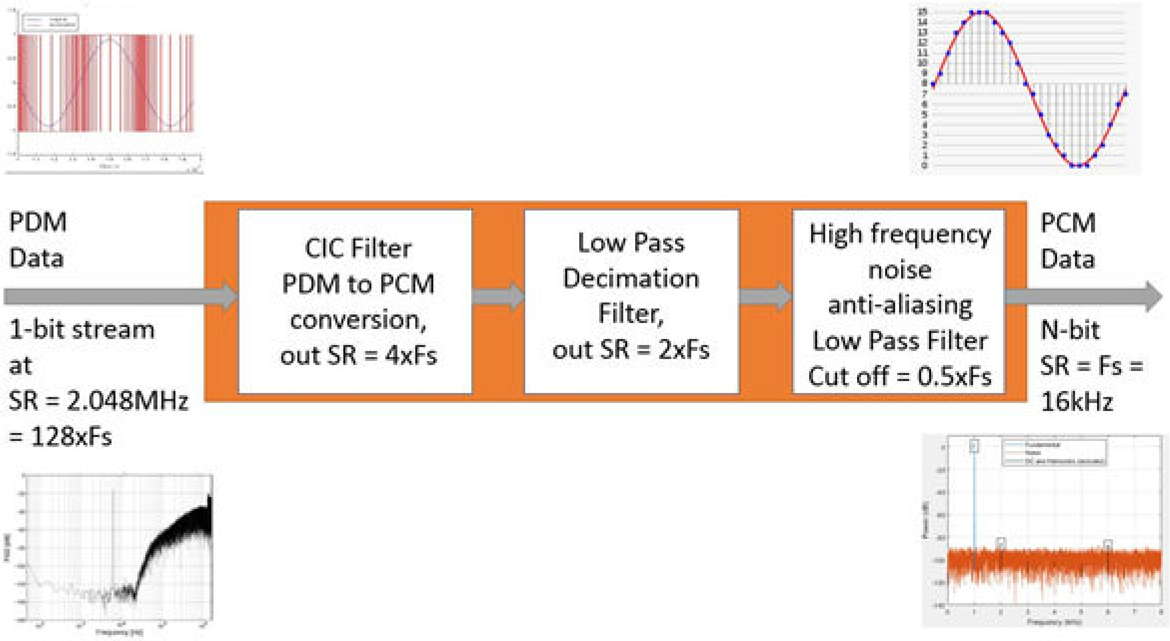


Figure 2. PDM to PCM conversion

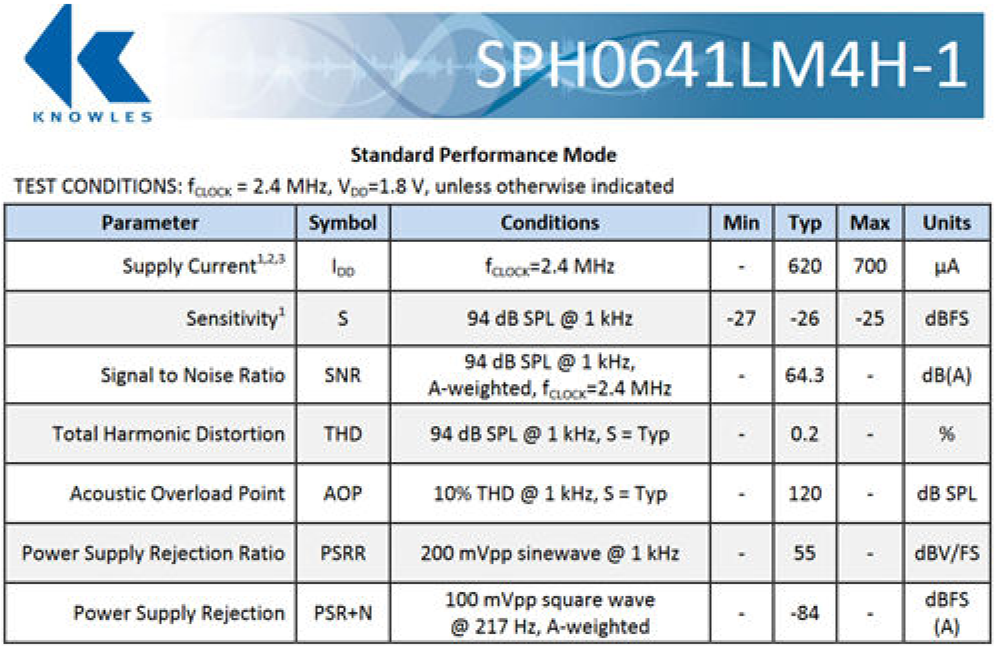
2 RT600 hardware setup

2.1 RT600 main board

The main RT600 board referenced in this document is X-MIMXRT685-EVK, Rev. B.

2.2 RT600 microphone board

The RT600 microphone board uses a digital microphone from Knowles (SPH0641LM4H-1). Here are the specifications:



SPH0641LM4H-1

Standard Performance Mode

TEST CONDITIONS: f_{CLOCK} = 2.4 MHz, V_{DD}=1.8 V, unless otherwise indicated

Parameter	Symbol	Conditions	Min	Typ	Max	Units
Supply Current[1,2,3]	I_{DD}	f_{CLOCK}=2.4 MHz	-	620	700	µA
Sensitivity[1]	S	94 dB SPL @ 1 kHz	-27	-26	-25	dBFS
Signal to Noise Ratio	SNR	94 dB SPL @ 1 kHz, A-weighted, f_{CLOCK}=2.4 MHz	-	64.3	-	dB(A)
Total Harmonic Distortion	THD	94 dB SPL @ 1 kHz, S = Typ	-	0.2	-	%
Acoustic Overload Point	AOP	10% THD @ 1 kHz, S = Typ	-	120	-	dB SPL
Power Supply Rejection Ratio	PSRR	200 mVpp sinewave @ 1 kHz	-	55	-	dBV/FS
Power Supply Rejection	PSR+N	100 mVpp square wave @ 217 Hz, A-weighted	-	-84	-	dBFS (A)

Figure 3. Specification of digital microphones used on the RT600 platform

The DMIC VDD is provided by the RT600 main board. The VDD value is 1.8 V.

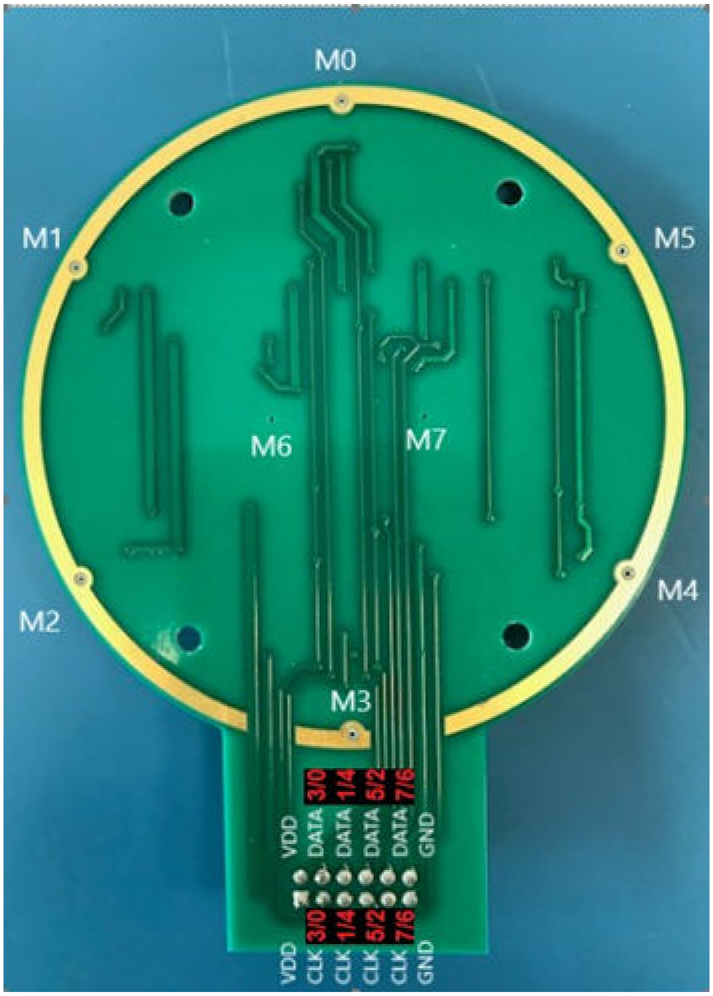


Figure 4. Top view and channel mapping of the RT600 DMIC daughter board

3 Software settings

3.1 Simulation results

The correct gain setting in the 24-bit configuration to match the microphone sensitivity is “dmic_channel_cfg.gainshft = 6”. The output signal peak level is -26 dBFS for an input PDM stimulus level of -26 dBFS. This means that the full chain has 0-dB gain.

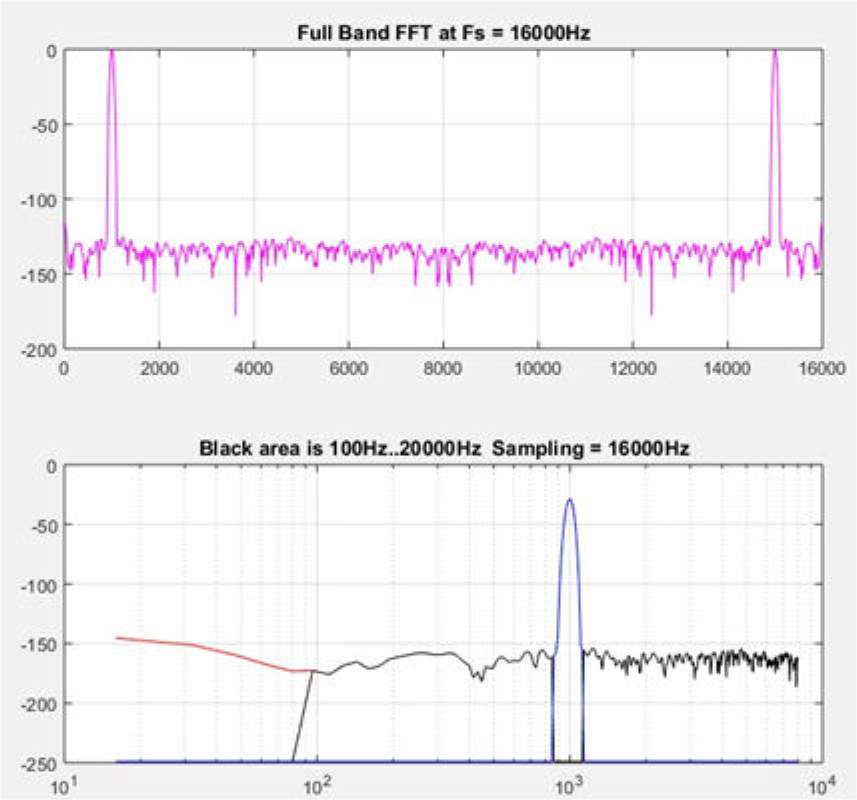


Figure 5. Decimator simulation

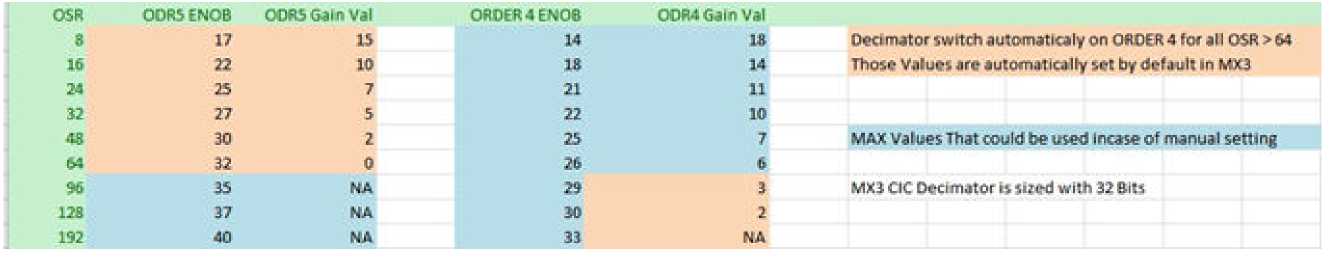
The gain shift is designed to compensate for the ENOB reduction and to use a full 32-bit dynamic range on the CIC. The half-band filters then produce 24-bit output. Theoretically, the gain shift value can be calculated using these formulas:

gain shift=32- ENOB

ENOB= ceiling(log_2 *(*OSR^{Order} *))+B_Input*

where: - PDM Input bit depth, = 2

In the presented case, there is a CIC of 5th order when OSR is 32 and ENOB is 27 (assumption). The gain shift value should then be 5 (fine-tuned to 6 with the microphone used). For more details, see Table 4. The 16-bit mode can be achieved by downshifting by 8 and enabling the saturate configuration.



OSR	ODR5 ENOB	ODR5 Gain Val	ORDER 4 ENOB	ODR4 Gain Val	
8	17	15	14	18	Decimator switch automaticaly on ORDER 4 for all OSR > 64
16	22	10	18	14	Those Values are automatically set by default in MX3
24	25	7	21	11	
32	27	5	22	10	
48	30	2	25	7	MAX Values That could be used incase of manual setting
64	32	0	26	6	
96	35	NA	29	3	MX3 CIC Decimator is sized with 32 Bits
128	37	NA	30	2	
192	40	NA	33	NA	

Figure 6. CIC settings

3.2 Software configuration

The source code is in the dmic_i2s_dma.cfile. This code extract is the configuration of the 24- and 16-bit modes:

```
/* Common parameters: */
dmic_channel_cfg.osr = 32U;
dmic_channel_cfg.gainshft = 6U;

#ifdef DMIC_24BITS

    dmic_channel_cfg.saturate16bit = 0U;
    dmic_channel_cfg.enableSignExtend = true;
    dmic_channel_cfg.post_dc_gain_reduce = 0U;

#else

/* DMIC_16BITS                                */
/* Right shifter and saturation configurations */
dmic_channel_cfg.saturate16bit = 1U;
    dmic_channel_cfg.enableSignExtend = false;
    dmic_channel_cfg.post_dc_gain_reduce = 8U;

#endif
```

4 Hardware validation

4.1 Digital validation

The PDM signal is generated by the APx.

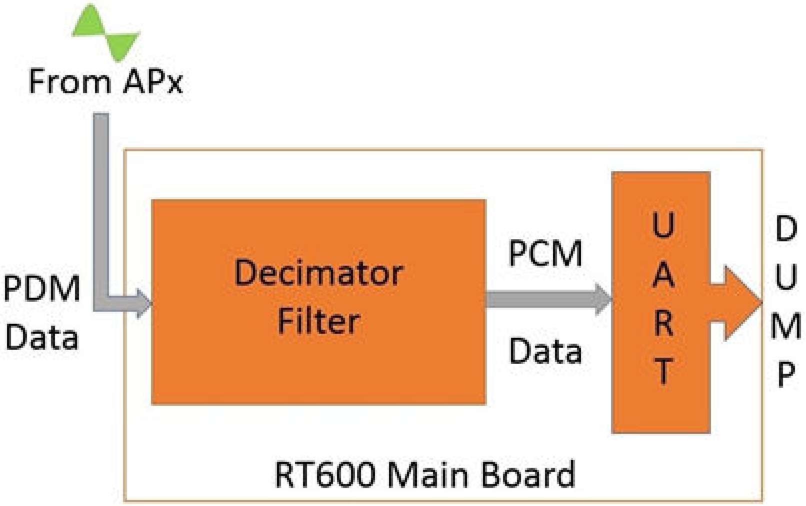


Figure 7. PDM-to-PCM hardware validation

4.2 Validation setup

- Perform the electrical measurements using the APx PDM generator.
- Generate a PDM signal at 1 kHz and -26 dBFS using the PDM interface of the APx.
- The recording is done in the PCM format.

- The PDM clock is provided by the RT600 main board to the APx analyzer (the APx analyzer is in the slave mode).
- The APx analyzer settings are:
  - The PDM clock frequency from the RT600 main board is 2.048 MHz.
  - The APx signal generator sine wave is 1 kHz at the level of -26 dBFS and Fs of 16 kHz.

4.3 24-bit mode

Using the Matlab script on the PCM dump, you get these results:

Amplitude:

- The average RMS (A_RMS) is -28.94 dB.
- The peak value (A_dBFS) is -25.93 dBFS (the chain gain is 0 dB).

The SNR is 108.13 dB.

The noise RMS is -137.07 dB.

The THD is -134 dB.

The DC is 0 %.

4.4 16-bit mode

Using the Matlab script on the PCM dump, you get these results:

Amplitude:

- The average RMS (A_RMS) is -28.94 dB.
- The peak value (A_dBFS) is -25.93 dBFS.

The SNR is 66.17 dB.

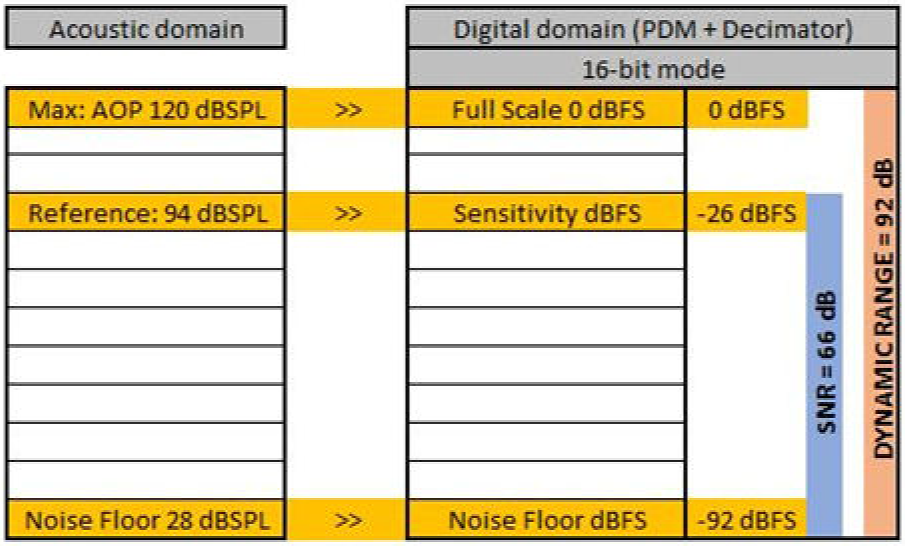
The noise RMS is -95.11 dB.

The THD is -96.33 dB.

The DC is 0 %.

4.5 Optimal setting for RT600 microphones

This table shows the relationship between the acoustic and digital domains for the 16-bit mode, based on digital validation:

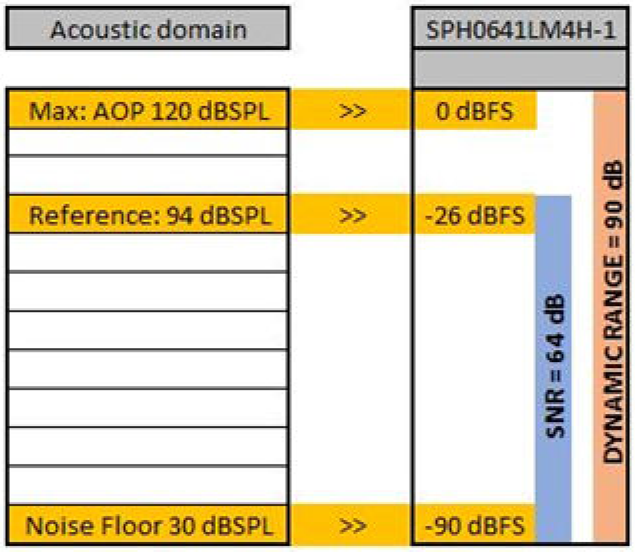


Acoustic domain		Digital domain (PDM + Decimator) 16-bit mode	
Max: AOP 120 dBSPL	>>	Full Scale 0 dBFS	0 dBFS
Reference: 94 dBSPL	>>	Sensitivity dBFS	-26 dBFS
Noise Floor 28 dBSPL	>>	Noise Floor dBFS	-92 dBFS

SNR = 66 dB

DYNAMIC RANGE = 92 dB

This table shows the application of the SPH0641LM4H-1 digital microphone specifications:



Acoustic domain		SPH0641LM4H-1
Max: AOP 120 dBSPL	>>	0 dBFS
Reference: 94 dBSPL	>>	-26 dBFS
Noise Floor 30 dBSPL	>>	-90 dBFS

SNR = 64 dB

DYNAMIC RANGE = 90 dB

Conclusion:

The SPH0641LM4H-1 microphone performance (SNR of 64 dB, dynamic range of 90 dB, and AOP of 120 dB SPL) fits this case in the 16-bit mode, which is sufficient to handle this microphone specification. The calculation must be reconsidered for any other microphone devices, especially in case of higher AOP and dynamic range characteristics.

5 Acoustic measurements in reverberant test room

When compared to the hardware digital results, acoustic measurements imply limitations due to the measurement environment (non-anechoic test room) and equipment performances (analog front end, speaker characteristics).

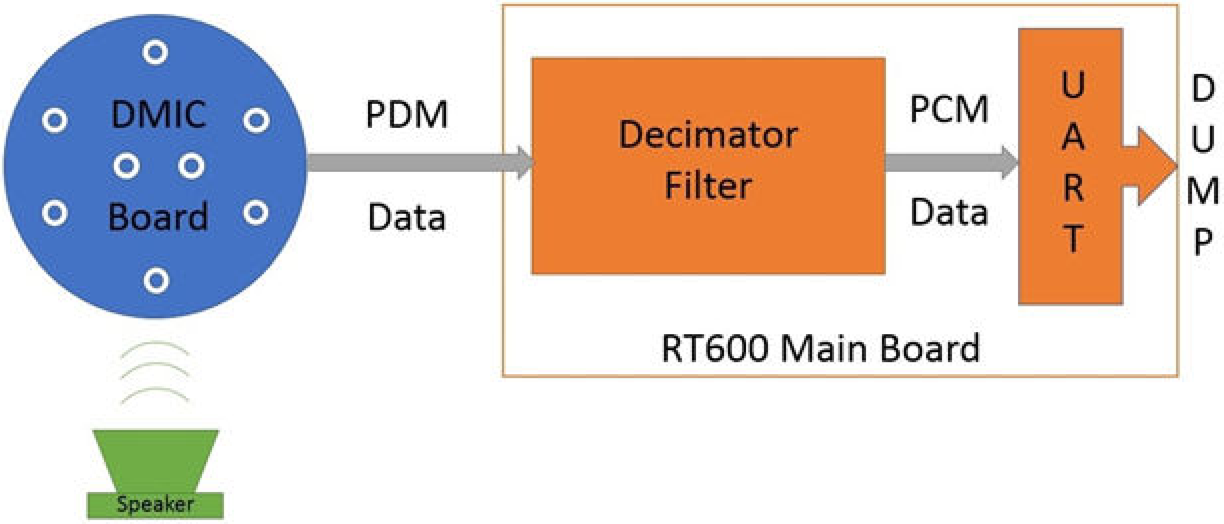


Figure 10. Acoustic measurements

5.1 Acoustic setup

- The test room maximum noise level is 30 dB(A).
- The room is in a reverberant configuration.
- The calibration is performed at 1 kHz and 94 dB SPL level, 2.54 cm above the center of the DMIC daughter card.
- The stimulus is a sine-wave signal at 1 kHz and 94 dB SPL settings.
- The boards' position is flat.
- The speaker and boards are at the same height.
- The loudspeaker is placed 80 cm from the DUT.

Figure 11. Lab configuration

5.2 Acoustic results

Table 1 compares:

- the (Matlab) results at the RT600 PCM output for both the 24- and 16-bit modes.
- the measurement of the DMIC data by the APx audio analyzer on the DMIC daughter card.

Despite some spread during the measurements, the results for the 24- and 16-bit modes are very similar. The sensitivity results match the digital validation results and Knowles DMIC specifications. The SNR performance is limited by the test room noise floor. Distortion comes from the amplification/speaker at a 94-dB SPL level.

Table 1. RT600 acoustics results summary

Device/measure	RT600 24-bit mode	RT600 16-bit mode	DMIC SPH0641 through APx525
Sensitivity	-25.2 dBFS	-26.4 dBFS	-26.5 dBFS
SNR	61.5 dB	58.2 dB	59 dB
Distortion	0.32 %	0.38 %	0.42 %
DC offset	0 %	0 %	0 %

6 References

1. ITU-T P.341: *Transmission characteristics for wideband digital loud speaking and hands-free telephony terminals*
2. IEEE1329: *IEEE Standard Method for Measuring Transmission Performance of Speakerphones*

7 Abbreviations

Table 2. Abbreviations

AOP	Acoustic Overload Point
APx	Audio Precision APx525 Audio Analyzer
CIC	Cascaded Integrator-comb Filters
dBFS	Used Convention: Sine Wave full scale at 0 dBFs ó -3 dB RMS
DUT	Device Under Test
ENOB	Effective Number of Bits = SNR/(20*Log(2))
OSR	Oversampling Ratio
PCM	Pulse-Code Modulation
PDM	Pulse-Density Modulation
SLR	Sending Loudness Rating
SNR	Signal to Noise Ratio = 20*Log(2^ENOB)
SPL	Sound Pressure Level
UART	Universal Asynchronous Receiver Transmitter

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