Achieve good audio quality in portables

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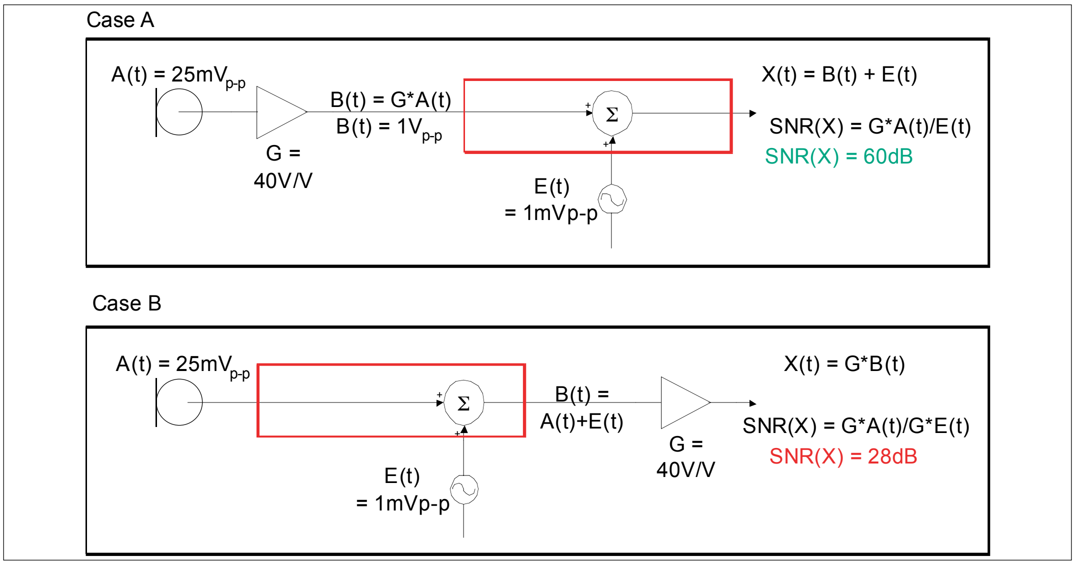

Figure 1: In Case A, the signal is amplified near the microphone before the trace travels across the board and noise is coupled, resulting in a system SNR of 60dB. In Case B, the signal is amplified after the trace travels across the board and noise is coupled, resulting in a system SNR of only 28dB.

Today's converged portable multimedia devices have more and more features integrated into their smaller systems. Audio is a basic feature of any system marketed with multimedia capability. System designers, however, often put more engineering focus on portable multimedia devices' "glamorous" features such as wireless connectivity, video processing, image capture and display. Hence, audio circuits end up wedged in the system wherever space can be found among the "important" components, resulting in mediocre or downright poor audio quality. However, with a little care and attention, good audio quality can be seamlessly integrated with the myriad of other features demanded by consumers. This article provides suggestions for a good system design and PCB layout practices relevant to the design of any portable system that includes audio playback and/or recording functionality.

There are many sources of poor audio quality in portable audio systems, but this article focuses on sources of audible noise on analog audio signals. Non-harmonically related noise—whether white (flat) or tonal—can be very annoying to the end-user. White noise is perceived as a background hiss, which is very noticeable during quiet audio passages. Tonal noise can be perceived as a buzz, hum or whine, depending on the frequency content. Unnecessary noise corruption of audio signals can be avoided with good system design PCB layout practices.

Most portable audio systems use a DAC or codec IC to convert digital audio to analog signals. The layout surrounding the audio codec or DAC is critical.

Codec or DAC devices have both analog and digital circuits in the same IC. Thus, multiple supply pins are used for analog and digital power, often marked AVDD and DVDD. Other analog supply pin names can be HPVDD, DRVDD, SPKVDD and PVDD. Other digital supplies can be IOVDD and BVDD. These power-supply pins are separate because the digital circuits can be very noisy due to high-speed switching currents, and the analog circuits are very sensitive to noise on the supplies. An important point of audio system design and layout is that analog supply pins should be provided with a very "clean" supply with minimum ripple and transients. Any noise on the analog supply pins can corrupt the audio I/O signals in various ways.

In a portable audio system, the primary power source is usually a battery. The battery can be very noisy due to transients caused by other parts of the system, including wireless transceivers, storage devices and displays. Instead of using the battery voltage directly, a good practice is to use a low-dropout (LDO) regulator with good power-supply rejection ratio and low output noise. This would provide the analog supply voltage to the audio codec or DAC, along with any other audio signal-path devices such as amplifiers, thus ensuring that a "clean" supply is present for the analog circuitry. Care should be taken to select an LDO regulator that has an adequate current rating for the circuitry being powered as well. Proper decoupling capacitor usage on analog supplies is also important. Large decoupling capacitors (10µF or greater) are useful for filtering the supply voltage. Smaller-value decoupling capacitors (1µF or smaller) are also needed to supply fast transient currents when the IC calls for it. Decoupling capacitors should be placed as close to the analog supply pins as possible, avoiding PCB vias between the capacitor, and the supply and ground connections if possible. Smaller decoupling capacitors should be placed closer to the IC pins than larger capacitors, as series resistance affects response times of smaller capacitors more noticeably.

Digital power supplies in audio converter ICs are much less sensitive to noise than analog supplies, so the digital circuits can be supplied with a more efficient switched-mode power supply (SMPS). SMPS usually has a higher output riddle and noise, but its 80 percent efficiency and higher yield significantly improve battery life. Often, large decoupling capacitors are not necessary in digital power supplies. However, multiple smaller capacitors (i.e. 1µF and 1nF) should be used to supply very high-frequency switching currents in the digital circuitry. Again, smaller-valued decoupling capacitors should be placed closer to the IC pins than larger capacitors.

Good design practices

Another source of noise corruption in portable audio systems is noise coupling into analog I/O signals. Noise-coupling mechanisms can be inductive or capacitive, but good system design and PCB layout can minimize noise coupling.

System area	Recommended design practices
IC analog power supplies	Use LDO with low noise, low ripple and high PSRR.
	Use proper decoupling capacitors placed close to IC.
IC digital power supplies	Use SMPS with high efficiency.
	Use proper decoupling capacitors placed close to IC.
Analog audio signals	Use differential signals and connections.
	If gain is necessary, amplify signal near source.
	Minimize PCB trace lengths.
Microphone bias voltage	Filter bias voltage near microphone.
Speaker/headphone outputs	Use short, wide PCB traces to minimize resistance.

Table 1: Good quality audio can be achieved in a low-cost, low-power portable audio system.

One practice that yields very good noise immunity is using differential signals wherever possible in the analog audio signal path. PCB traces used for differential signals should be routed as parallel pairs with matched impedance, so that any noise will couple equally (as a "common-mode" signal) into both sides of the differential signal path. The common-mode rejection property of the differential circuits used rejects any coupled noise, which in turn, reduces the noise heard by the user. Although there are cases where differential signals cannot be used, they are still a very useful tool.

Another good system-design practice is to use the highest possible voltage levels for signals that travel across the PCB and are thus susceptible to noise coupling. It is valid to assume that the magnitude of coupled noise does not increase with the level of the signal being transmitted. So if noise level stays constant and signal level increases, the SNR will increase. A higher SNR measurement indicates a higher-performance audio system. If a low-level signal is run across a PCB, gain must be applied. This can increase the noise along with the signal, reducing overall system SNR. Amplifying a low-level signal near its source is a good practice.

Figure 1 shows an example of this principle. A microphone generates a 25mV$_{p-p}$ signal A(t), which must go across a PCB and be amplified to 1V$_{p-p}$ for further processing. The red box indicates the trace traveling across the board, which receives coupled noise, indicated by the signal E(t). In Case A, the signal is amplified near the microphone before the trace travels across the board and noise is coupled. This results in a system SNR of 60dB. In Case B, the signal is amplified after the trace travels across the board and noise is coupled. This results in a system SNR of only 28dB and illustrates the difference in performance realized by good system design.

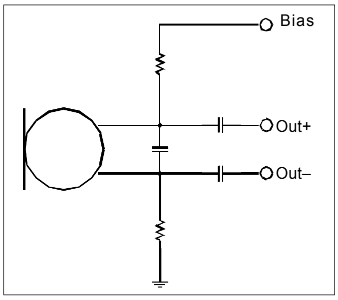

Figure 2: Filtering the bias voltage with resistors and capacitors near the microphone is a good practice.

For signals that cannot be amplified near the source due to system cost or size constraints, reducing the PCB trace length to the minimum possible is critical. Short PCB traces are less susceptible to noise coupling by capacitive and inductive mechanisms.

A final type of signal that should be carefully designed in systems with built-in microphones is the microphone bias circuitry. Most of the electret capsule microphones (ECM) used in portable audio systems require a bias voltage in the 2-3V range. Often, the bias voltage is provided by an IC located away from the microphone. In this case, the bias voltage picks up noise on its way to the microphone capsule. This noise couples directly into the microphone output. In this face, filtering the bias voltage with resistors and capacitors near the microphone is a good practice. **Figure 2** shows a microphone circuit design with a pseudo-differential connection and RC filter to attenuate noise from the bias voltage.

All audio systems need some type of transducer so that users can hear the audio produced. In most systems, a headphone output is present. Some systems include a built-in speaker or outputs to drive external speakers. Because headphones (>16Ω) and speakers (> 4Ω) can require high-power signals, minimizing the impedance of the circuit traces associated with these transducers is critical. If PCB traces have unnecessary high resistance, power can be lost in the PCB traces and not delivered to the transducer. This results in loss of audio quality, reduced battery life and unnecessary heat build-up in the system. Making speaker and headphone traces as wide and as short as possible will reduce the impedance and minimize the negative effects mentioned.

Table 1 summarizes all the recommendations discussed. When these practices are followed, good quality audio can be achieved in a low-cost, low-power portable audio system.