AUDIO CIRCUITS

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APPLICATION NOTE 3573

# Feature-Rich, Complete Audio Record/Playback for GSM/GPRS Cell Phones

*Short design cycles, coupled with PC-board area and cost pressures, continue to compel higher levels of integration in cell phone circuitry. Basic (low-tier) phone functionality is headed inexorably towards a one-chip solution. Meanwhile, in high- and mid-tier market segments in which model differentiation is critical, demand for high-performance, feature-rich peripheral components become increasingly important. Given that market push and the continually evolving feature set, a complete analog and digital-audio solution optimized for GSM/GPRS cell phones would provide a solid core for ongoing designs. The optimized solution must also include voice-band audio functionality (microphone, receiver speaker amplifier, ADC, and DAC) and flexible multimedia features (high-resolution ADC and DAC, voice recording, stereo microphones, headphones, and 8Ω speaker amplifiers). With that combination of features, the integration of both cell phone and application-based audio functions would be seamless.*

## Introduction

The complexity and high circuit density of cell phones challenge the system designer wishing to establish a highquality, audio record/replay path that meets provider specifications. New models with additional multimedia features, such as a still camera, ring-tone generators, MP3 replay, and voice memo, usually require incremental product changes. This necessitates not only new components, but also PC-board layout changes that can give rise to nonideal grounding and subsequent new noise problems.

Noise and interference issues in a cell phone analog-audio signal path can usually be attributed to demodulation of RF into the audio band or shared/nonideal grounding.

When subjected to high levels of RF energy from the phone antenna, the comparatively low-bandwidth audio circuitry in a phone can unintentionally demodulate the complex RF-transmitted signal. This degrades the noise floor of the audio path. While certain techniques and topologies can be applied to minimize this degradation within the audio amplifier circuitry, suppression components placed adjacent to the input pins are a cost-effective cure. Low-value capacitors-to-GND are often used, due to the designer selecting the minima of the capacitor's impedance to correspond with the carrier frequency of the radio.

One effective audio solution that minimizes shared/nonideal grounding integrates all the typically required analog-audio I/O functions into a single IC. This design transfers most of the problematic grounding issues from the PC-board layout engineer to the IC manufacturer. Besides including the necessary analog-audio I/O functions, that same IC must provide digital-audio interfacing sufficient to support voice band and any multimedia (i.e., application processor) functions. The IC also must provide partitioned shutdown control over the various blocks to maximize battery life.

The following discourse highlights some of the analog-/digital-audio issues that arise in single-chip implementation. The MAX9851 is used as an example of techniques and features that simplify GSM/GPRS cell phone design.

## Analog Audio—Minimizing Microphone Noise

High-gain audio circuits, such as microphone amplifiers (mic amps), are subject to degradation from poor grounding. This is particularly true for single-ended circuit topologies, in which small voltage differences between the mic-amp ground reference and the source ground reference (in this case, the GND pin of the mic capsule) are amplified into the signal path. In a complex product, like a cell phone, where audio ground planes are often shared with other circuits, degradation can be problematic, because the copper plane is not "zero ohms" (as is frequently assumed). Consequently, any current flowing through this finite resistance can cause small potential differences across the plane.

The grounding problem can be addressed with a mic amp that features a fully differential input. This approach, which allows remote sensing of the mic's GND pin, is incorporated into the MAX9851. Remote sensing forces any AC-voltage differences between the CODEC reference and the mic GND to appear as a common-mode signal to the mic amp. These differences are then reduced by the common-mode rejection ratio (CMRR) of the amp, thereby significantly attenuating the effective noise contributed to the signal path. The only penalties with this design are an extra PC-board trace to the mic from the CODEC and an extra coupling capacitor.

The MAX9851 also allows stereo, external mic inputs to be switched in to override the internal mic. These inputs typically would be sourced from a car kit or other external headset. In this case, using the amp input's CMRR, the EXTMICGND pin acts as a 'Kelvin sense' for both L and R channels, canceling ground differential noise in the same way as described above. The EXTMICGND PC-board trace should be extended to the GND connector of the car kit jack or headset connector for best results (**Figure 1**).

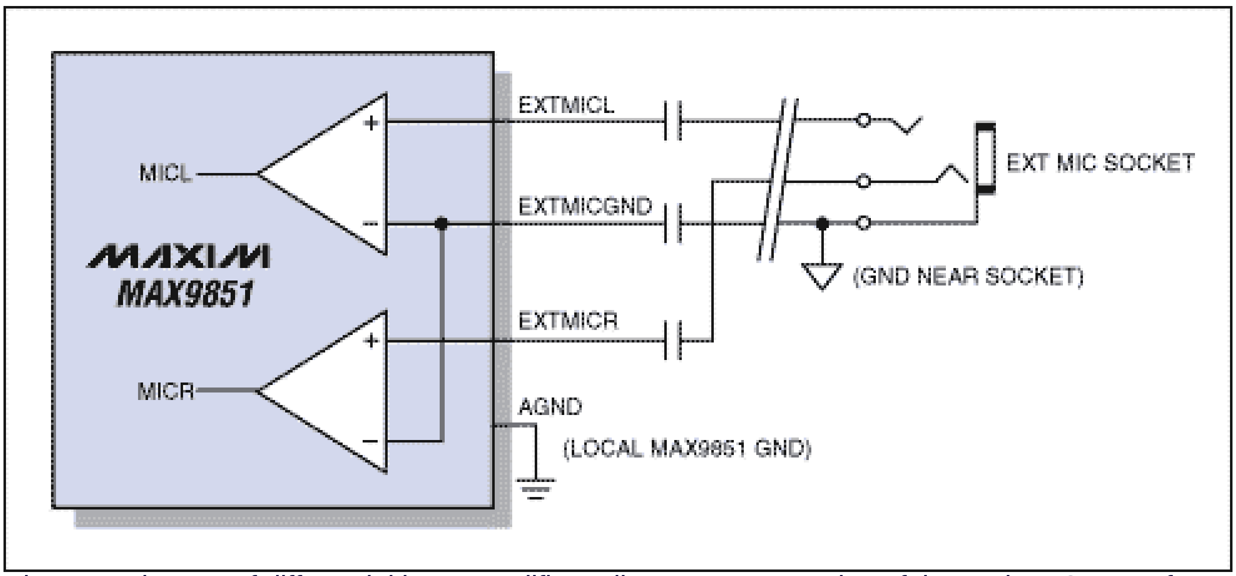


*Figure 1. The use of differential input amplifiers allows remote sensing of the socket "GND" reference. Any AC voltages between local and socket grounds are largely rejected, and not amplified by the mic amp gain.*

Mic bias circuits can introduce significant noise in the signal path. A large percentage of any bias output-voltage noise appears directly across the mic-amp input. More proficiently designed mic amps, as in the MAX9851, provide a regulated, low-noise bias voltage with output-noise levels matched to the internal mic-amp input noise.

## Analog Audio—Stereo DirectDrive™ Headphone and Receiver Outputs

The ability to replay compressed music files at near-CD quality places high demands on headphone audio reproduction. Signal-to-noise ratio (SNR), linearity, and bandwidth must be improved over the basic 300Hz to 4kHz voice-path requirement. Low-frequency extension can be problematic, as headphone drivers typically need a series capacitor to prevent the headphone amp's DC bias from appearing across the headphone transducer. The typical impedance range of common stereo headphones extends down to 16Ω, so that any series capacitor forms a highpass filter to low-frequency content. For extension of the listening response down to 100Hz, for example, two 100µF DC-blocking capacitors would be needed to guarantee 16Ω stereo headphone operation.

Use of Maxim's DirectDrive technique allows headphone operation without series capacitors, due to the amp outputs being referenced to 0V. The low-frequency content is then limited by either a DC-removal filter (digital source), as designed into the MAX9851, or by the input-coupling capacitors on the line or mic inputs (analog source). A further advantage of the DirectDrive design is the inherently low click/pop levels when bringing the device into or out of shutdown. As there are no series capacitors to charge or discharge, no net turn on/off current flows through the headphones.

The stereo headphones outputs of the MAX9851 are also capable of bridged-mono operation, (**Figure 2**) which enables compatibility for different headsets and accessories. The same socket could accommodate stereo headphones or a mono (mic plus hook switch and speaker) headset. The outputs remain ground referenced in this mode, so no DC voltage appears on the headset cable. Therefore, fault or short-circuit conditions are less problematic.

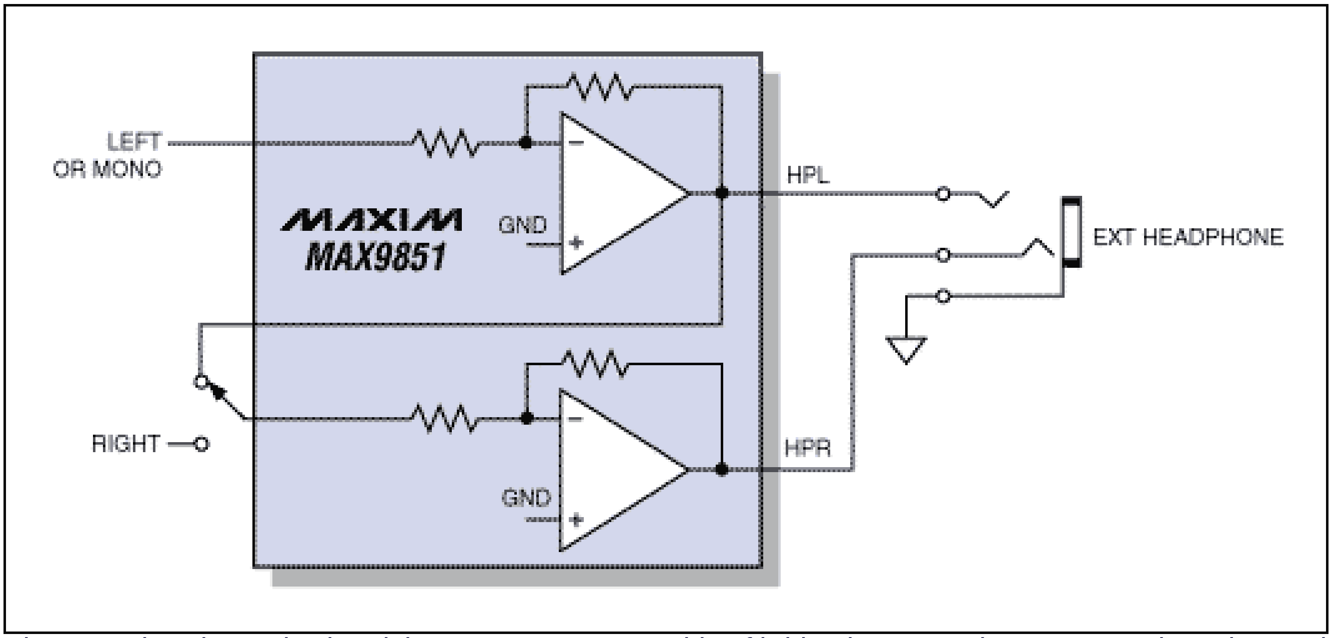


*Figure 2. The DirectDrive headphone outputs are capable of bridged mono and stereo operation. The Maxim proprietary GND-referenced output means no series coupling capacitors are necessary, saving cost and PC-board area.*

The receiver speaker output also uses the on-board charge pump in the DirectDrive design, so the output is single ended and the negative speaker connection is GND (0V). The output still has nearly the same voltage swing of a more typical BTL (differential) output, because the inverting charge pump provides a negative rail almost equal to that of the applied $AV_{DD}$. The resulting peak-to-peak output across the receiver speaker is almost 2 x $AV_{DD}$.

## Analog Audio—Class D Speaker Amplifiers

The MAX9851 incorporates Maxim's third-generation class D technology to drive 8Ω (or 4Ω) speakers. The main advantage of class D (switching) amps over class AB (linear) amps is efficiency. Class AB amps dissipate significant power in the output devices unless the amp is driven into clipping. However, because class D topologies have their output devices either on or off, their thermal dissipation is less and battery life can be extended. Extended battery life can be significant if a cell phone is used in speakerphone mode frequently, or supports push-to- talk (PTT) operation.

There are caveats to using class D topologies, however, especially in a product whose core function is RF transmission/ reception, such as a cell phone. The fast-switching edges associated with efficient class D amp operation can lead to RF-emission problems, especially with long PC-board traces and speaker leads. To counter the RF-emission problem, the MAX9851 stereo, class D speaker amps incorporate a proprietary EMI-reduction topology (active emission limiting) that suppresses high-frequency RF-emissions from speaker leads/traces at the expense of slight efficiency degradation. State-of-the-art IC construction techniques also minimize any interaction between the class D switching output stages and any of the sensitive, low-noise analog circuitry on the CODEC.

With the ability to connect to an unregulated, single-cell Li+ battery, the stereo amps are capable of 1W output from a 4.2V supply into an 8Ω speaker (**Figure 3**). More power is available if lower impedance speakers are used, but 4Ω speakers are not generally found in the smaller diameter drivers commonly used in cell phone designs.

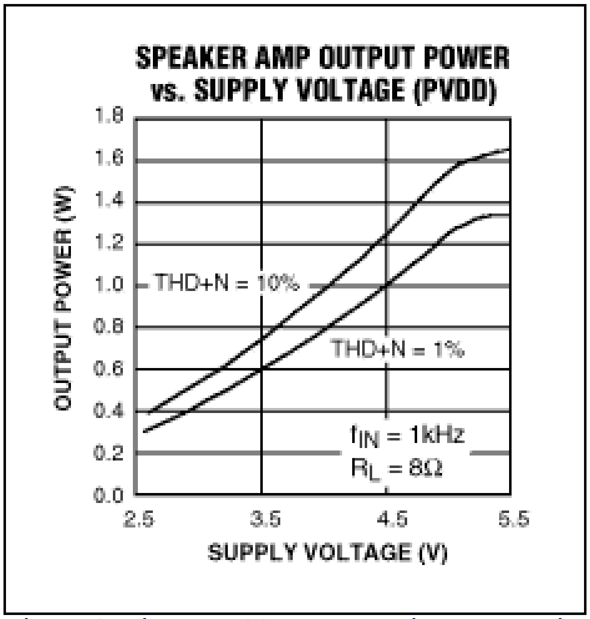


*Figure 3. The MAX9851 stereo class D speaker amplifier can operate from a raw battery supply, delivering 1W continuous (at 10% THD+N, 1kHz signal) from a 4.2V supply.*

## Digital Audio—General Architecture, Signal Flow

To allow conversations, the basic functionality of the GSM/GPRS cell phone must be supported by an 8kHz (or optional 16kHz) sampling ADC/DAC path, with a 16-bit depth in either direction. In the MAX9851, this I/O function is fully synchronous to the 13MHz (or 26MHz for $f_S$ = 16kHz) MCLK input to ensure no dropped or repeated samples. The S1 digital I/O lines in GSM voice mode (**Figure 4**) are used to access this basic functionality; the S1 digital interface can operate in MASTER or SLAVE mode.

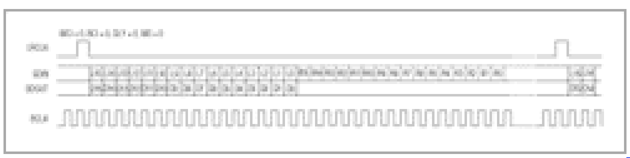

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*Figure 4. Basic GSM speech conversion functionality is supported by a generic, GSM voice-mode protocol on the S1 outputs. This can be operated as a MASTER or SLAVE; SLAVE mode expects both BCLK and LRCLK to be provided by the host.*

Many mid- and high-tier phones are commonly required to provide additional DAC functions at higher bit depths and sample rates. Examples of these functions are WMA/MP3 replay or generating WAV file ring tones. Combining digital-to-analog conversion for these functions with existing voice converters allows high integration, and gives one 'point source' for all data conversion. These features can be useful in product design, in which ground loops and audio level differences can prove problematic when trying to sum the two functions in the analog domain.

Therefore, having one converter to combine voice and multimedia data seems an ideal solution. The main difficulty with this approach is that any voice conversion must remain synchronous to the GSM/GPRS rate dictated by the MCLK input. The multimedia replay, moreover, often demands an unrelated sample rate: 44.1kHz or 48kHz, for example. The MAX9851 solves these challenges by implementing an algorithm similar to sample-rate conversion (SRC) on the digital-input data, thus allowing a single DAC to transform the summed voice and multimedia data in a synchronous manner.

In SLAVE mode, the incoming sample rate for the GSM voice data is necessarily sample-rate accurate (as dictated by MCLK). However, an internal digital PLL locks to the incoming LRCLK on the S2 digital input, allowing a precise (averaged over many samples) replication of nonsynchronous multimedia audio data. In MASTER mode, again the voice data is correctly aligned to the desired integer division of MCLK, but the S2 LRCLK data rate is approximated with a slight $f_S$ error, which is usually insignificant. Sample rates from 8kHz to 48kHz are supported on either S1 or S2 inputs.

The MAX9851 S2 digital I/Os have an interface that supports $I^2S$ and popular minor variations thereof. When not operating in GSM voice mode, the S1 interface can be programmed to support $I^2S$, maximizing the interface flexibility often needed in feature-rich, high-end phones.

## Digital Audio—GSM Filters

As can be seen in **Figure 5**, the S1 digital I/Os have extra filters that can be enabled in GSM voice mode. These digital blocks are an efficient implementation of tightly specified lowpass and highpass filters. This implementation suppresses energy near the Nyquist band edge and at low frequencies. The filters can prove beneficial to meet noise and signal-leakage envelopes when a phone undergoes testing and qualification. **Figure 6** gives the frequency response of enabled filters.

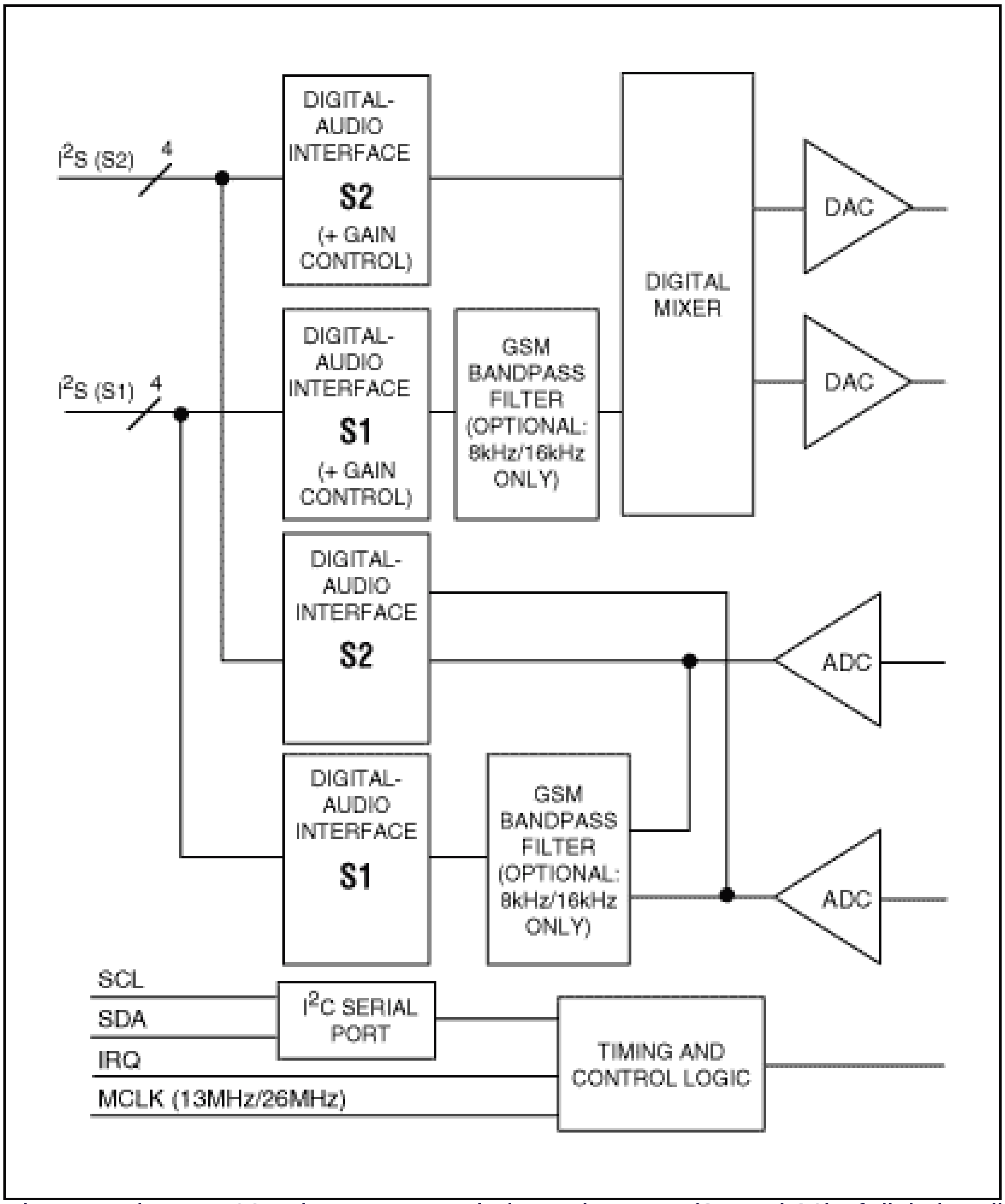


*Figure 5. The MAX9851 integrates two independent sets (S1 and S2) of digital-audio interface I/Os. For DAC replay, each interface can run at differing and noninteger-related sample rates in either MASTER or SLAVE modes.*

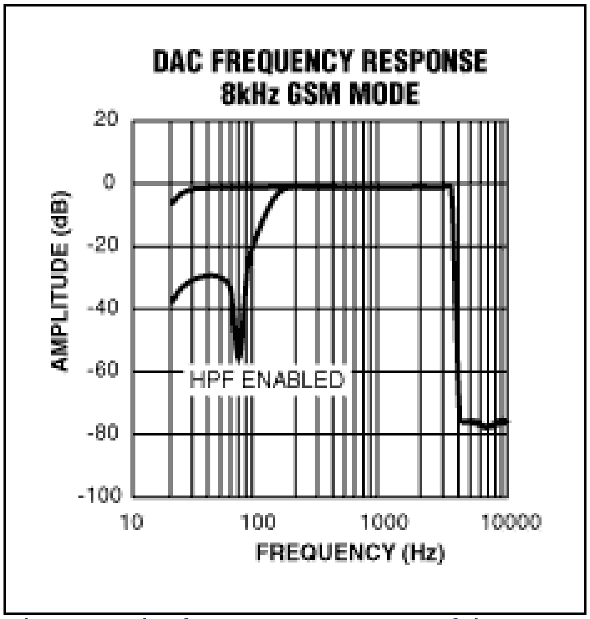


*Figure 6. The frequency response of the GSM replay path has enabled GSM filters. At $f_S$ = 8kHz, note the steep roll-off just before the Nyquist frequency (4kHz). The highpass filter (HPF) can be optionally defeated.*

## Summary

The previous examples highlight only a few issues that cell phone system designers/architects must address. Design cycles are notoriously short for this end application and feature sets continue to mature and change, almost on a per-model basis. Therefore, investing time in a well-engineered, flexible, comprehensive core chipset architecture is a worthwhile decision.

Controlling low-noise analog circuitry, which interfaces with multiple replay/record systems at different sample rates, is only part of the overall design task. It is also important to integrate the following features in one solution:

- Analog functionality and high performance
- A one-point, digital-/analog-audio interface
- Digital interfacing flexibility
- Comprehensive power management and partitioned shutdown

These features address a significant number of system-design, architecture, and topology issues. The MAX9851 is a 48-pin, 7mm x 7mm single-chip solution that resolves these issues, and forms the basis of either mid- or high-end GSM/GPRS cell phone audio designs.

**More Information**

MAX9851: QuickView -- -- Free Samples

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