Challenges for designers of digital-camera audio subsystems

MORE THAN A FEW DESIGN TRICKS GO INTO PROVIDING THE IMPRESSIVE AUDIO PERFORMANCE OF LOW-COST DIGITAL-VIDEO CAMERAS.

From a system designer’s point of view, digital-video cameras combine the worst of the portable consumer and home-entertainment worlds. There are rigid constraints on space, power consumption, and component cost. But end users still expect their recordings to look and sound well when played back on stationary equipment in which recording flaws are much more visible—and audible. High-quality microphone recording is therefore indispensable. Acoustic- and mechanical-noise issues associated with handheld use in outdoor environments, as well as electrical interference from switching power supplies, motors, and digital circuitry within a camera, make it all the more difficult to achieve good sound quality.

Audio playback presents additional challenges. The tiny loudspeakers that portable equipment uses limit both audio volume and audio quality. Fortunately, countermeasures ranging from analog-circuit design to DSP (digital-signal processing) and PCB layout are available to address these issues.

THE SIGNAL CHAIN

Consumer cameras have one or more internal condenser microphones as well as an input jack for external condenser or dynamic microphones. A preamplifier, an ADC, and DSP functional blocks follow these elements (Figure 1). Each stage adds noise and distortion. Without external noise sources, the weakest link in this chain limits the overall SNR and THD (total harmonic distortion). The weakest link is usually the preamp or the ADC, in which audio performance is most expensive, not only in component cost, but also in power consumption.

An electret condenser-microphone capsule incorporates a built-in FET buffer that makes the signal less susceptible to interference on its way to the preamplifier. However, the FET also generates thermal noise. Moreover, any noise in the microphone-bias voltage that powers the buffer also adds to the audio signal, severely degrading the SNR because of the very low signal amplitude at this point. Providing a clean supply is therefore even more important for the microphone capsule than for the subsequent stages.

Handheld and built-in microphones also require ALC (automatic level control) to counteract seemingly random variations in signal volume as the microphone moves toward or away from the sound source. The ALC keeps the recording volume approximately constant by introducing a variable gain or attenuation stage into the signal path. This variable gain affects not only the signal, but also the noise added before the ALC stage.

In all-digital implementations operating on the digitized signal, the result is that the ADC’s constant quantization noise is amplified along with the signal during quiet passages, progressively worsening SNR as the signal volume decreases. Amplifying the signal in the analog domain, before the ADC adds its quantization noise, results in better SNR.

However, even analog ALC circuits amplify thermal noise

Figure 1 The cleverness that produces the high-quality audio is not immediately apparent from examining a typical portable camera’s audio-signal chain.
generated in the FET buffer. To avoid filling up periods of silence with loud, white noise (a phenomenon known as noise pumping), designers should use ALC in conjunction with noise gating, which cuts off the signal when its amplitude drops close to the noise floor. If possible, designers should also tailor ALC timing to the type of signal; speech usually sounds best with shorter hold and decay times than those that are appropriate for music, whose volume changes are often intentional.

**ACOUSTIC NOISE**

By a common definition, “acoustic noise” refers to any unwanted sound that enters the microphone, such as voices, music, or traffic noise. These noise sources usually originate off-axis; the sound does not come from where the camera lens is pointing. Highly directional, shotgun-type microphones pick up far less off-axis noise but are equally insensitive to all off-axis sound. For capturing background sounds, a cardioid or omnidirectional microphone is more appropriate. Video professionals have a set of microphones to choose from. For more modest amateur requirements, a zoom microphone is a good compromise. Such a microphone is actually an arrangement of two microphones with different directionality, or a pair of back-to-back-mounted membranes, whose signals are mixed in varying proportions depending on the optical zoom. In wide-angle scenes, the combined response is omnidirectional but gets progressively narrower as the lens zooms in.

Air blowing directly into the microphone generates wind noise. It is mostly random in nature and occupies the low audio frequencies. Foam windscreens effectively suppress wind noise, but designers should supplement them with electronic highpass filtering in windy conditions. Second-order filters with cutoff frequencies near 200 Hz have worked well in practice. Because effective shock mounting is impractical in consumer cameras, the only option is filtering. As with plosive noise, a smaller microphone-coupling capacitance blocks the worst mechanical-noise peaks. If necessary, you can supplement this measure with additional highpass filtering in the digital domain. However, for higher frequency components of handling noise (fingernails tapping against the enclosure, for example), the cutoff frequency would be well within the audible range. Therefore, it is preferable to rely on the user’s more gentle handling of the equipment, rather than indiscriminately suppressing the entire bass range.

**MECHANICAL NOISE**

The microphone membrane reacts to mechanical vibrations and shocks just as much as it does to sound waves. In handheld cameras, this sensitivity leads to problems in handling noise. First, internal microphones often pick up the rattling of moving parts, such as pushbuttons or loose cables within the camera enclosure. You should prevent any such parts from moving when the camera shakes. Second, the mechanical action of fingers and palms on the camera enclosure generates large peaks in the bass and infrasonic regions.

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![Figure 2 Digital filtering can suppress noise that is limited to one part of the audio spectrum, such as low-frequency wind noise and narrowband zoom noise.](image-url)

**ELECTRICAL NOISE**

Transients in dc/dc converters and digital signals generate EMI (electromagnetic interference) that nearby pc-board tracks pick up. The connection from the microphone to the preamplifier is particularly sensitive to EMI because the signal amplitude is relatively low in relation to the induced spikes. EMI covers a wide bandwidth, and, thus, filtering cannot remove it. However, a differential microphone preamplifier can suppress EMI spikes that appear in both the microphone signal and the microphone ground. To use this approach, you must treat the microphone ground as a signal in its own right and route it next to the microphone-signal track. The differential amplifier then subtracts the ground voltage from the signal, canceling out the EMI and leaving the desired signal.

This remedy is imperfect, though. In a real-world pc-board layout, one of the two tracks is slightly longer or closer to the noise source, making the EMI spikes larger in one track than in the other. Moreover, differential amplifiers’ finite CMRR (common-mode rejection ratio) results in attenuation—rather than complete cancellation—of even perfectly identical signals. You should therefore design the board to reduce EMI pickup. First, make the microphone connection as short as possible. If you use a cable, shield it. You can produce a shielding effect on
the board itself by placing grounded lands on both sides of the signal track and creating ground planes on adjacent copper layers. The next step is to maximize the physical distance between the EMI source and the analog audio circuitry. Finally, wherever possible, minimize the EMI that digital and power-switching circuits emit within the camera.

The power supply provides another point for noise entry into analog circuits. As digital and power-management circuits pollute their supply rails with switching noise, the microphone preamplifier and the analog part of the ADC need a separate, low-noise supply. Carefully separate digital and analog grounds. Finally, good decoupling of all of the supply voltages short-circuits supply noise before it can spread around the system.

Zoom noise is an issue peculiar to cameras. Originating from the stepper motor attached to the zoom lens, it can spread acoustically, mechanically, or electrically—through the supply, or as EMI—through the alternating magnetic field that the motor windings emit. Zoom noise is periodic and falls into a narrow band around the motor’s step frequency, usually within the audio range; harmonics and mechanical gears can add peaks at other frequencies. Digital notch filtering can remove this type of noise but also suppresses genuine audio signals within the same frequency band. To minimize such collateral damage, the notch filter should be highly selective with a narrow stopband, and you should disable it whenever the lens motor is inactive.

You can use the same approach to guard against narrowband noise from motors in hard-disk drives or other electromechanical components.

**AUDIO PLAYBACK**

Although cameras are primarily recording devices, audio-playback quality is also an important part of a user’s experience. The options for playback are headphones, built-in loudspeakers, and line-out connections to a home high-fidelity system. The line-out connection is the most demanding in signal purity and requires a low-noise, low-distortion DAC and an output buffer. In practice, headphone outputs can often double as line outputs, because their THD is much lower with a high-impedance line load than with 16 or 32Ω headphones. Higher signal amplitudes make the playback signal chain less noise-sensitive than the recording side. Nevertheless, the usual precautions for mixed-signal circuits apply: Separate analog from digital, provide clean analog supplies, and protect analog signals from EMI.

If the signal is good enough for a line-out connection, it’s good enough for headphones, too; the transducer rather than the electronics usually limits overall performance. However, producing sufficient volume with built-in loudspeakers can be difficult. Small magnets and membrane diameters limit their energy efficiency, and cranking up the signal level only causes distortion and even more quickly depletes the battery.
DSP techniques can boost perceived loudness without increasing the peak signal level that the loudspeaker and the analog-signal path see. One option is dynamic compression, which applies extra gain when the signal amplitude is low. This technique squeezes the signal’s dynamic range into a narrower region near full-scale.

Another technique is peak limiting, which amplifies the signal beyond full-scale and instantaneously ramps down the gain during signal peaks to prevent clipping (Figure 3). In effect, the method compresses the top part of the original signal’s dynamic range and amplifies the lower part. Both methods reduce the signal’s dynamic range, and you should employ them when you use headphones or a line-out connection.

An additional problem with small loudspeakers is their poor bass response. Bass-boost circuits can help mitigate this issue; alternatively, equalizers offer a more complete approach that can also iron out other kinks in the speaker’s frequency response. Although obtaining flat response to 20 Hz remains elusive with subminiature speakers, bass boost and equalization deliver drastic improvements. You should use both in conjunction with peak limiting to prevent signal clipping when you apply a large gain to one or more frequency bands. You can relatively inexpensively implement dynamic compression, peak limiting, bass boost, and equalization inside an audio chip. As with other signal-processing functions, a dedicated-hardware or DSP approach invariably uses less power than does a software algorithm running on a general-purpose processor.

Figure 3 Using dynamic compression (a) or peak limiting (b) coaxes more volume from small speakers.

**AUTHOR’S BIOGRAPHY**
Yan Goh is a product-marketing engineer at Wolfson Microelectronics (www.wolfson.co.uk) with responsibility for portable products. Before joining Wolfson in 2004, he worked at NewLogic Technologies for four years in sales and marketing in the United States and Austria. He has also held IC-design positions at Infineon and Tritech Microelectronics. He holds a bachelor’s degree with honors in electrical and electronic engineering from the University of Edinburgh (Scotland) and a diploma in electronics and communication engineering from Singapore Polytechnic.