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Application Note 07

AN07

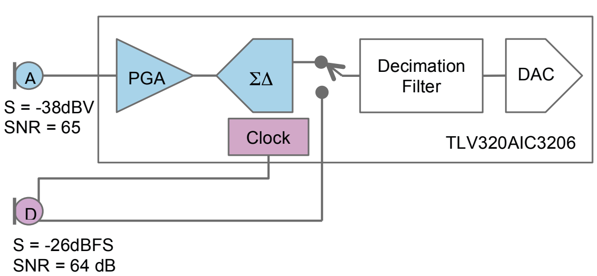
Analog vs Digital Microphones — System Considerations

It is very tempting to only consider analog microphones in audio designs because they are less expensive, use less current, and have lower noise than digital microphones. Although this is true, when the external circuitry required by analog microphones is taken into account, digital microphone designs often have better performance in each of these areas, as well as others.

Setting Up The Proper Signal Path Gain Structure

The majority of audio systems are digital in nature. Even when an analog microphone is used it is usually connected to a codec to convert the signal to the digital domain. A digital microphone's output is already in the digital domain and doesn't require as much circuitry.

The TLV320AIC3206 codec from Texas Instruments is helpful in comparing the performance of analog and digital microphones because it has inputs for both analog and digital microphones as shown below:

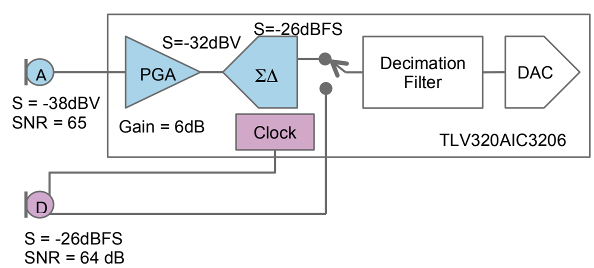


Codec Signal Path

For this example the SiSonic™ microphones SPH1611LR5H-1 and SPH0641LM4H-1 will be used for the analog and digital microphones, respectively.

For a fair comparison the same input level, 94 dB SPL for example, should yield the same output level, -26 dBFS. The TLV320AIC3206 has a Sigma-Delta (ΣΔ) converter gain such that a -6 dBV input yields a 0 dBFS (i.e., a fullscale) output. For a 94 dB SPL input signal, the analog microphone produces a -38 dBV output signal which would correspond to a 32 dBFS output signal. This is 6 dB smaller than the signal obtained from the digital microphone. For the microphone levels to match, the PGA must be programmed to a gain of 6 dB.

With this gain lineup the signal levels look like this:



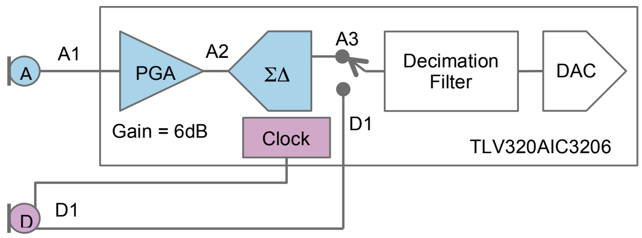
Codec Signal Levels

This gain lineup allows both microphones to produce the same output level (e.g., -26 dBFS) when exposed to the same input level (e.g., 94 dBSPL). The codec is now setup to allow for fair comparisons between a system built around an analog microphone and one built around a digital microphone.

Theoretical Noise Performance Comparison

With the gains setup as shown above, the noise floor at the various points in the signal chain can be calculated. First, the noise floor of the raw microphones is easy to calculate per the table below:

Signal Path Point	S	SNR	Noise Floor
A1	-38 dBV	65 dB(A)	-103 dBV(A)
D1	-26 dBFS	64 dB(A)	-90 dBFS(A)



Noise Floor at Various Points in the Signal Chain

The complete analog microphone signal path is much more problematic since the PGA and ΣΔ converter also add noise. In addition, the noise contributions of the PGA and ΣΔ converter are not specifically mentioned in the TLV320AIC3206 datasheet. However, proper design technique should result in the frontend being the dominant noise source. Assuming this applies to the codec then the SNR listed in the datasheet is dominated by the PGA (-93 dB for a -6 dBV fullscale signal = 99 dBV noise floor).

Mic Noise @ A1	PGA Gain	Mic Noise @ A2	PGA Noise @ A2
-103 dBV(A)	6 dB	-97 dBV(A)	-99 dB(A)

Total Noise @ A2	ΣΔ Gain	Total Noise @ A3
-94.9 dBV(A)	6 dB	-88.9 dBFS(A)

The SNR at the point where both signals are digital can now be calculated as shown in the table below:

Signal Path Point	S	Noise Floor	SNR
A3	-26 dBFS	-88.9 dBFS(A)	62.9 dB(A)
D1	-26 dBFS	-90 dBFS(A)	64 dB(A)

That is, while the raw analog microphone performance was 1 dB better than the digital microphone, once the rest of the analog circuitry was added the analog microphone mow performs over 1 dB worse.

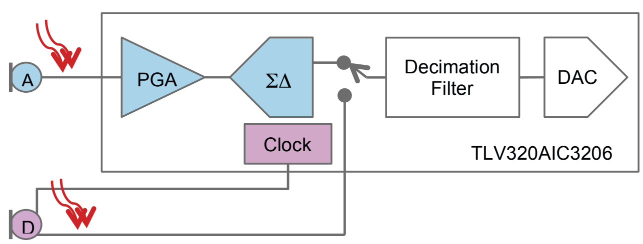
Noise Floor Verification By Measurements

In order to test if digital microphones really perform better in a system, the circuit was built and measured. Great care was taken to insure a clean design with as few signal perturbations as possible. The results are shown below:

Microphone	Datasheet SNR	Theoretical SNR	Measured SNR
Analog	65 dB(A)	62.9 dB(A)	61.8 dB(A)
Digital	64 dB(A)	64 dB(A)	63.2 dB(A)

The digital system degraded an extra 0.8 dB while the analog system degraded 1.1 dB. The analog system is now almost 1.5 dB worse than the digital system!

This indicates there must be something else limiting analog performance. In fact, all analog systems will degrade due to crosstalk, interference, coupling, etc. caused by signal routing. This degradation doesn't show up in digital systems since it takes a lot more interference to change a "0" into a "1" (which is on the order of 1V) compared to corrupting a small 10mV analog signal. In addition, even if a bit was lost in the PDM stream it would have very little signal impact since the conversion to PCM involves a significant amount of averaging, reducing the effect of single bit errors.



Interference Couples Onto Signal Traces

In fact, we can estimate the amount of degradation in this design since Total Noise = Mic Noise + PGA Noise + Interference Noise. In this case the interference adds -106.4 dBV(A) of noise, which is 4.8μV A-weighted. As can be seen, it doesn't take much to corrupt the noise floor of an analog microphone.

Power Supply Rejection Ratio

In many designs the power supply contains all sorts of interference and signals, especially in RF systems such as cellphones. These signals can find their way into the output signal through the microphone itself. This leakage path can be measured and is called Power Supply Rejection Ratio (PSRR). It is a positive number that indicates how much the signal is attenuated as it passes from the power supply terminal to the output signal. Examples of digital and analog microphone PSRR is shown below:

Microphone	S	PSRR	Output for 100mVpp Interference
Analog	-38 dBV	60 dB	-89 dBV
Digital	-26 dBFS	55 dBV/FS	-84 dBFS

Which one has less interference? If just the raw numbers are compared it may look like the analog mic performs better. It has more rejection (60 vs 55) and the output level of the interference is lower (-89 vs -84). However, the units are different. Analog microphones give an output in dBV while digital microphones give an output in dBFS. The only way to compare analog and digital microphone performance is to find a measurement that is common to both, dB SPL. All of these numbers must be Referred To their Input (RTI).

This is accomplished by using their respective sensitivities. The result is the equivalent input sound pressure level that will produce the same output level as the interference. An equivalent way to think of this is to not look at the raw output level of the interference but look at how far below sensitivity it is. The equation is Output (RTI) = Output level –S +94 and the table below shows this result:

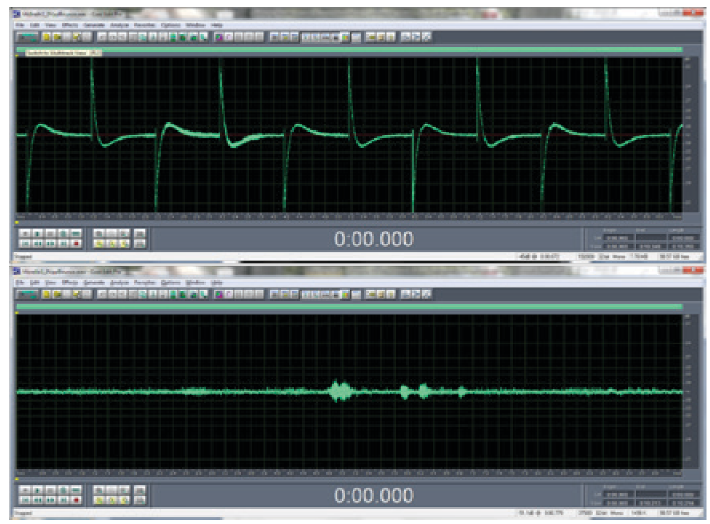
Microphone	S	Output Level	Output (RTI)
Analog	-38 dBV	-89 dBV	43 dB SPL
Digital	-26 dBFS	-84 dBFS	36 dB SPL

Now it can clearly be seen that this digital microphone performs 7 dB better than this analog microphone. When exposed to the same system noise on the power supply, the digital microphone will sound 7 dB quieter than the analog microphone. Even though PSRR performance is always improving, at any given time digital microphones will generally outperform analog microphones in PSRR.

System Noise on the Ground Plane

This is similar to PSRR except that noise often appears on the microphones ground terminal. It is often referred to as Ground Bounce and can be caused by the starting and stopping of high-current circuitry such as displays and power amplifiers. Ideally these should use their own grounding schemes, far away from sensitive circuits like microphones, but this is often not possible in constrained spaces, such as in a cellphone.

When a 40mVpp squarewave is applied to the ground terminal, the output of an analog microphone and digital microphone are shown below:



Output Due To 40mVpp Ground Bounce
(Analog Top, Digital Bottom)

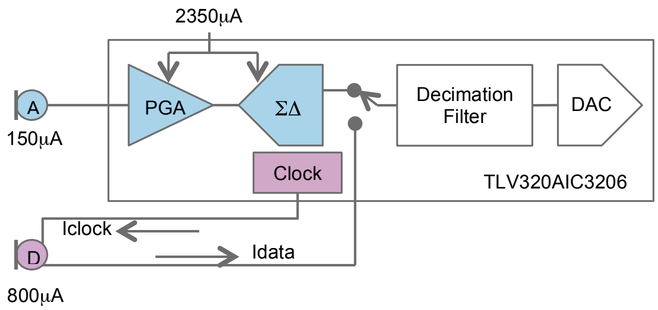
As can be seen the digital microphone is FAR more immune to this effect than the analog microphone. This is due to the fact that in digital microphones both the analog stages and the ADC bounce together, so the bounce does not get encoded. In analog microphones the analog circuitry is in the microphone while the ADC is in the codec. The interface between them is vulnerable to ground bounce (as well as other degradations as described in the noise section above).

In addition, the digital PDM output stream is inherently immune to noise. It takes many bit errors before a pop can be heard.

System Current Consumption

Digital microphones are known for requiring more current than analog microphones but this is not a fair comparison since analog mics require a PGA and ADC. In the codec shown below, the Clock is required for both analog and digital microphones. But the PGA and ADC, which draw 2.35mA (TLV320AIC3206 Application Reference Guide 2.5.2.5 shows 10.1mW which also includes the decimation filter, 2.35mA, 4.23mW, assumes most of the current is used by the decimation filter) are only required for the analog microphone. These will be turned off if a digital microphone is used.

Total current in an analog microphone system = 150+2350 = 2500µA.



Current for a digital microphone system is more complicated to predict because it also requires current to charge the bus capacitance of both the Clock and Data lines. These currents must be accounted for because they could make a significant contribution to the total system current requirement.

Current Consumption Of Analog And Digital Microphone Systems

Digital microphone datasheets contain the following equation which describes the additional current required to charge a load capacitance:

$I_{data} = 0.5 * V_{dd} * F_{clk} * C_{load}$

Likewise, for the clock:

$I_{clock} = V_{dd} * F_{clk} * C_{load}$

The factor of 0.5 is required for Idata since digital microphones are only ON for half the clock cycle.

For example, if V_{dd} = 1.8v. F_{clk} = 2.4MHz, and C_{load} = 15pF (not unusual for a PCB trace + input capacitance), then Idata = 0.5 * 1.8 * 2.4e6 * 15e-12 = 32.4 µA

Likewise, Iclock = 64.8 µA.

Total current in this digital microphone system = 800+32.4+64.8~ 900 µA.

The two systems are compared below:

Microphone	Datasheet Current	System Current
Analog	150µA	2500µA
Digital	800µA	900µA

When the total system is included, the current of the analog system has increased over x10 more than the raw analog microphone current! Digital microphones require the lowest amount of power.

Summary

Even though analog microphones look very appealing when looking at microphone datasheets, in many systems digital microphones may provide these benefits:

- Lower System Noise
- Higher System SNR
- Better Power Supply Rejection
- Better immunity to Ground Bounce
- Less System Current Consumption

In addition, signals from digital microphones are easier to route since they are immune to crosstalk, interference, coupling, etc. The success rate for viable PCB designs could be much higher for digital microphone designs.

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