

High Performance, Low Noise Studio Microphone with MEMS Microphones, Analog Beamforming, and Power Management

CIRCUIT FUNCTION AND BENEFITS

The circuit shown in Figure 1 implements a professional grade studio or live performance microphone using up to 32 analog MEMS microphones connected to op amps and a difference amplifier. The circuit is designed to be very low noise and its output is linear for acoustic inputs up to 131 dB sound pressure level (SPL). The $\pm 9\text{ V}$ and $+1.8\text{ V}$ power rails are generated from two voltage regulators powered from a single $+9\text{ V}$ battery.

The ADMP411 consists of a MEMS microphone element and an impedance matching amplifier. This microphone supports acoustic inputs up to 131 dB SPL and has a low frequency response that is flat to 28 Hz. These features make this microphone ideal for full bandwidth, wide dynamic range, audio capture applications, such as in a recording studio or on stage.

The ADA4075-2 op amps are used to perform several different functions in this circuit, including a summing amplifier and an allpass filter. This op amp is low noise, low power, and low distortion, making it a good choice for a battery-powered, high performance audio application.

The AD8273 converts the single-ended microphone signal into a differential signal that can be output on a standard microphone XLR connector. The gain setting resistors are internal to the difference amplifier, so it can create a high performance differential signal with no external components. The difference amplifier has very low distortion, low noise, and good output drive capability, making it a good choice for driving a differential microphone output.

The power supplies for this circuit are generated from an ADP1613 dc-to-dc switching converter and an ADP1720 linear regulator (see Figure 8 for schematic). The ADP1613, in a SEPIC-Ćuk configuration, generates the $\pm 9\text{ V}$ rails for the amplifiers, and the ADP1720 generates the 1.8 V supply of the MEMS microphones. These regulators efficiently generate the necessary voltage supplies with very low ripple.

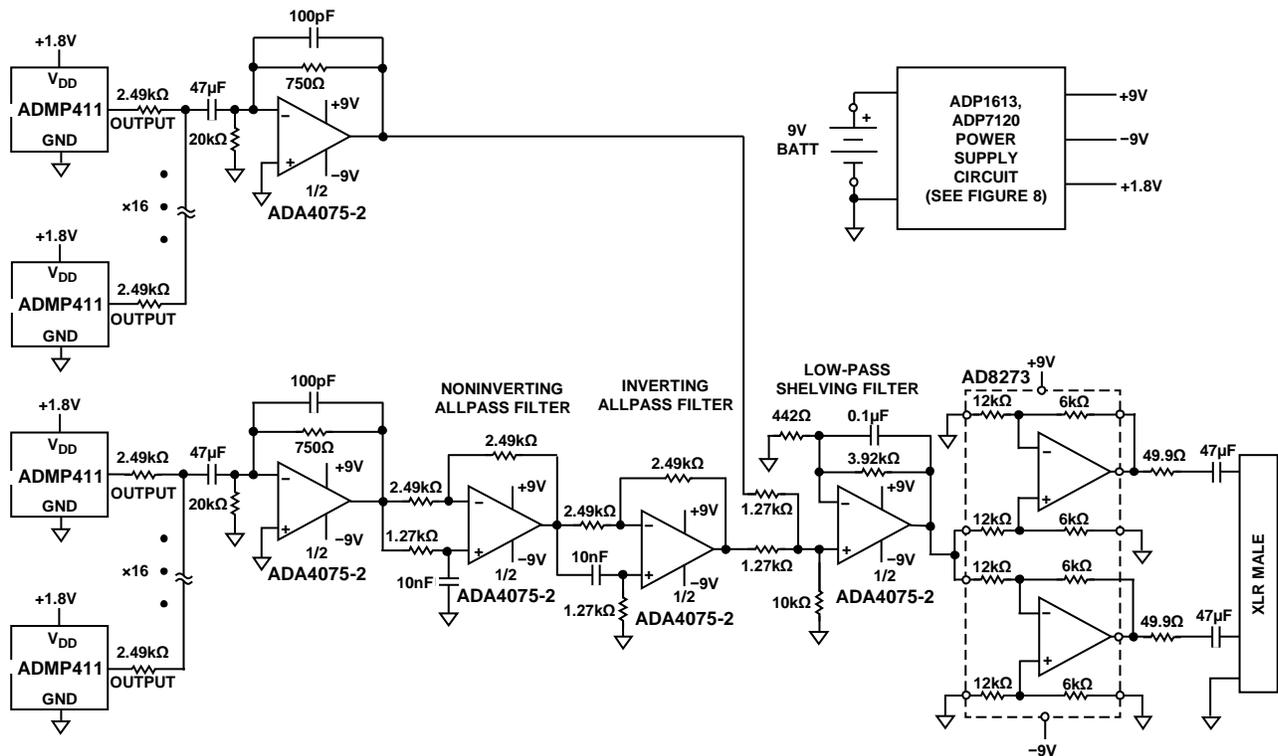


Figure 1. Microphone Circuit Diagram
(Simplified Schematic: All Connections and Decoupling Not Shown)

CIRCUIT DESCRIPTION

Table 1. Devices Connected/Referenced

| Product | Description |
|---------------------------|---|
| ADMP411 | High SPL microphone with extended low frequency response and analog output |
| ADA4075-2 | Ultralow noise amplifier at lower power |
| AD8273 | Very low distortion, dual channel, audio difference amplifier |
| ADP1613 | 650 kHz/1.3 MHz step up PWM dc-to-dc switching converter with 2.0 A current limit |
| ADP1720 | 50 mA, high voltage, micropower linear regulator |

An array of many ADMP411 MEMS microphones are closely spaced in this circuit to improve the overall signal-to-noise ratio (SNR) of the system to a point that it can be used for very low noise recording studio applications. The circuit can be used with either one or two clusters of microphones, depending on the desired directionality. Using two microphone arrays and some simple filtering enables some basic beamforming to achieve a supercardioid directional response; whereas the directional response of a single microphone array is basically omnidirectional, like the response of a single MEMS microphone.

Following the summing, filtering, and beamforming, the signal is still single-ended. The [AD8273](#) difference amplifier converts this single-ended signal into a differential signal that is output on the XLR jack.

This circuit can be powered from a single 9 V battery, and two regulators generate the ± 9 V and +1.8 V supplies. The amplifiers use the bipolar supply, while the MEMS microphones operate from the single low voltage 1.8 V supply.

Summing Amplifiers

The [ADA4075-2](#) is used as a summing amplifier in two places in this circuit. First, the outputs of each of the 16 ADMP411 microphones in the array are summed together in an inverting summing amplifier with a gain of 0.31. The input summing resistors are all 2.49 k Ω ; therefore, the contribution of each microphone to the output is equal. Every time the number of microphones in the array is doubled, the overall SNR increases by about 3 dB. The SNR increases because the microphone signals sum coherently, increasing the amplitude of the output by 6 dB for each doubling of the number of microphones, while the noise adds incoherently, adding 3 dB to the noise floor. The result is a net increase of 3 dB in the SNR. This circuit uses 16 ADMP411 microphones, each with an individual SNR of 62 dBA; therefore, after four doublings of the number of microphones, the SNR of the array improves by 12 dB to 74 dBA.

The ADMP411 has an output impedance of 200 Ω , so the 2.49 k Ω summing resistor reduces the output amplitude of each microphone signal by 7.5%, or 0.7 dB.

The feedback resistor in the summing amplifier is 750 Ω , which gives the amplifier an overall gain of 0.31 (–10 dB) following the summing node. The [ADA4075-2](#) is unity-gain stable, so this amplifier can be used for a gain <1. If another op amp is used in this circuit, check its closed-loop gain to ensure that it is stable at this low gain.

This reduction in level ensures the output at the XLR plug is similar to a typical recording microphone. A microphone with a sensitivity above about –30 dBV may be considered too hot and therefore difficult to use in a typical recording setup. If a microphone with higher sensitivity is desired, then the size of this feedback resistor can be increased, thereby increasing the gain. Other than maintaining typical microphone output levels, only the supply voltages limit the output levels.

The input of the [ADA4075-2](#) is also ac-coupled because the ADMP411 output is biased at 0.8 V. The output of this stage, and the rest of the amplifier stages, are biased at 0 V.

The second [ADA4075-2](#) summing amplifier is used either to sum two in phase arrays or to sum one microphone array with another delayed and inverted array for beamforming processing.

Polarity

The amplifier summing the signals from the 16 ADMP411 microphones is inverting. This inversion is done to preserve the polarity of the input acoustic signal. The ADMP411 has an acoustically inverted output, meaning that a positive pressure input results in a negative output voltage. This signal is inverted in the first amplifier stage so that it is not inverted with respect to the acoustic input at the XLR output jack. All other stages in the signal path are noninverting, except for the inversion necessary in one of the two microphone paths for the beamforming processing.

Beamforming

Beamforming involves processing the output of multiple microphones (or in this case, multiple microphone arrays) to create a directional pickup pattern. For recording and live sound applications, it is important that the microphone only picks up sound from one direction, such as from the singer or instrument, and attenuates the sound that is off the main axis. Beamforming is implemented in this design using analog delays, an equalization filter, and a summing amplifier.

As shown in Figure 2, a two element array is set up by placing two microphone boards a distance, *d*, apart. A cardioid pattern (see Figure 3) is achieved by delaying the signal from one array board by the amount of time it takes sound to travel between the two boards, and subtracting this delayed signal from the signal from the first microphone array board. With this type of spatial response, the microphone rejects sounds from the sides and rear, while picking up sounds incident to the front of the microphone.

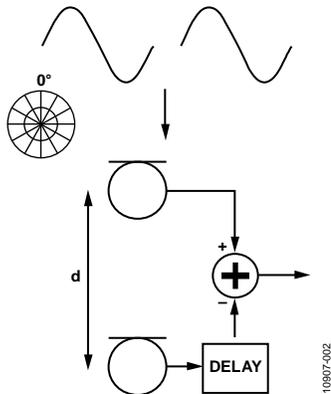


Figure 2. Endfire Beamforming Array

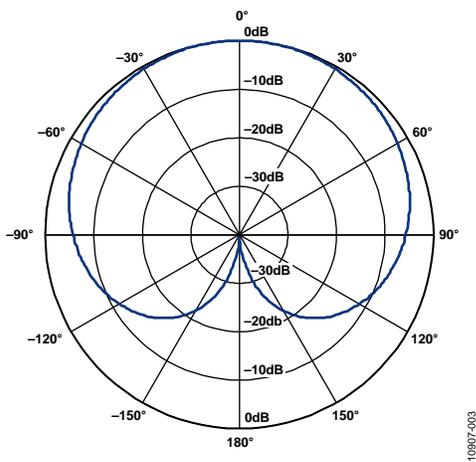


Figure 3. Cardioid Response

Allpass Filters

In this design, the two microphone array boards are separated by about 18 mm. To create a perfect cardioid pattern, the output of the rear microphone array needs to be delayed by 52.4 μs. A series of two allpass filters delay the signal, as shown in Figure 4.

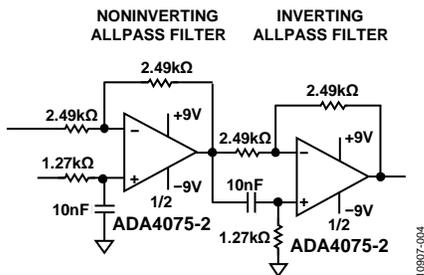


Figure 4. Noninverting and Inverting Allpass Filters

In the allpass filter design, there is a trade-off between the length of the delay and the frequency range over which the delay is constant. As the group delay at lower frequencies is increased, the group delay is constant over a smaller frequency range. In this circuit, each allpass filter contributes about 25 μs of delay up to about 4 kHz (with 10% tolerance). Above this frequency, the group delay at the filter output starts to decrease rapidly (see Figure 5), which quickly degrades the performance of the beamforming algorithm. At these higher frequencies, the primary contribution to the directionality of the design is the physical size of each microphone board (1.1 inch diameter).

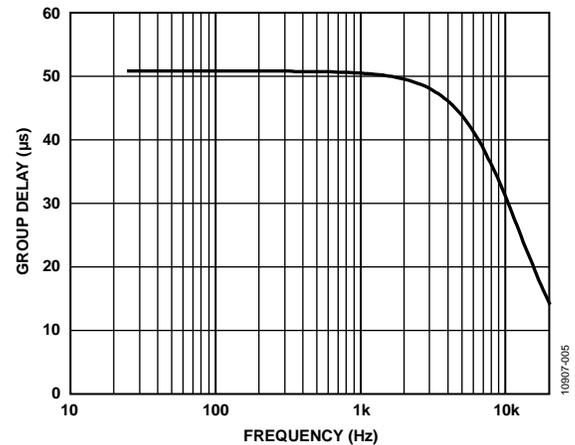


Figure 5. Two Stage Allpass Filter Group Delay

The delayed signal from the rear array also needs to be inverted so that it can be subtracted from the signal of the front array in the summing amplifier circuit. This inversion is done by implementing the first allpass filter with a noninverting output and the second allpass filter with an inverting output.

The group delay (*t_G*) of the filter is calculated by the equation

$$t_G = \frac{2RC}{1 + (\frac{f}{f_0})^2}$$

where $f_0 = 1/(2\pi RC)$.

In this design, R = 1.27 kΩ and C = 10 nF, so $f_0 = 12.5$ kHz. The design and equations for these allpass filters were taken from the [MT-202 Tutorial](#) and the Linkwitz Lab website.

Shelving Filter

The frequency response of the signal following the beamforming processing has a first-order roll-off at low frequencies. If this is uncorrected, the output of the microphone has poor low frequency response; therefore, a shelving filter is used to boost the response at these frequencies. The first-order, noninverting, low-pass shelving filter shown in Figure 6 has a gain of 20 dB at low frequencies with a shelf rising from 4 kHz down to 400 Hz. The frequency response is shown in Figure 7. This shelving filter flattens out at 20 dB of gain at about 100 Hz, so the response of the microphone still rolls off with a -6 dB/octave slope below this frequency.

This amplifier also sums the outputs from the front and rear microphone array boards.

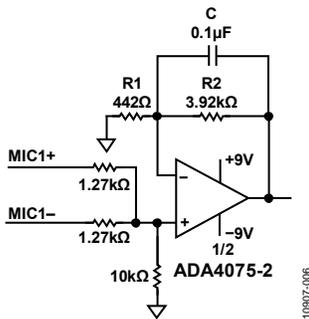


Figure 6. Shelving Filter

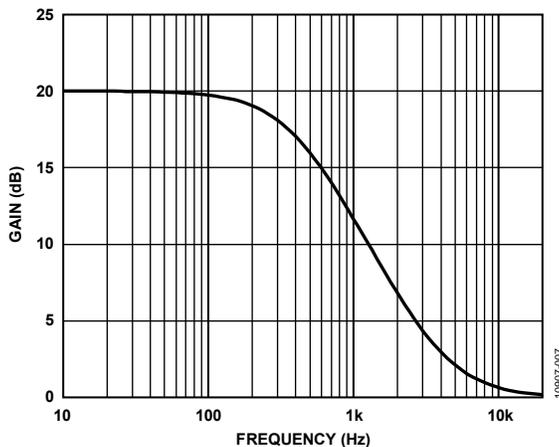


Figure 7. Shelving Filter Response

The gain (V_{OUT}/V_{IN}) of the filter is calculated by the equation

$$\frac{V_{OUT}}{V_{IN}} = G_1 \sqrt{\frac{1 + \left(\frac{f}{f_2}\right)^2}{1 + \left(\frac{f}{f_1}\right)^2}}$$

where $G_1 = f_2/f_1 = 1 + R_2/R_1$.

The corner frequencies of the filter, f_1 and f_2 , are calculated by

$$f_1 = \frac{1}{2\pi CR_2}$$

And

$$f_2 = \frac{1}{2\pi C \left(\frac{R_1 R_2}{R_1 + R_2} \right)}$$

In this design, $R_1 = 442 \Omega$, $R_2 = 3.92 \text{ k}\Omega$, and $C = 0.1 \mu\text{F}$, so $f_1 = 406 \text{ Hz}$ and $f_2 = 4.1 \text{ kHz}$. The design and equations for this shelving low-pass filter were also taken from the Linkwitz Lab website.

If a single microphone array board is used in the design, then the allpass filter circuits and the shelving filter circuits are not needed, and the output of the summing amplifier can be connected directly to the [AD8273](#) for the single-ended-to-differential conversion.

Single-Ended to Differential Conversion

The [AD8273](#) difference amplifier performs the single-ended to differential conversion for the output signal. This amplifier is configured for a gain of 1 (0 dB) to keep the output level of the microphone from being too high. One of the internal amplifiers of the [AD8273](#) is set up in a noninverting configuration, and the other is inverting. Individually, each of these amplifiers has a gain of 0.5 (-6 dB), but the single-ended to differential conversion results in a $+6$ dB gain, resulting in the overall 0 dB gain in this sub circuit. The [AD8273](#) does not require any external components except for the power supply decoupling capacitors. All gain setting resistors are internal to the IC, so they are very well matched, and the output of the amplifier has very low distortion.

The [AD8273](#) has good output drive capability and can easily drive a highly capacitive load. This is necessary because the output might be connected to a long (multiple meters) XLR cable.

The outputs of the [AD8273](#) drive 49.9Ω series resistors and a $47 \mu\text{F}$ ac coupling capacitor. The series capacitor is necessary because the microphone can be connected to an input that provides 48 V of phantom power, which is often used as a bias and supply for an electret microphone. Phantom power can typically only supply less than 10 mA, which is more than this circuit uses, so it is not used as the supply. The maximum voltage rating of the capacitor is 63 V, so the circuit is protected from the 48 V bias that may be present on both the positive and negative outputs.

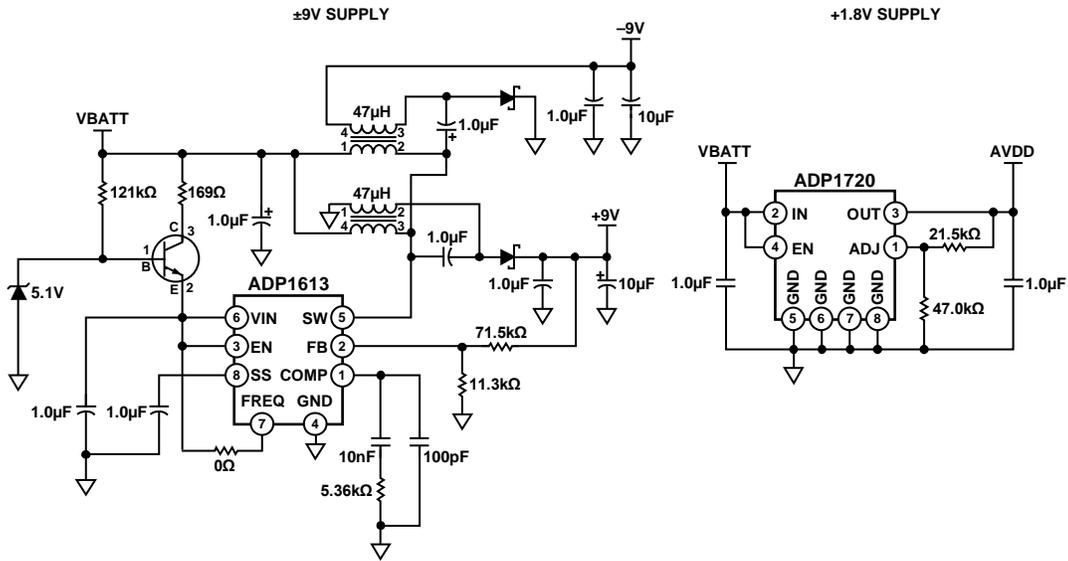


Figure 8. Details of Power Supply Circuit

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±9 V Power Supply

All of the amplifiers in this design are powered from ±9 V supplies. These voltages are generated from an ADP1613 in a SEPIC-Ćuk configuration. This circuit generates the positive and negative supply rails from a single input voltage as shown in Figure 8.

The design for the bipolar supply design was created using the ADIsimPower™ tools. The ADP161x SEPIC-Ćuk Design Tool takes some basic design parameters as an input and generates a schematic, bill of materials (BOM), and performance specifications for the given circuit. This design was created using the following specifications:

- V_{IN} minimum = 7.5 V
- V_{IN} maximum = 9.0 V
- V_{OUT} = 9.0 V
- I_{OUT} = 40 mA
- Ambient temperature = 55°C
- Design optimized for lowest cost

The total current drawn from the battery when the complete system is operational is 82 mA. Of this total current, about 17.5 mA is used by each microphone board, and 47 mA is used by the power board. With this load, a typical 9 V battery lasts about five hours.

1.8 V Power Supply

The 1.8 V supply is used to power the ADMP411 MEMS microphones and is generated from the ADP1720 linear regulator. This regulator drops the regulated +9 V supply to the necessary 1.8 V in a very small printed circuit board (PCB) footprint, requiring only one small (1 µF) bypass capacitor, on both the input and the output, and two resistors to set the output voltage. The ADMP411 draws a maximum current of 220 µA with a 1.8 V supply, so the highest current needed from this regulator output (with 32 ADMP411 devices connected) is 7.04 mA. At this maximum load, the regulator dissipates 50.7 mW:

$$P = (9\text{ V} - 1.8\text{ V}) \times 7.04\text{ mA} = 50.7\text{ mW}$$

ADDITIONAL CIRCUITRY

This section describes the function of additional components that are used in the circuit but are not part of the core circuit design. This primarily includes RF filtering and overvoltage protection circuitry

RF Filtering

Between the differential output of the AD8273 devices and the XLR plug, there are a few components intended to filter high frequency noise that may be picked up by the board or the microphone XLR cable. Each half of the differential signal has an LC filter, which removes electromagnetic interference (EMI) or RF noise. There is also a common-mode choke between the two differential legs to remove common-mode currents, while allowing differential currents to pass.

The connection of the microphone signals between the microphone boards and the power board also has an LC filter to reduce high frequency noise that may be picked up by the ribbon cable connection. A similar LC filter (see Figure 9) is also used on the output of all of the regulated voltage supplies.

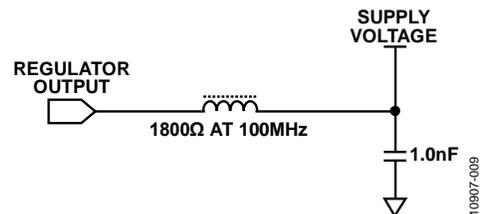


Figure 9. RF Noise Filter

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Spike Suppression

Back to back 5.1 V Zener diodes are connected between each side of the differential output signal and ground. These diodes are used to clamp voltage spikes greater than ±5.1 V that may be conducted on the output cable.

Low Battery Indicator

An LED is used to indicate when the battery voltage is low and should be replaced. The LED is placed in series between the battery positive terminal and the regulated 9 V supply. When the battery voltage drops below 9 V by more than the LED forward voltage (2 V), then the LED indicator turns on. The output voltage of many 9 V batteries starts to drop rapidly once it falls below 7 V.

PERFORMANCE AND MEASUREMENTS

Sensitivity Performance

The sensitivity of the microphone array is higher than that of each individual microphone because their outputs are summed. The ADMP411 has a sensitivity of -46 dBV. With the outputs of 16 microphones summed together and the other gains applied to the signal path, the sensitivity of the circuit with a single array is -33 dBV. If the beamforming part of the circuit is enabled when two array boards are used, the sensitivity is -27 dBV.

Frequency Response Performance

The frequency response of the microphone design is dependent on the orientation of the microphone. Figure 10 shows the frequency response for the sound incident on axis. Off axis, there is less of a low frequency roll-off caused by the beamforming processing, so the fixed shelving filter results in a higher low frequency boost.

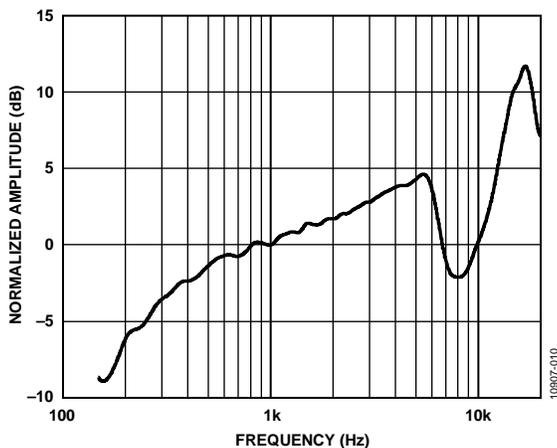


Figure 10. On Axis (0°) Frequency Response

The dip in the response at 8 kHz is a result of the beamforming processing; this dip is the point at which the inverted signal from the rear array board is most nearly in phase with the noninverted front microphone signal, resulting in significant attenuation of the overall subtracted output signal.

The summing amplifiers and single-ended to differential circuit all have very flat frequency response across the audio bandwidth, so they do not contribute significantly to the overall frequency response of the system design. The AD8273 has very good output drive capability, so it does not contribute a roll-off at high frequencies, even with long, highly capacitive XLR cables connected.

Noise Performance

Summing the outputs of 16 ADMP411 microphones theoretically improves the SNR by 12 dB from that of a single microphone. For each doubling of the number of microphones in the array, the SNR increases by about 3 dB. This SNR increase is because the signals add coherently and increase the amplitude by 6 dB, while the noise adds incoherently for a 3 dB increase in noise level.

The ADMP411 has an A-weighted 62 dB SNR, so the array of 16 microphones has a theoretical SNR of about 74 dBA. For a -33 dBV sensitivity, this means that the noise floor across a 20 kHz bandwidth is at -107 dBV (4.47 μ V rms), which is equal to an acoustic noise floor of 20 dB SPL. In practice, the measured noise floor is typically 1 to 2 dB higher than the 20 dB SPL, possibly because the spacing of the 16 microphones in the summing array is large enough that the individual microphone signals are not perfectly coherent.

The op amp and difference amplifier circuits following the microphones are significantly lower noise than the microphones themselves, so the microphones are the limiting factor in the noise of the overall design.

When two of the 16 microphone array boards are used for beamforming, the overall system SNR is degraded by about 3 dB. The SNR is about 71 dB when configured in a beamforming array, which is an equivalent acoustic noise floor (or equivalent input noise) of 21 dB SPL.

Total Harmonic Distortion (THD) and Linearity Performance

The primary contribution to the distortion in the circuit comes from the ADMP411 microphones. The other analog amplifier components operate in a linear region, with the signals not close to the supply rails. The ADMP411 has an acoustic overload point of 131 dB, which is the point at which THD is 10% for an individual microphone and is commonly referred to as the clipping point in audio applications. All microphones in the array are simultaneously exposed to similar SPLs, so the total distortion curve of the circuit looks very similar to that of an individual microphone. The distortion curve for the ADMP411 is shown in Figure 11.

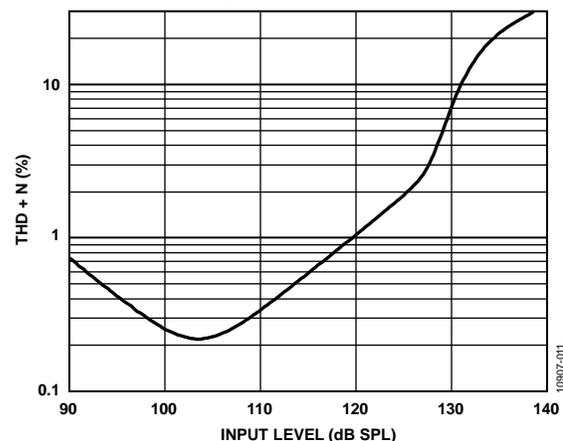


Figure 11. ADMP411 dB SPL vs (THD+N)%

With a -27 dBV sensitivity, the output of the microphone with 131 dB SPL input is $2.27 V_{RMS}$ ($6.40 V$ p-p). This is well within the linear region for the amplifiers operating with a $\pm 9 V$ supply, so the linearity and distortion is controlled by the ADMP411 performance. Figure 12 shows the linearity (input in dB SPL vs. output in dBV) for this circuit design.

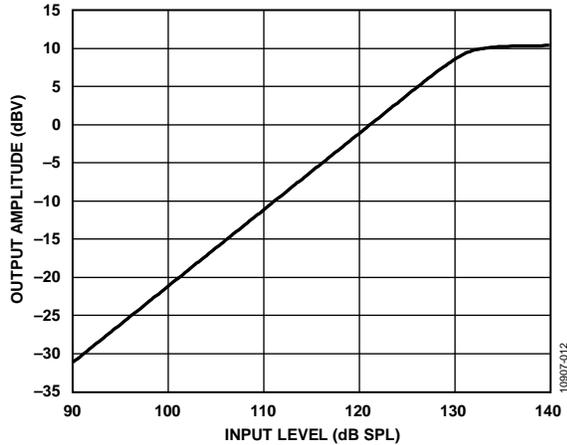


Figure 12. Measured Linearity of Circuit

Directionality Performance

The ADMP411 microphones are omnidirectional when used individually. When put in a large array like this design, the microphones have some directionality. That is, the level of the output depends on the orientation of the array with regard to the position of the sound source.

A single array board exhibits little directionality at low frequencies (<4 kHz). In the mid frequency range, the array boards attenuate the sounds from the rear and sides by as much as 5 dB to 6 dB because of the acoustic shadowing from the PCB itself. At some higher frequencies where the wavelength of sound is on the order of the size of the array PCB, such as around 8 kHz, the array has significant directionality. Directionality measurements using this design are shown in Figure 13.

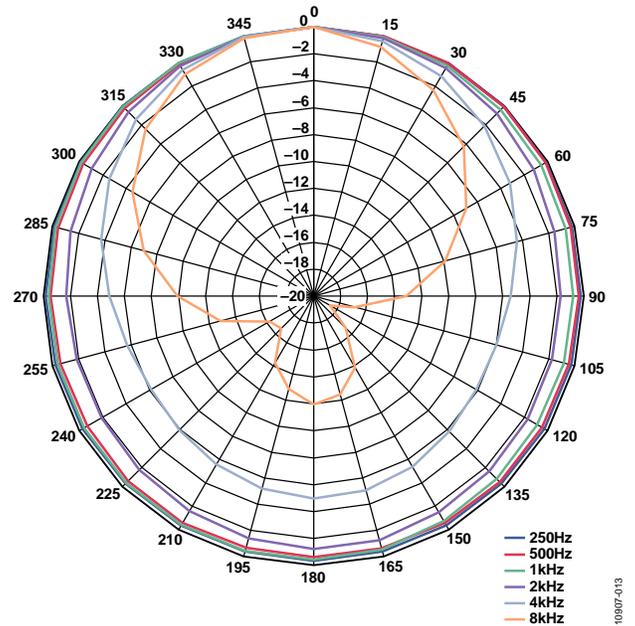


Figure 13. Directionality of a Single Array Board at Different Frequencies

The primary purpose of having two array boards spaced a fixed distance apart from each other is to create a directional response. The combination of the distance between the two boards and the beamforming circuit results in an array that has considerable rejection of sounds from the sides and rear of the assembly. Figure 14 shows the response of two boards whose faces are spaced 18 mm apart from each other. The measurements of this setup show a supercardioid directional response, with significant attenuation of sounds from the rear and sides, but with a small rear lobe. The nulls in the directional response of this design are at about 135° and 225° . At most frequencies in the voice range (250 Hz to 4 kHz), there is at least 15 dB of off axis attenuation.

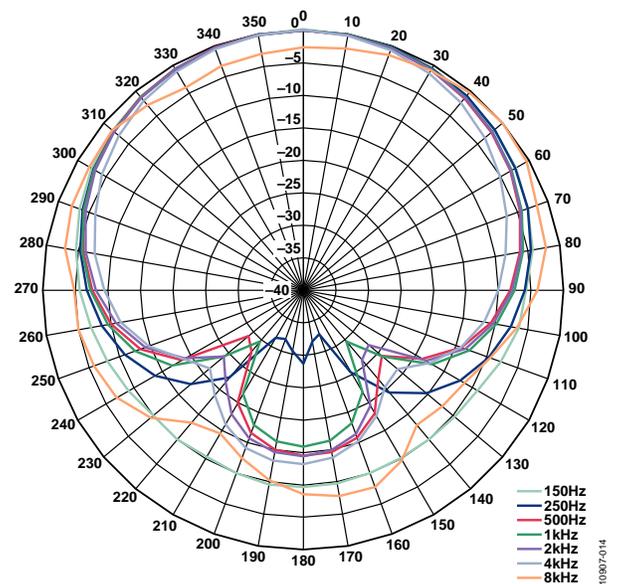


Figure 14. Directionality of a Two-Board Endfire Array at Different Frequencies

COMMON VARIATIONS

Other op amps or difference amps can be used, depending on the need of the specific design. Analog Devices, Inc., has op amps with lower noise, lower power, or lower operating voltage rails than the [ADA4075-2](#), some of which may be a better choice if the design parameters are different than what is presented here. The [AN-1165 Application Note](#) lists many op amps that are suitable alternatives.

The ADMP510 is another analog microphone in an even smaller package and has a lower noise floor and ADMP411. The ADMP510 has an SNR of 65 dBA, while the other microphone has a 62 dBA SNR. A summed array of 16 ADMP510 microphones can have an SNR as high as 77 dBA. However, the acoustic overload point of the ADMP510 devices is 124 dB SPL, compared with 131 dB SPL for the ADMP411. This the lower acoustic overload point makes the ADMP510 a better choice for microphone designs that may not be used in loud environments and require a lower noise floor.

A fixed output version of the [ADP1720](#) can also be used if the microphones operate from a 3.3 V supply. This version eliminates the need for the two external output, voltage setting resistors.

Figure 15 and Figure 16 show the hardware described in this application note.



Figure 15. Two Microphone Boards with 18 mm Spacer

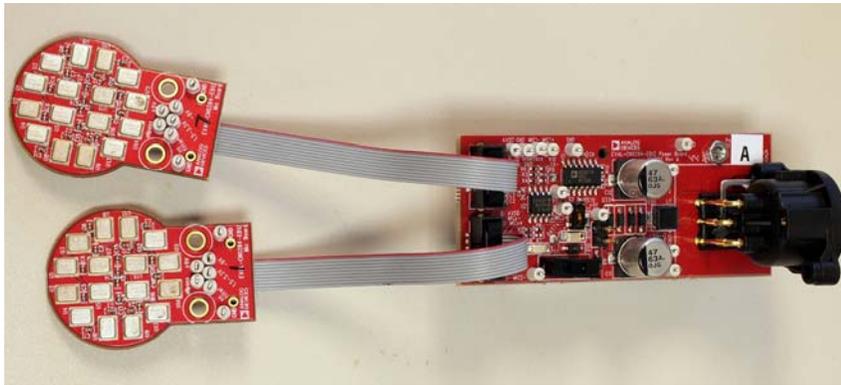


Figure 16. Hardware Configuration

LEARN MORE

The ADMP MEMS microphone products mentioned in this application note are manufactured by InvenSense, 1745 Technology Dr., San Jose, California 95110.

[Analog Devices EngineerZone Audio Community.](#)

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Data Sheets and Evaluation Boards

[ADA4075-2 Data Sheet](#)

[AD8273 Data Sheet](#)

[ADP1613 Data Sheet](#)

[ADP1720 Data Sheet](#)

REVISION HISTORY**11/14—Rev. 0 to Rev. A**

| | |
|---|-----------|
| Changed Title of Document from CN-0284 to AN-1328 | Universal |
| Updated Format | Universal |
| Deleted Evaluation and Design Support Section..... | 1 |
| Added Table 1; Renumbered Sequentially..... | 2 |
| Changes to Additional Circuitry Section..... | 5 |
| Deleted Circuit Evaluation and Test Section..... | 9 |
| Changes to Learn More Section and Data Sheets and Evaluation Boards Section | 9 |

7/13—Revision 0: Initial Version