Getting the Most Out of Your Battery

Handsets can now be enabled with hi-fi subsystems more easily than ever using a new generation of flexible and highly integrated audio hub devices, designed for easy adoption, minimal power consumption and small PCB footprint.

Summary

Enrichment of handset features and reduction in size continue to drive trends towards silicon integration in portable multimedia applications. Higher quality audio and louder speakers are now also being demanded by consumers, and reduced battery life is not tolerated. A new generation of highly integrated "audio hub" devices, optimised for portable Lithium battery-powered applications, makes it possible to meet all these needs while also reducing costs and simplifying system development.

Handset Design Challenges

Adding more features to a handset usually has the effect of increasing the drain on the battery, whether as a result of power-hungry video and imaging functions such as large colour displays or multi-megapixel cameras, or using additional signal processing to perform more advanced tasks such as speech recognition or MPEG decoding.

In the audio domain, supporting movie playback, mobile TV, gaming and other multimedia features also has the effect of drawing more current from the battery. While older handsets only required a mono loudspeaker for playing a ringtone for maybe only a few seconds every few hours, recent multimedia phone designs incorporate stereo speakers which are active for much longer periods, e.g. during TV streaming or gaming. These application features also require a higher level of audio quality at the speaker outputs than before. However, stereo speakers require twice the power of a mono speaker, and a movie clip which lasts 10 minutes will drain 120 times more energy from the battery than a mono ringtone which lasts only 10 seconds. Higher levels of volume are also expected nowadays, with 1W speaker output power a fairly typical requirement, and this places further demands on the battery.

Adding features to a handset usually goes hand-in-hand with the addition of circuitry, and in the ever-shrinking handset that means less space than ever before for the battery. Adding yet more power-hungry audio features to a handset while providing it with the smallest possible battery forces designers to look closely at every cause of power wastage and inefficiency in the handset, so that precious battery power can be saved wherever possible. This need for longer battery life is driving the trend towards Class D amplifier technology, which can remove the largest source of inefficiency in the audio circuitry.

Reducing form factors are also driving integration of mixed-signal audio functions, but integration also presents new challenges of improving levels of audio quality without further increasing power consumption or requiring additional external components such as regulators or passive components. This complex design puzzle is increasingly being solved by a new generation of "audio hub" devices. Three major considerations are driving innovation of these devices:

1. Improving audio quality.

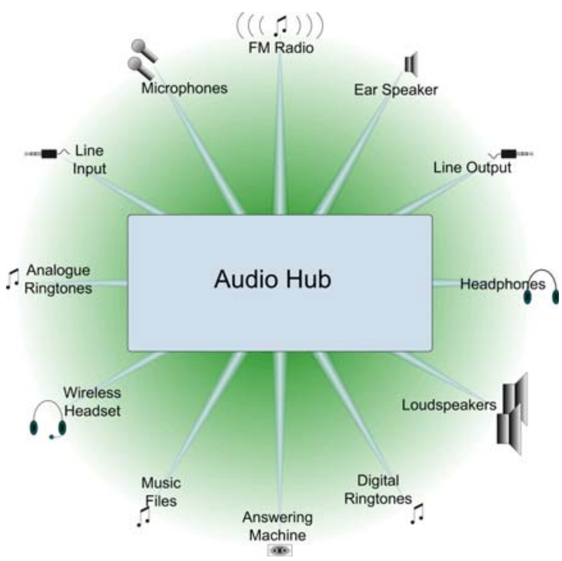
   This is not simply limited to improving SNR and THD performance of key components. It also involves rejecting noise created by other components; eliminating pops, clicks, zipper noise and other transients; and maintaining high performance at higher volume levels - all of which contribute to a better audio experience from the perspective of the end user.

2. Minimising power consumption (in active and standby modes).

3. Reducing PCB footprint and component count.

Audio Hub Concept

Portable multimedia devices such as mobile phones typically contain a number of analogue and digital audio sources in diverse data formats and are required to convert and mix combinations of these audio streams before outputting to the real world via various transducers (ear speakers, loudspeakers, headphones, headsets). In order to save space, cut cost and reduce design complexity it is advantageous to group these audio processing functions into a single device, the "audio hub".



The audio hub must be capable of interfacing analogue signals of varying magnitudes, source impedances, DC offsets and bandwidths, such as FM receivers, microphones, send/receive voice data, ringtones or hi-fi line input. Flexible input configurations can provide support for these diverse signal characteristics in different system architectures while minimising pin count, saving space and reducing costs.

Digital data sources can also exist in different data formats, word lengths and sample rates. While telephony usage modes normally only require the audio hub to handle mono 8kHz data in PCM format, the integration of digital music playback features requires audio hub devices to handle different sample rates, word lengths and data formats (e.g. stereo 16-bit 44.1kHz I^2S data). A flexible digital audio interface and clocking scheme on the audio hub together with hi-fi quality data converters enable digital music playback on the handset with no additional mixed signal components.

Mixing in the analogue domain in the audio hub can eliminate sample rate conversion difficulties, and flexible mixing paths can enable new application features. Devices such as WM8983 and WM8985 allow mixing of any combination of microphone input, digital music, FM receiver, received voice data, and provide the ability to re-digitise this mix, enabling features such as karaoke recording.

Diverging wafer process trends between digital multimedia processors and mixed-signal devices further strengthens the drive to integrate mixed-signal audio functions in a single audio hub.

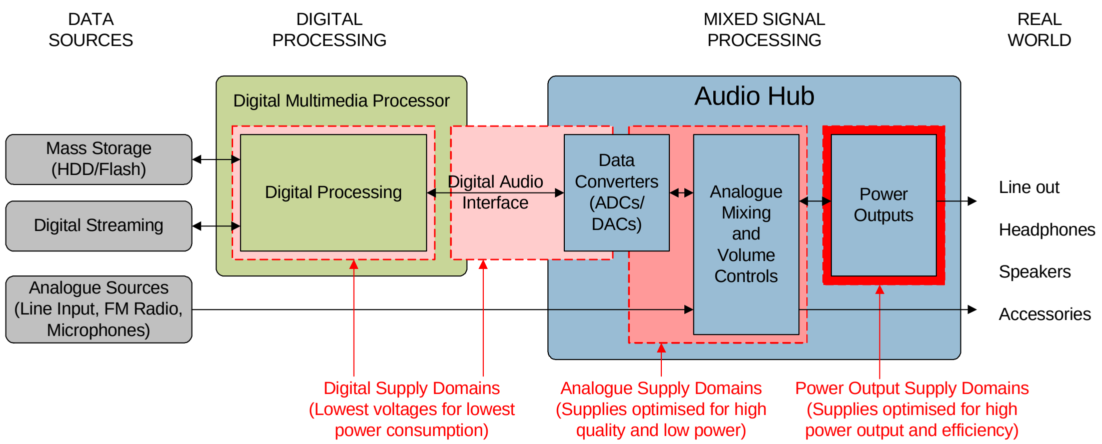


Figure 1: Typical Portable Multimedia System

Power supply requirements for the audio functions in this signal chain are most diverse in the audio hub device, where three or four separate supply domains are typical, each with its own voltage, current capability and noise characteristics. Audio hub devices need to be carefully designed to operate within the various limitations of these supplies. Minimising power consumption without degrading audio signals is the key to providing portable devices with hi-fi quality music without the penalty of unreasonably reduced battery life. Different power-saving techniques must be used for each of the power supply domains.

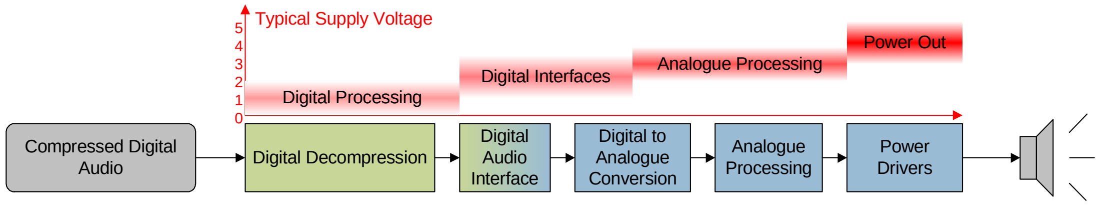


Figure 2: Digital Audio Playback Signal Chain

Digital Supplies - Saving Power

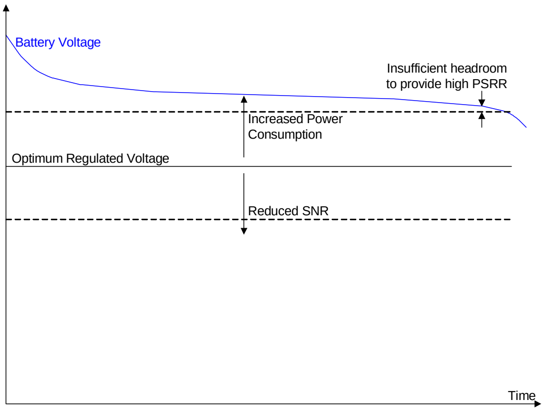
Since audio quality is unaffected by reducing digital supply voltages, the digital core will be powered by the lowest possible voltage in order to save power. At these low voltages the use of DC/DC converters can bring significant efficiency savings compared with linear regulators. The supply ripple caused by high frequency switching of the converter can be tolerated, as digital circuitry is relatively insensitive to ripple.

In a similar way, the digital I/O buffer supply will consume less power at low voltage and audio quality will not be affected, although for practical reasons this supply is sometimes higher than the digital core supply (e.g. for maintaining compatible signalling levels between devices which communicate with each other).

Analogue Supplies - High Quality with Low Power

Unlike digital functions, analogue signal processing elements such as ADCs, DACs, mixers, amplifiers and microphone interfaces are very sensitive to noise. Signal-to-noise ratios can be improved by increasing analogue supply voltages, but at the expense of power consumption. System designers must make this trade-off according to their own audio quality and power consumption targets.

Maintaining a stable, clean analogue supply is also important to prevent supply noise from degrading audio quality. While good design and differential techniques can improve supply rejection, a high-PSRR linear regulator is usually employed to power analogue circuitry on the audio hub. It is important to have sufficient headroom between the regulator's output voltage and the minimum input voltage to maintain high PSRR levels as the battery discharges. Analogue supplies of between 2.7V and 3.0V are fairly typical in portable audio applications.



Although switching regulators can provide greater efficiency, they are not normally used for analogue supplies since they introduce switching noise, require larger and more expensive components, and the efficiency benefits are marginal due to the relatively high regulated output voltage.

The most effective strategy for saving power in analogue circuitry on audio hub devices is to provide flexible and granular power management control, so that circuits not essential for a given usage scenario can be disabled. For example, most hub devices have at least two ADCs and two DACs on board, but a voice record function requires only one ADC to be enabled, while a PCM voice call requires one ADC plus one DAC, and MP3 playback requires two DACs for stereo operation. Where there is a power consumption vs audio quality trade-off in a particular circuit, low power modes can be provided so that when quality requirements are reduced (e.g. during voice communication), performance can be suitably reduced in order to save power. As audio hub device complexity grows to match increasing handset capabilities, the number of possible device configurations also increases and low-level control of the various blocks becomes essential to avoid wasting power.

Speaker Supplies - High Voltage for High Power Output

A separate power supply is normally used for the speaker driver (and often for the headphone driver). Speaker supplies will tend to be higher than other analogue supplies, in order to maximise output power. Using separate speaker supplies also isolates other analogue circuitry from the effects of drooping supply voltage during periods of high load current in the speaker.

Although it is desirable to use a higher supply voltage for speaker drivers, the extra components and increased cost of gererating an additional supply rail, plus the additional drain on the battery caused by boosting the battery voltage make this an unattractive solution. For these reasons many designers prefer to connect the speaker supply directly to the battery. This in turn creates its own challenges.

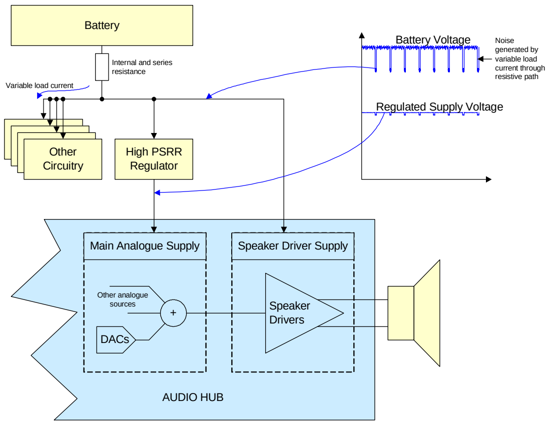


Figure 3: Typical Power Supply Arrangement in Portable Multimedia Applications

Speaker Power Direct from the Battery

High PSRR - High noise levels are frequently present on the battery in portable systems. This noise is usually the result of voltage drops caused by highly variable load currents in other parts of the system (e.g. operation of hard disk drives or pulsing of RF power amplifier during cellular communication). When connecting speaker supplies directly to the battery, a high power supply rejection ration (PSRR) is required in the output amplifier to prevent this noise from feeding through to the audio outputs.

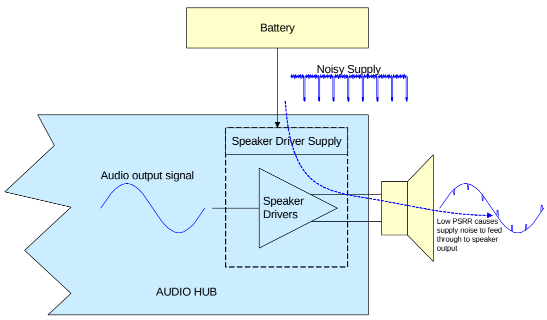


Figure 4: Effects of Low PSRR on Speaker Output

Low Leakage - When the speaker driver supplies of an audio hub device are permanently connected to the battery, any leakage current which flows into the device will be a permanent drain on the battery which will reduce standby time and off time, which is unacceptable to end users. Low leakage into this supply is essential, regardless of whether the other supplies are enabled or not.

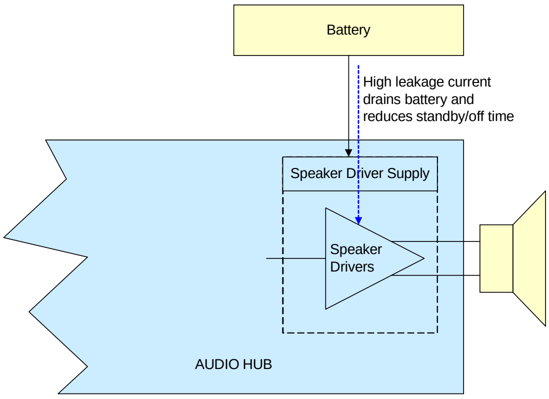


Figure 5: Leakage Current in Speaker Drivers

Signal Boost - To achieve the highest possible volume levels from the loudspeaker(s) in portable applications, the full dynamic range of the speaker supply must be utilised. Since the signal at the speaker driver input is derived from circuits powered by a different (lower) supply voltage, the signal magnitude at the speaker will not make use of the full available range without some additional gain being applied. To avoid waveform clipping, common mode level shifting is also required. Many solutions use external speaker drivers which require additional passive components to set these AC and DC gain values, while other devices such as the WM8960 integrate both speaker drivers and the additional gain stages, saving space and cost.

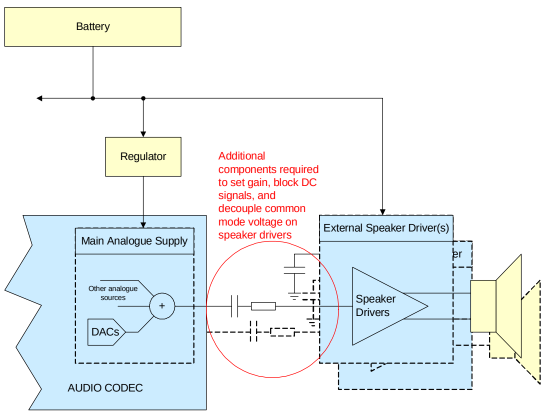


Figure 6: External Speaker Drivers: Increased Component Count

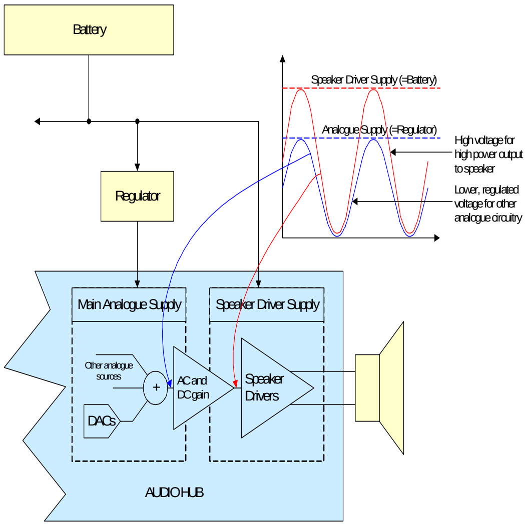


Figure 7: Internal Signal Boosting to Maximise Speaker Volume

Class D Speaker Drivers - The technology used in the speaker driver has by far the largest impact on overall efficiency. Class AB speaker drivers typically waste much more power in the device than they transmit to the speaker, reducing battery life and potentially causing device overheating.

For example, a stereo class AB speaker driver delivering 1W per channel at 40% efficiency will drain no less than 5W from the battery. 3W of this is dissipated as heat in the device. The sum total of all other audio-related power can be two orders of magnitude less than this in some applications, making the speaker driver the dominant source of inefficiency and unecessary battery discharge.

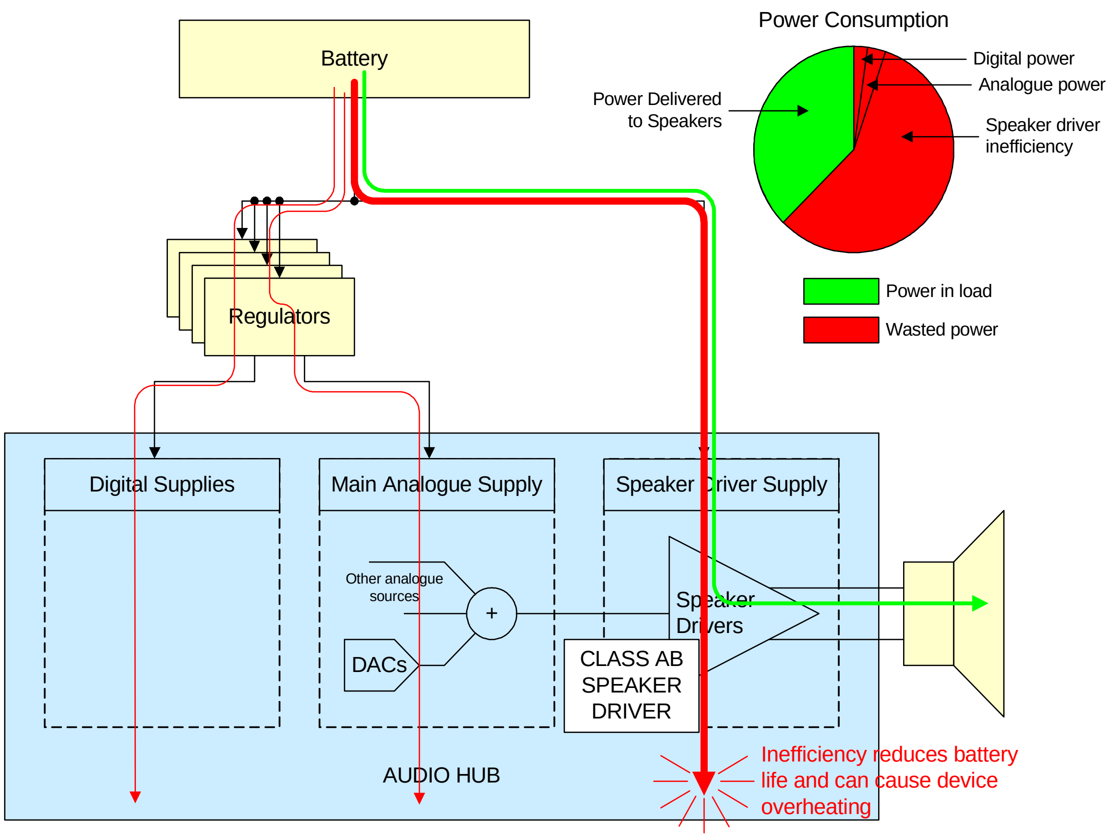


Figure 8: Class AB Speaker Drivers: Reduced Battery Life and Potential Overheating

Class D speaker drivers are increasingly being employed to increase efficiency, extending battery life and simplifying thermal management issues which can constrain device functionality and increase costs.

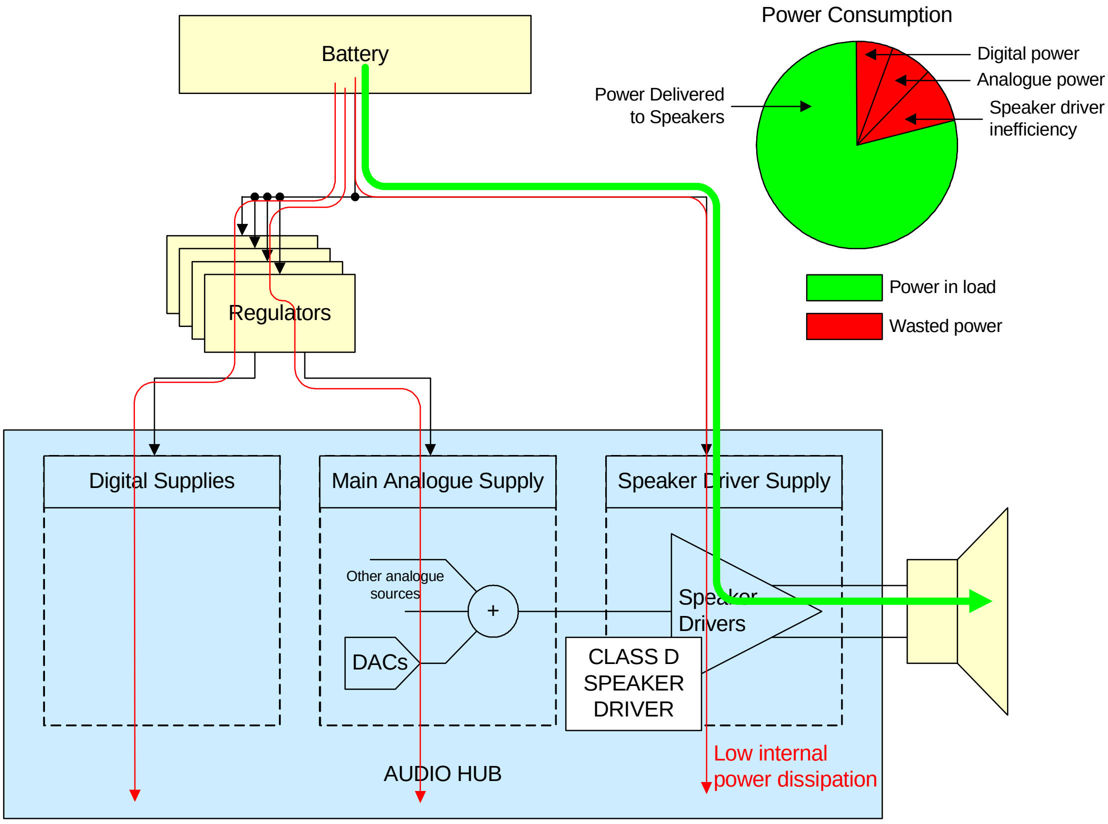


Figure 9: Class D Speaker Drivers: Increased Efficiency

In portable applications which support movie playback, gaming or other multimedia functions, the loudspeakers can be active for a significant period of time and Class D technology is highly effective at extending battery life. Even mobile phones which until recently spent relatively small periods of time with loudspeakers active (e.g. during ringtone playback), now support speakerphone and multimedia streaming functions which make use of the speaker drivers for much longer periods of time. For this reason, Class D is increasingly replacing class AB in mobile handset designs.

Pop and Click Suppression - Audible pops and clicks during the setting-up of audio circuitry degrade the listener's experience and a large amount of effort is usually put into eliminating these noises during system development. An audio hub device with integrated pop and click suppression mechanisms can further reduce development time and increase perceived levels of audio quality. Interestingly, high quality audio which is free of pops, clicks and other unpleasant noises also has the effect of increasing the perceived quality of video images.

Conclusion

With good design, audio hub devices can provide user benefits of hi-fi audio, enhanced handset functionality in a smaller form factor and extended battery life. For the handset designer the additional benefits of easy adoption, reduced component count and flexible power management features are compelling arguments for the use of these devices.

Class D technology is highly effective at extending battery life in portable multimedia devices, allowing new features such as gaming and TV streaming to operate for longer periods of time. High levels of silicon integration provide the obvious benefits of reduced PCB footprint and component count, but as hi-fi audio becomes more important there are other challenges to be met which have not previously been high priorities in handsets designed only for voice communication. For integration to provide real benefits to the end user, constant attention to audio quality is essential. Mixed signal designers must maintain audio quality throughout the signal chain, across multiple supply domains, and keep in mind the power supply constraints of mobile handsets. With good design, a better audio experience does not have to come with excessive power consumption or the expense of numerous additional components.

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