Application Note

Analog Microphone and ADC System in Far-field Application

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ABSTRACT

Far-field audio applications such as AI speakers and soundbars, AI TVs, and other voice-activated products need the following:

- Wide DR (Dynamic Range)
- High AOP (Acoustic Overload Point)
- Low THD (Total Harmonic Distortion) even for very loud signal
- Low equivalent input noise
- Small form factor with low power consumption

All of these electrical specifications have led to the voice commander getting precise voice command recognition without any unexpected errors. This application report discusses and shows the benefits and strengths of analog microphones with Texas Instruments' TLV320ADC51x0 system in far-field audio application.

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1 Near-Field and Far-Field

1.1 Definition

Near-field and far-field are components of the physical distance from the sound source, as shown in Figure 1-1. It depends on how far away a listener is from the sound-projecting object. At different distances, the acoustic energy the wavelength emits is heard differently. It is a major issue to comprehend these differences and design measurements precisely. Figure 1-1 implies that up to the second wave length is near-field, while from the second wave length is a range of far-field.

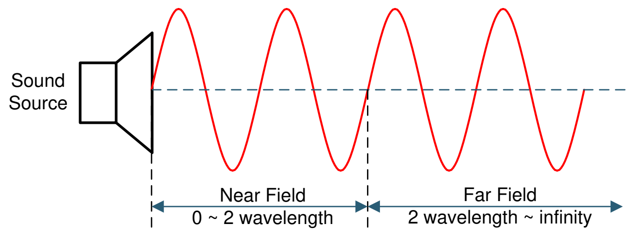

Figure 1-1. Near-Field and Far-Field from Sound Source

The essential idea within the range of a far-field is that 6 dB of sound pressure decreases for each doubling of distance away from the sound source. For the most of the voice recognition system, the commander stands within the range of a far-field.

2 Constituents in the Far-Field Application

For a more accurate voice-activated system, far-field applications must obtain these elements, especially when used for a voice recognition application.

Table 2-1 explains constituents that must be used for a better voice recognition system.

Table 2-1. Constituents of Far-field Application

APPLICATION NEEDED	SYSTEM SPECIFICATION NEEDED
• Crystal clear • Distortion-free for audio capture	Wide dynamic range microphone and ADC
Microphone close to speakers in loud, noisy environments	• High AOP (Acoustic Overload Point) • Low THD (Total Harmonic Distortion even for very loud signal)
Far-field recording	Low EIN (Equivalent Input Noise)
Portability	• Small form factor • Low power consumption

3 Understanding of Digital and Analog Microphone

A microphone is a kind of transducer that converts sound pressure into electrical signal. There are two kinds of microphones in collection with the sound waves. One of them is digital type, and the other is known as analog type. This section contains the representing format of microphones, a digital PDM (Pulse Density Modulation) microphone, and an analog microphone with ADC.

3.1 Digital PDM Microphone System

A digital PDM microphone works 1-bit stream as output with the direct output of the Sigma-delta modulator. The sample rate of the PDM is typically between a few hundred kHz to 3.072 MHz. This Sigma-delta modulator needs a decimation filter so that the PDM data can process further. The decimation filter is implanted in either the codec, or the DSP where the PDM microphone is connected. This output from the filter sends out data at a lower sample rate of 16 and 48 kHz. This is a weaker point than an analog microphone with an ADC system shown in Section 3.2. However, a PDM microphone does have some advantages over an I2S microphone. The PDM microphone typically requires only two signal lines (PDM clock and PDM data) for audio interface, while I2S requires three or four lines (bit clock, word clock, audio data, and occasionally master clock). This leads to a beneficial idea where the hardware engineers are easily able to minimize the physical interface line.

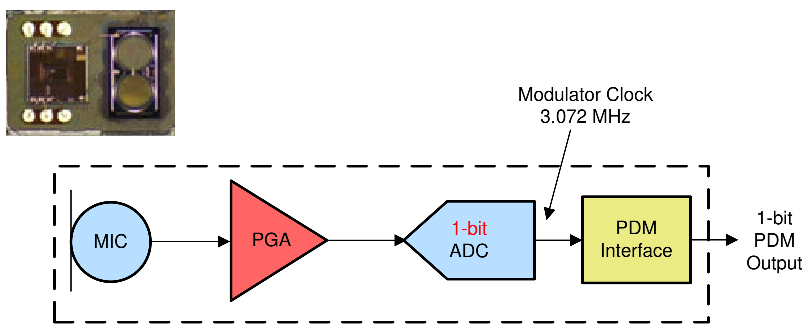

Figure 3-1. Digital PDM Microphone for 1-bit Output

3.2 Analog Microphone with ADC System

An analog microphone with an ADC system based on Texas Instruments' TLV320ADC51x0 uses multi-bit modulation ADC to project audio data with a range higher than 24-bit, along with I2S (or PCM, TDM, DSP) interface, as shown in Figure 3-2. This is unlike the PDM microphone, which uses the 1-bit modulation with PDM interface. For the analog microphone with ADC to project as shown in Figure 3-2, the I2S audio signal of ADC output requires one or two more interface lines than the two of the PDM signals. The analog microphone has advantages in in-band (20 Hz to 20 KHz) and out-of-band (above 20 KHz) for quantization noise characteristics (further explained in Section 4). This is because while the digital PDM microphone uses a 1-bit ADC modulator with a 3.072 MHz clock, an analog microphone with ADC (Texas Instruments' TLV320ADC51x0) uses multi-bit architecture with a modulator clock of 6.144 MHz, which allows it to have superior qualities in quantization noise density.

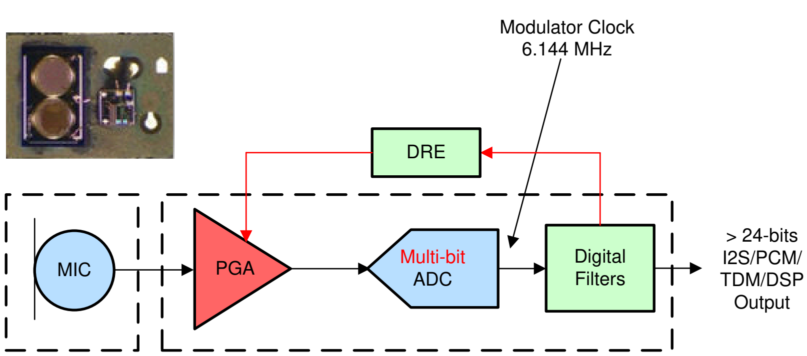

Figure 3-2. Analog Microphone with ADC for Multi-bit Output

4 Quantization Noise Density for Each Microphone

The 1-bit modulator in the digital PDM microphone has architecturally high quantization noise that limits the PDM microphone SNR and dynamic range compared to what can be achieved with an analog microphone along with multi-bit ADC. The TLV320ADC51x0 uses a multi-bit modulator which has in-band quantization noise up to 15 times lower, and better, than the typical 1-bit modulator. Moreover, the total quantization noise of the TLV320ADC51x0 multi-bit modulator is 30 times lower than the total typical quantization noise of a 1-bit PDM modulator. The high quantization noise in in-band has worse voice recognition, and every 6 dB improvement in microphone and system performance can be improved by the record distance and doubled sensitivity.

The higher total quantization noise of a 1-bit PDM modulator drastically limits the maximum high amplitude signal that it can record with low distortion compared to a multi-bit ADC. This limitation makes the digital microphone the incorrect choice to be used for soundbars, HDTVs, professional speakers, and so forth, where the microphone is required to record very loud speaker output with low distortion so that the echo cancellation can be achieved with good quality, and voice command detection successfully is enabled.

Figure 4-1 shows the typical quantization noise density difference in all bands for the 1-bit PDM modulator and multi-bit analog modulator.

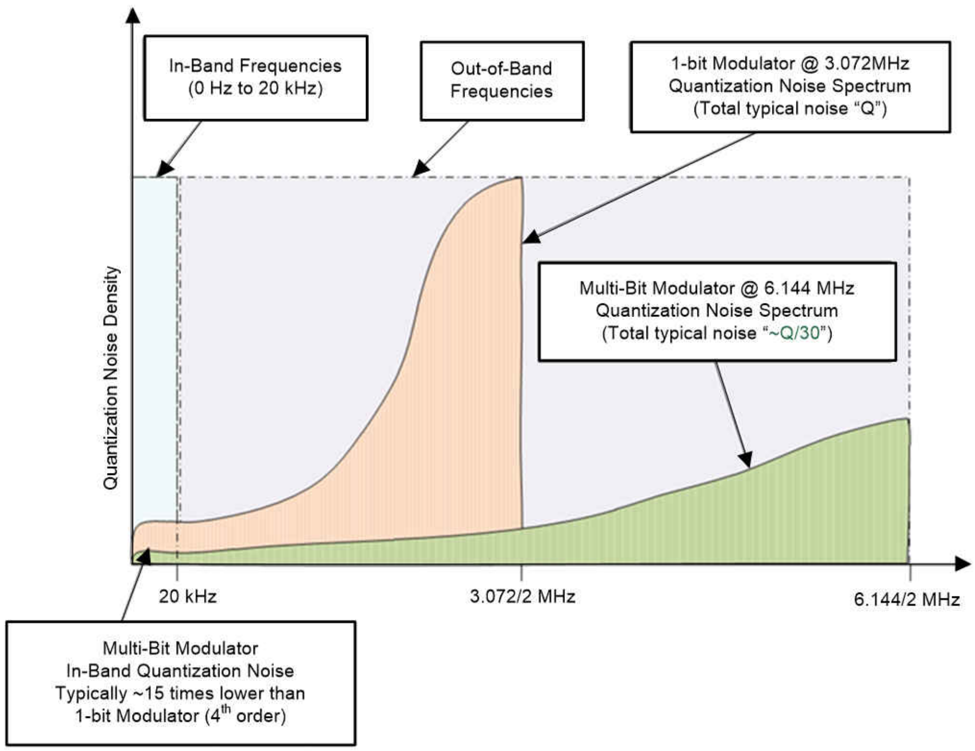

Figure 4-1. Quantization Noise Density

5 Dynamic Range in Far-Field

5.1 DR

Dynamic range (also defined as DR, DNR, or DYR) describes the ratio between the level at which the greatest sound converted with very low distortion to the softest sound in the system. In other words, the audio field uses dynamic range to depict the ratio of the softest sound to the loudest sound level in the musical instrument system. Therefore, dynamic range works as the SNR (Signal-to-Noise Ratio) for the case when the signal is the loudest possible in the system.

5.2 DR in Microphone

As shown in Section 5.1 for dynamic range, better SNR, lower THD (Total Harmonic Distortion), and better AOP (Acoustic Overload Point) create high dynamic range. Here, AOP is the point at which the microphone can record the loud sound with distortion as high as 10%. The microphone equivalent input noise (EIN) is used to define microphone SNR to the point at which the microphone no longer effectively projects the difference between the actual sound signal level and microphone self-noise level. It no longer works effectively as a sound pressure sensor at this point.

5.3 DRE in the TLV320ADC5140

The TLV320ADC5140 device integrates an ultra-low noise front-end PGA with 120 dB dynamic range performance with a low noise, low-distortion, multi-bit delta-sigma (ΔΣ) ADC with a 108 dB dynamic range. The dynamic range enhancer (DRE) is a digitally assisted algorithm to boost the overall channel performance. The DRE monitors the incoming signal amplitude and adjusts the internal PGA gain accordingly and automatically. The DRE achieves a complete channel dynamic range as high as 120 dB. At a system level, the DRE scheme enables far-field, high-fidelity recording of audio signals in very quiet environments, and low-distortion recording in loud environments. This algorithm is implemented with very low latency (a few µs loop time), and all signal chain blocks are designed to minimize any audible artifacts that can occur resulting from dynamic gain modulation.

The block diagram in Figure 5-1 describes the complete signal chain performance with a high-performance analog microphone.

As shown in Figure 5-1, if the system uses the high performance analog microphone with effective dynamic range performance as high as 114 dB, along with a low-cost, 108 dB ADC without DRE, then the overall signal chain dynamic range gets limited to 106.8 dB (114 dB–7.20 dB) due to the 108 dB ADC. However, once the DRE scheme of the TLV320ADC5140 is enabled, an overall signal chain dynamic range of 112.97 dB (114dB–1.03dB) can be achieved in a very economical manner, even after using a low-cost, 108 dB ADC.

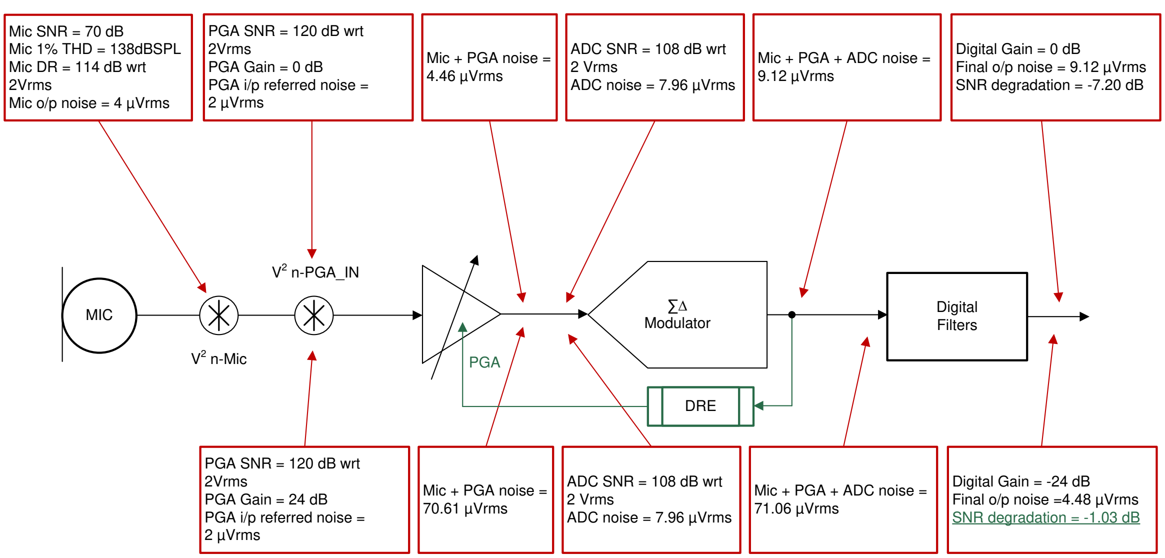

Figure 5-1. DRE Block Diagram

5.4 DRE Performance in TLV320ADC5140

Figure 5-2 and Figure 5-3 show the THD+N measurement results with and without DRE. The measurement graph with DRE enabled (DRE threshold set to -36 dB) shows DR improvement of 12 dB and it also improves the record distance and sensitivity by four times.

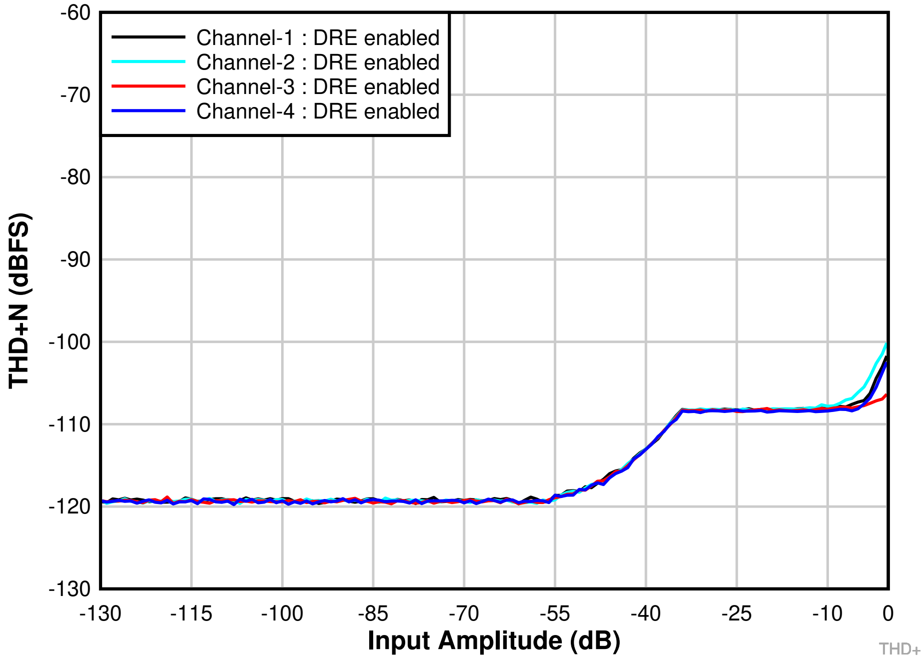

Figure 5-2. THD+N Performance Measurement with DRE Enabled

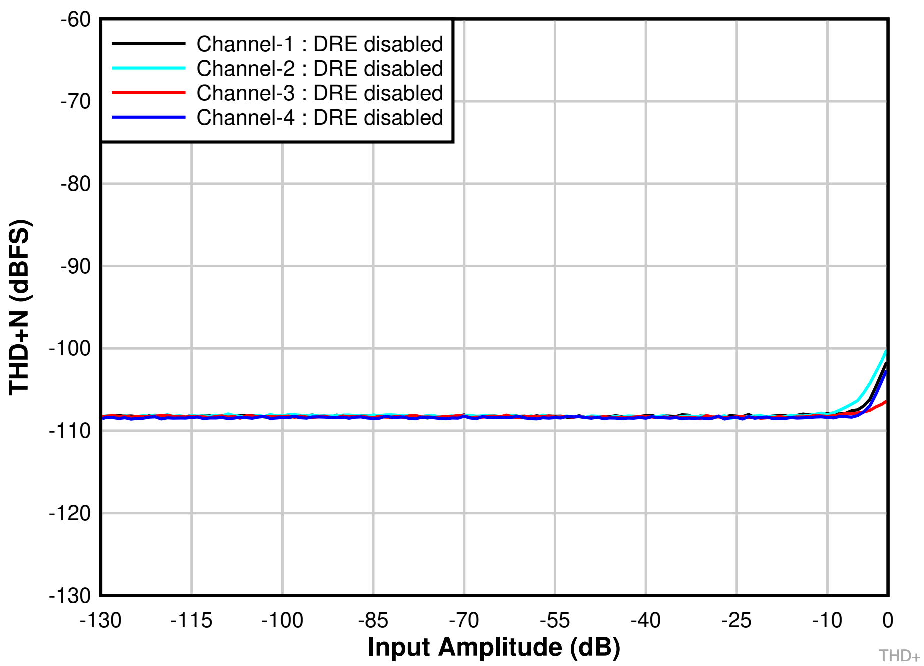

Figure 5-3. THD+N Performance Measurement with DRE Disabled

6 Design of Any Microphone with the TLV320ADC51x0

6.1 Structure of the TLV320ADC51x0

Texas Instruments' TLV320ADC51x0 is four analog microphones, eight digital microphones, or a combination of both that an audio analog-to-digital converter can design as the input with four internal Sigma-delta ADCs and eight digital microphone interfaces. This enables the hardware designer to have access to flexible design structures, especially supporting the microphone array design.

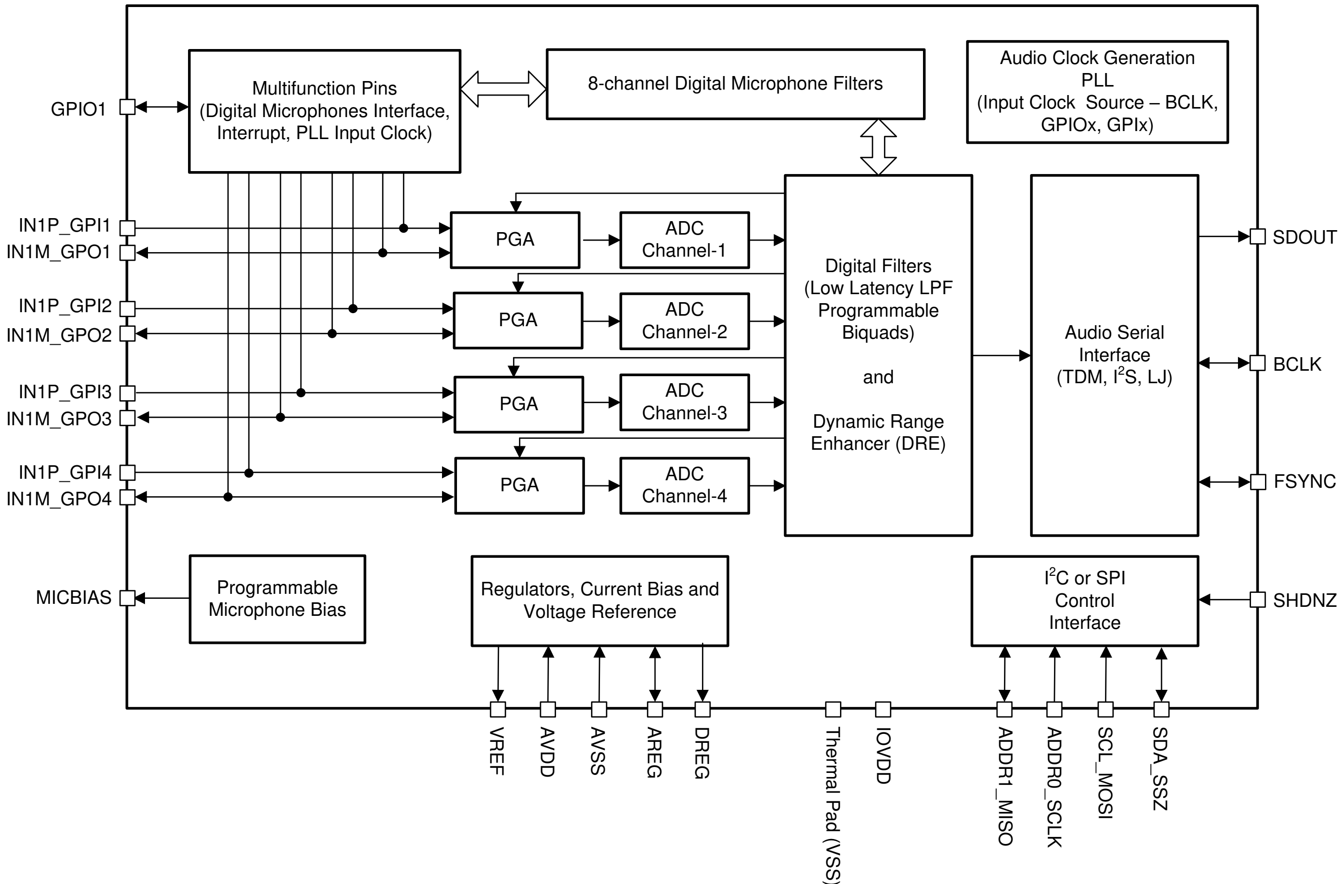

Figure 6-1. Internal Block Diagram of the TLV320ADC5140

6.1.1 Design Example 1: Only Analog Microphone System

A differential analog microphone can be formed up to four multi-channels. In this case, the output signal changes the front-end PGA into gain control, then uses high-performance, multi-bit sigma-delta AD to be converted by ADC. After this process, it is sent to the decimation filter.

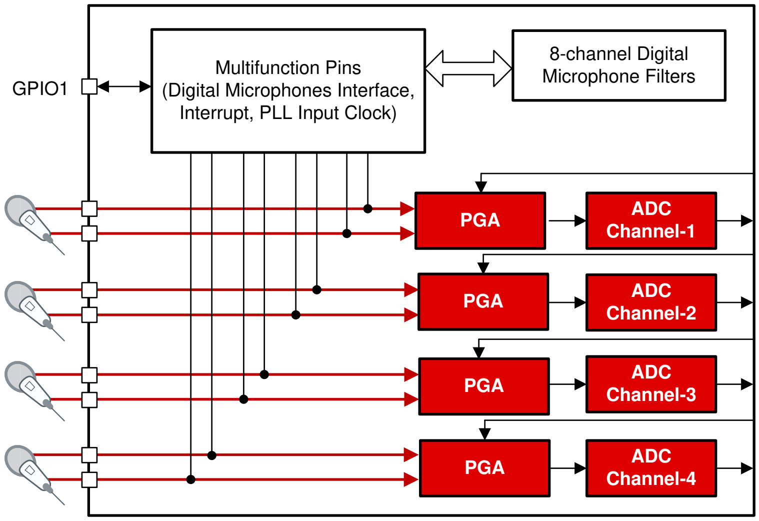

Figure 6-2. Analog Microphones System Block Diagram

6.1.2 Design Example 2: Only Digital Microphone System

The digital PDM microphone can be formed up to eight multi-channels. In this case, the PDM output from the microphone goes through PDM interface to be sent to the decimation filter.

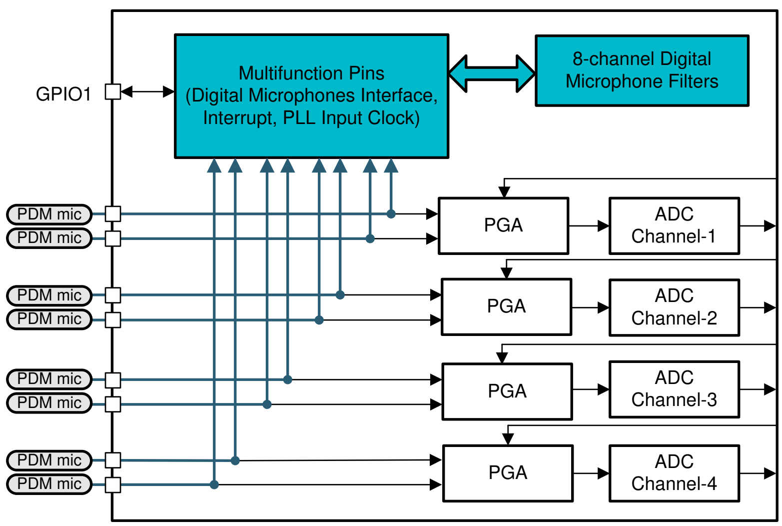

Figure 6-3. Digital PDM Microphones System Block Diagram

6.1.3 Design Example 3: Analog and Digital Microphone Combination System

The system can be formed as desired with analog and digital PDM microphones combined. In this case, each input goes through design example 1 and 2, and is gradually sent to the decimation filter.

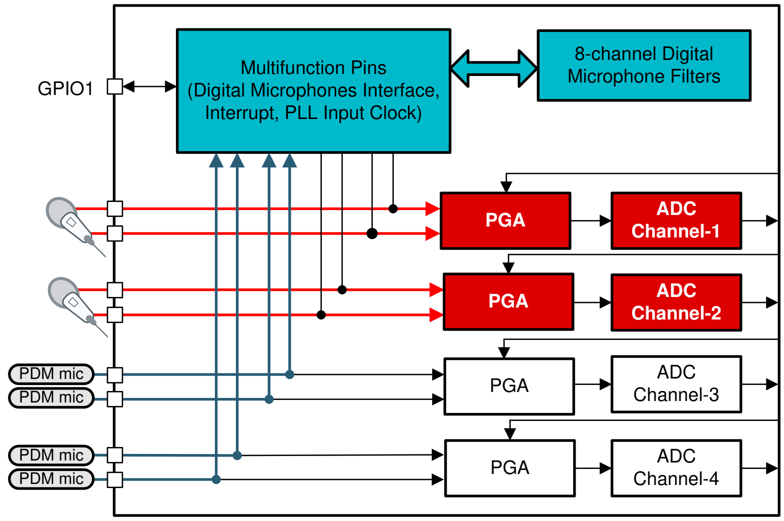

Figure 6-4. Combination Microphones System Block Diagram

7 Conclusion

By using the advantages of the analog microphones with Texas Instruments' TLV320ADC51x0 system that brings in the far-field application, quantization noise density due to the modulation improves, which brings up the height to the DR system. This leads to the near perfection of the voice recognition system that far-field application requires. The TLV320ADC51x0 is an analog or digital microphone input used for an audio analog-to-digital converter that can be designed to be accommodated as needed.

8 Revision History

Changes from Revision * (June 2019) to Revision A (May 2022)	Page
• Updated the numbering format for tables, figures, and cross-references throughout the document	2

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