*Click [here](https://www.maximintegrated.com/en/storefront/storefront.html) to ask about the production status of specific part numbers.*

# **MAX98396 20V Digital Input Class-DG Amplifier with I/V Sense and Brownout Prevention**

## <span id="page-0-0"></span>**General Description**

The MAX98396 is a high-efficiency, mono Class-DG speaker amplifier with industry-leading quiescent power featuring I/V sense, brownout-prevention engine (BPE), and dynamic-headroom tracking (DHT). The IC enables ultrasound applications by providing support for sample rates up to 192kHz, a higher passband (for  $f_s > 50$ kHz), and a bypass path for the ultrasound signals through the amplifier so it is not attenuated by the audio processing. Precision output current and voltage monitoring (I/V sense) enables the host device to run speaker protection algorithms. Spread-spectrum modulation (SSM) and edge rate control minimize EMI and eliminate the need for the output filtering found in traditional Class-D devices.

To achieve industry-leading quiescent power, the Class-DG amplifier employs two supply rails VBAT (3V to 5.5V) and PVDD (3V to 20V) to supply the speaker amplifier. The Class-DG amplifier switches between the two supply rails depending on the input signal level and/or the supply headroom. The brownout-prevention engine in the device allows it to reduce its contribution to the overall system power consumption by either attenuating or limiting the amplifier output when the device supply drops below a set of programmable thresholds. Additionally, as the power supply voltage varies due to sudden transients and declining battery life, DHT automatically optimizes the headroom available to the Class-DG amplifier to maintain consistent distortion and listening levels.

The device provides a PCM interface for audio data and a standard I2C interface for control data communication. The PCM interface supports audio playback using  $1^2$ S, left-justified, and TDM audio data formats. A unique clocking structure eliminates the need for an external master clock for PCM communication, which reduces pin count and simplifies board layout. Enabling thermal foldback automatically reduces the output power when the temperature exceeds a user specified threshold. This allows for uninterrupted music playback even at high ambient temperatures. Traditional thermal protection is also available in addition to robust overcurrent protection.

The device is available in a 0.4mm pitch, 35-bump waferlevel package (WLP). The device operates over the extended -40°C to +85°C temperature range.

## <span id="page-0-1"></span>**Applications**

- Mobile Speakers Smart Speakers Smart IoT Tablets
- Notebook Computers Soundbars

*SMBus is a trademark of Intel Corp.* 

### **Benefits and Features**

- Wide Input Supply Range (3.0V to 20V)
- Class-DG Operation Enables Industry-Leading Quiescent Power
	- 12.7mW at  $V_{\text{PVDD}}$  = 12V,  $V_{\text{VBAT}}$  = 3.8V
	- 18mW at  $V_{\text{PVDD}}$  = 19V,  $V_{\text{VBAT}}$  = 5.0V
- Ultra-Low Noise Floor
	- 15.5µV<sub>RMS</sub> Output Noise
	- 118dB Dynamic Range
- Low Distortion
	- -82dB THD+N at 1W into 8 $Ω$ ,  $f = 1$ kHz
	- -76dB THD+N at 1W into 8Ω,  $f = 6kHz$
- Output Power at 1% THD+N:
	- 20W into 8Ω,  $V_{\text{PVDD}}$  = 19V
	- 19W into 4Ω,  $V_{\text{PVDD}}$  = 14V
- $\bullet$  60W Peak Output Power into 4Ω, V<sub>PVDD</sub> = 19V
- Speaker Amplifier Efficiency:
	- 87% at 1W into 8Ω, V<sub>PVDD</sub> = 12V, V<sub>VBAT</sub> = 3.8V
	- 83% at 1W into 4Ω,  $V_{\text{PVDD}}$  = 12V,  $V_{\text{VBAT}}$  = 3.8V
	- 83% at 1W into 8Ω, V<sub>PVDD</sub> = 19V, V<sub>VBAT</sub> = 5.0V
	- 91% at 20W into 8Ω, V<sub>PVDD</sub> = 19V, V<sub>VBAT</sub> = 5.0V
- Class-D EMI Reduction Enables Filterless Operation
- Spread-Spectrum Modulation
- Switching-Edge Rate Control
- Integrated Speaker Current and Voltage Sense do not Require External Components
- Flexible Brownout-Prevention Engine
- $\bullet$  I<sup>2</sup>S/16-Channel TDM and I<sup>2</sup>C Digital Interfaces
- Playback Support for 16-, 24-, and 32-Bit Data Words
- Playback and I/V Sense Support Sample Rates up to 192kHz
- Audio Processing Bypass Path
- Dynamic-Headroom Tracking Maintains a Consistent Listening Experience
- Extensive Click-and-Pop Suppression
- 35-Bump, WLP (0.4mm Pitch)

*Ordering Information appears at end of data sheet.* 



## **Simplified Block Diagram**

<span id="page-1-0"></span>

## MAX98396

# 20V Digital Input Class-DG Amplifier with I/V Sense and Brownout Prevention

### **TABLE OF CONTENTS**



### TABLE OF CONTENTS (CONTINUED)



## MAX98396

# 20V Digital Input Class-DG Amplifier with I/V Sense and Brownout Prevention

### TABLE OF CONTENTS (CONTINUED)



## TABLE OF CONTENTS (CONTINUED)



### LIST OF FIGURES



## LIST OF FIGURES (CONTINUED)



### LIST OF TABLES



### <span id="page-9-0"></span>**Absolute Maximum Ratings**



All Other Digital Pins to DGND ............. -0.3V to  $V_{\text{DVDIO}} + 0.3V$ Short-Circuit Duration Between OUTP, OUTN, and PGND or PVDD or VBAT ...Continuous Short Circuit Duration Between OUTP and OUTN .....Continuous Continuous Power Dissipation for Multilayer Board ( $T_A$  = +70°C, derate 21.58mW/°C above +70°C) ..................... mW to 1.73mW Junction Temperature ...+150°C Operating Temperature Range .............................-40°C to +85°C Storage Temperature Range ..............................-65°C to +150°C Soldering Temperature (reflow) ..+260°C

Stresses beyond those listed under "Absolute Maximum Ratings" may cause permanent damage to the device. These are stress ratings only, and functional operation of the device at these or any other conditions beyond those indicated in the operational sections of the specifications is not implied. Exposure to absolute maximum rating conditions for *extended periods may affect device reliability.* 

### <span id="page-9-1"></span>**Package Information**

### <span id="page-9-2"></span>**35 WLP**



For the latest package outline information and land patterns (footprints), go to *[www.maximintegrated.com/packages](http://www.maximintegrated.com/packages)*. Note that a "+", "#", or "-" in the package code indicates RoHS status only. Package drawings may show a different suffix character, but the drawing pertains to the package regardless of RoHS status.

Package thermal resistances were obtained using the method described in JEDEC specification JESD51-7, using a four-layer board. For detailed information on package thermal considerations, refer to *[www.maximintegrated.com/](http://www.maximintegrated.com/thermal-tutorial) [thermal-tutorial](http://www.maximintegrated.com/thermal-tutorial)*.

### <span id="page-10-0"></span>**Electrical Characteristics**



### **Electrical Characteristics (continued)**



### **Electrical Characteristics (continued)**

(V<sub>VBAT</sub> = 5.0V, V<sub>PVDD</sub> = 19V, V<sub>AVDD</sub> = 1.8V, V<sub>DVDD</sub> = 1.2V, V<sub>DVDDIO</sub> = RESET = 1.2V, C<sub>VBAT</sub> = 1x10µF, 1x0.1µF, C<sub>PVDD</sub> = 1x220µF, 2x10µF, 2x0.1µF, С<sub>АVDD</sub> = 1µF, С<sub>DVDD</sub> = 1µF, С<sub>DVDDIO</sub> = 0.1µF, С<sub>VREFC</sub> = 1µF, Z<sub>SPK</sub> = Open, f<sub>s</sub> = 48kHz, AC Measurement Bandwidth = 20Hz to 22kHz, SPK\_GAIN\_MAX = 0xF (19dB), Data Width = 24-bit, DG Mode, T<sub>A</sub> = T<sub>MIN</sub> to T<sub>MAX</sub> unless otherwise noted, Typical values are at  $T_A$  = +25°C) (Note 2)



**Note 1:** Package thermal resistances were obtained using the method described in JEDEC specification JESD51-7, using a four-layer board. For detailed information on package thermal considerations, refer to *[www.maximintegrated.com/thermal-tutorial](http://www.maximintegrated.com/thermal-tutorial)*.

**Note 2:** 100% production tested at T<sub>A</sub> = +25°C. Specifications over temperature limits are guaranteed by design. Typical values are based on one sigma characterization data unless otherwise noted.

**Note 3:** Minimum and/or maximum limit is guaranteed by design and by statistical analysis of device characterization data. The specification is not guaranteed by production testing.

**Note 4:** Assumes device is fully programmed (SPK\_EN = 1) and EN = 1 is the last I<sup>2</sup>C write in the sequence.

**Note 5:** Digital filter performance is invariant over temperature and is production tested at T<sub>A</sub> = +25°C.

**Note 6:** Applies to all transitions in/out of full operation with noise gate enabled/disabled. Does not include state transitions due to fault conditions.

### <span id="page-25-0"></span>**Typical Operating Characteristics**

 $(V_{VBAT} = 5.0V, V_{DVDD} = 1.2V, DVDDIO = 1.2V, V_{AVDD} = 1.8V, V_{GND} = 0V, V_{PGND} = 0V, V_{PVDD} = 19V, C_{VBAT} = 10 \mu F + 0.1 \mu F,$  $C_{\text{PVDD}}$  = 0.1µF + 10µF + 220µF,  $C_{\text{DVDDIO}}$  = 0.1µF,  $C_{\text{DVDD}}$  = 1µF,  $C_{\text{VAVDD}}$  = 1µF,  $C_{\text{VREFC}}$  = 1µF,  $A_{\text{V}}$  = 19dB,  $Z_{\text{SPK}}$  =  $\infty$  between OUTP and OUTN, AC Measurement Bandwidth = 20Hz to 20kHz,  $f_S$  = 48kHz, 24-bit data,  $f_{BCLK}$  = 3.072MHz. Typical values are at T<sub>A</sub>  $= +25^{\circ}C$ 



### **Typical Operating Characteristics (continued)**

 $(V_{VBAT} = 5.0V, V_{DVDD} = 1.2V, DVDDIO = 1.2V, V_{AVDD} = 1.8V, V_{GND} = 0V, V_{PGND} = 0V, V_{PVDD} = 19V, C_{VBAT} = 10 \mu F + 0.1 \mu F,$  $C_{\text{PVDD}} = 0.1 \mu\text{F} + 10 \mu\text{F} + 220 \mu\text{F}$ ,  $C_{\text{DVDDIO}} = 0.1 \mu\text{F}$ ,  $C_{\text{DVDD}} = 1 \mu\text{F}$ ,  $C_{\text{VAVDD}} = 1 \mu\text{F}$ ,  $C_{\text{VREFC}} = 1 \mu\text{F}$ ,  $A_{\text{V}} = 19 \mu\text{B}$ ,  $Z_{\text{SPK}} = \infty$  between OUTP and OUTN, AC Measurement Bandwidth = 20Hz to 20kHz,  $f_S$  = 48kHz, 24-bit data,  $f_{BCLK}$  = 3.072MHz. Typical values are at T<sub>A</sub>  $= +25^{\circ}C$ 



### **Typical Operating Characteristics (continued)**

 $(V_{VBAT} = 5.0V, V_{DVDD} = 1.2V, DVDDIO = 1.2V, V_{AVDD} = 1.8V, V_{GND} = 0V, V_{PGND} = 0V, V_{PVDD} = 19V, C_{VBAT} = 10 \mu F + 0.1 \mu F,$  $C_{\text{PVDD}} = 0.1 \mu\text{F} + 10 \mu\text{F} + 220 \mu\text{F}$ ,  $C_{\text{DVDDIO}} = 0.1 \mu\text{F}$ ,  $C_{\text{DVDD}} = 1 \mu\text{F}$ ,  $C_{\text{VREDD}} = 1 \mu\text{F}$ ,  $C_{\text{VREFC}} = 1 \mu\text{F}$ ,  $A_V = 19 \text{dB}$ ,  $Z_{\text{SPK}} = \infty$  between OUTP and OUTN, AC Measurement Bandwidth = 20Hz to 20kHz,  $f_S$  = 48kHz, 24-bit data,  $f_{BCLK}$  = 3.072MHz. Typical values are at T<sub>A</sub>  $= +25^{\circ}C$ 



toc21

toc24

toc27

### **Typical Operating Characteristics (continued)**

 $(V_{VBAT} = 5.0V, V_{DVDD} = 1.2V, DVDDIO = 1.2V, V_{AVDD} = 1.8V, V_{GND} = 0V, V_{PGND} = 0V, V_{PVDD} = 19V, C_{VBAT} = 10 \mu F + 0.1 \mu F,$  $C_{\text{PVDD}}$  = 0.1µF + 10µF + 220µF,  $C_{\text{DVDDIO}}$  = 0.1µF,  $C_{\text{DVDD}}$  = 1µF,  $C_{\text{VAVDD}}$  = 1µF,  $C_{\text{VREFC}}$  = 1µF,  $A_{\text{V}}$  = 19dB,  $Z_{\text{SPK}}$  =  $\infty$  between OUTP and OUTN, AC Measurement Bandwidth = 20Hz to 20kHz,  $f_S$  = 48kHz, 24-bit data,  $f_{BCLK}$  = 3.072MHz. Typical values are at T<sub>A</sub>  $= +25^{\circ}C$ 



### **Typical Operating Characteristics (continued)**

 $(V_{VBAT} = 5.0V, V_{DVDD} = 1.2V, DVDDIO = 1.2V, V_{AVDD} = 1.8V, V_{GND} = 0V, V_{PGND} = 0V, V_{PVDD} = 19V, C_{VBAT} = 10 \mu F + 0.1 \mu F,$  $C_{\text{PVDD}} = 0.1 \mu\text{F} + 10 \mu\text{F} + 220 \mu\text{F}$ ,  $C_{\text{DVDDIO}} = 0.1 \mu\text{F}$ ,  $C_{\text{DVDD}} = 1 \mu\text{F}$ ,  $C_{\text{VAVDD}} = 1 \mu\text{F}$ ,  $C_{\text{VREFC}} = 1 \mu\text{F}$ ,  $A_{\text{V}} = 19 \mu\text{F}$ ,  $A_{\text{V}} = 19 \mu\text{F}$ ,  $Z_{\text{SPK}} = \infty$  between OUTP and OUTN, AC Measurement Bandwidth = 20Hz to 20kHz,  $f_S$  = 48kHz, 24-bit data,  $f_{BCLK}$  = 3.072MHz. Typical values are at T<sub>A</sub>  $= +25^{\circ}C$ 



## **Typical Operating Characteristics (continued)**

 $(V_{VBAT} = 5.0V, V_{DVDD} = 1.2V, DVDDIO = 1.2V, V_{AVDD} = 1.8V, V_{GND} = 0V, V_{PGND} = 0V, V_{PVDD} = 19V, C_{VBAT} = 10 \mu F + 0.1 \mu F,$  $C_{\text{PVDD}} = 0.1 \mu\text{F} + 10 \mu\text{F} + 220 \mu\text{F}$ ,  $C_{\text{DVDDIO}} = 0.1 \mu\text{F}$ ,  $C_{\text{DVDD}} = 1 \mu\text{F}$ ,  $C_{\text{VAVDD}} = 1 \mu\text{F}$ ,  $C_{\text{VREFC}} = 1 \mu\text{F}$ ,  $A_{\text{V}} = 19 \mu\text{B}$ ,  $Z_{\text{SPK}} = \infty$  between OUTP and OUTN, AC Measurement Bandwidth = 20Hz to 20kHz,  $f_S$  = 48kHz, 24-bit data,  $f_{BCLK}$  = 3.072MHz. Typical values are at T<sub>A</sub>  $= +25^{\circ}C$ 



### **Typical Operating Characteristics (continued)**

 $(V_{VBAT} = 5.0V, V_{DVDD} = 1.2V, DVDDIO = 1.2V, V_{AVDD} = 1.8V, V_{GND} = 0V, V_{PGND} = 0V, V_{PVDD} = 19V, C_{VBAT} = 10 \mu F + 0.1 \mu F,$  $C_{\text{PVDD}} = 0.1 \mu\text{F} + 10 \mu\text{F} + 220 \mu\text{F}$ ,  $C_{\text{DVDDIO}} = 0.1 \mu\text{F}$ ,  $C_{\text{DVDD}} = 1 \mu\text{F}$ ,  $C_{\text{VAVDD}} = 1 \mu\text{F}$ ,  $C_{\text{VREFC}} = 1 \mu\text{F}$ ,  $A_{\text{V}} = 19 \mu\text{F}$ ,  $A_{\text{V}} = 19 \mu\text{F}$ ,  $Z_{\text{SPK}} = \infty$  between OUTP and OUTN, AC Measurement Bandwidth = 20Hz to 20kHz,  $f_S$  = 48kHz, 24-bit data,  $f_{BCLK}$  = 3.072MHz. Typical values are at T<sub>A</sub>  $= +25^{\circ}C$ 



### **Typical Operating Characteristics (continued)**

 $(V_{VBAT} = 5.0V, V_{DVDD} = 1.2V, DVDDIO = 1.2V, V_{AVDD} = 1.8V, V_{GND} = 0V, V_{PGND} = 0V, V_{PVDD} = 19V, C_{VBAT} = 10 \mu F + 0.1 \mu F,$  $C_{\text{PVDD}} = 0.1 \mu\text{F} + 10 \mu\text{F} + 220 \mu\text{F}$ , C<sub>DVDDIO</sub> = 0.1 $\mu\text{F}$ , C<sub>DVDD</sub> = 1 $\mu\text{F}$ , C<sub>VAVDD</sub> = 1 $\mu\text{F}$ , C<sub>VREFC</sub> = 1 $\mu\text{F}$ , A<sub>V</sub> = 19dB, Z<sub>SPK</sub> =  $\infty$  between OUTP and OUTN, AC Measurement Bandwidth = 20Hz to 20kHz,  $f_S$  = 48kHz, 24-bit data,  $f_{BCLK}$  = 3.072MHz. Typical values are at T<sub>A</sub>  $= +25^{\circ}C$ 



## **Typical Operating Characteristics (continued)**

 $(V_{VBAT} = 5.0V, V_{DVDD} = 1.2V, DVDDIO = 1.2V, V_{AVDD} = 1.8V, V_{GND} = 0V, V_{PGND} = 0V, V_{PVDD} = 19V, C_{VBAT} = 10 \mu F + 0.1 \mu F,$  $C_{\text{PVDD}} = 0.1 \mu\text{F} + 10 \mu\text{F} + 220 \mu\text{F}$ ,  $C_{\text{DVDDIO}} = 0.1 \mu\text{F}$ ,  $C_{\text{DVDD}} = 1 \mu\text{F}$ ,  $C_{\text{VAVDD}} = 1 \mu\text{F}$ ,  $C_{\text{VREFC}} = 1 \mu\text{F}$ ,  $A_V = 19 \text{dB}$ ,  $Z_{\text{SPK}} = \infty$  between OUTP and OUTN, AC Measurement Bandwidth = 20Hz to 20kHz,  $f_S$  = 48kHz, 24-bit data,  $f_{BCLK}$  = 3.072MHz. Typical values are at T<sub>A</sub>  $= +25^{\circ}C$ 



### **Typical Operating Characteristics (continued)**

(VVBAT = 5.0V, VDVDD = 1.2V, DVDDIO = 1.2V, VAVDD = 1.8V, VGND = 0V, VPGND = 0V, VPVDD = 19V, CVBAT = 10µF + 0.1µF, C<sub>PVDD</sub> = 0.1µF + 10µF + 220µF, C<sub>DVDDIO</sub> = 0.1µF, C<sub>DVDD</sub> = 1µF, C<sub>VAVDD</sub> = 1µF, C<sub>VREFC</sub> = 1µF, A<sub>V</sub> = 19dB, Z<sub>SPK</sub> = ∞ between OUTP and OUTN, AC Measurement Bandwidth = 20Hz to 20kHz,  $f_S$  = 48kHz, 24-bit data,  $f_{BCLK}$  = 3.072MHz. Typical values are at T<sub>A</sub>  $= +25^{\circ}C$ 



## <span id="page-35-0"></span>**Pin Configuration**

### **35 WLP**

<span id="page-35-1"></span>

### <span id="page-35-2"></span>**Pin Description**


# **Pin Description (continued)**



## **Functional Diagrams**

## **Detailed Block Diagram**



## **Detailed Description**

### **Device State Control**

The device has three distinct power states: the hardware shutdown state, software shutdown state, and active state. When transitioning between states, the device always moves from the hardware shutdown state to the software shutdown state to the active state (or the reverse) based on the state transition requirements. Normal transitions between the software shutdown state and active state are reversible without waiting for an in-progress transition to be completed. State transitions due to fault conditions, supply removal, and reset conditions are not reversible and are always completed (once initiated) to protect the device.

#### **Hardware Shutdown State**

When the device is first powered up or after a hardware reset event, the device always initializes into the hardware shutdown state. In hardware shutdown, the device is configured to its lowest power state. Upon entering hardware shutdown, the device is globally placed into a reset condition. As a result, the  $12C$  control interface is disabled and all device registers are returned to their PoR states. When exiting hardware shutdown, the device initializes and then transitions into the software shutdown state. During this transition (as part of initialization), the OTP register trim settings are loaded. If the OTP load routine fails to complete successfully, an *[OTP\\_FAIL\\_\\*](#page-95-0)* interrupt is generated once the device reaches the software shutdown state.

When the hardware reset input (RESET) is asserted low, the device enters (or remains in) hardware shutdown. The device is also placed into hardware shutdown anytime the AVDD, DVDD, or DVDDIO supplies drop below their respective UVLO thresholds. The device only exits hardware shutdown when the AVDD, DVDD, and DVDDIO supplies are all above their respective UVLO thresholds and the hardware reset input (RESET) is asserted high. Once all of these conditions are met, the device automatically exits hardware shutdown and transitions into software shutdown.

### **Software Shutdown State**

The device enters the software shutdown state after it transitions out of the hardware shutdown state and when exiting the active state. In the software shutdown state, all blocks are automatically disabled except for the I2C control interface. In the software shutdown state, all device registers can be programmed without restriction and all programmed register states are retained.

The global enable bit (*[EN](#page-168-0)*) is used to transition the device into and out of software shutdown. When global enable (*[EN](#page-168-0)*) is set high, the device transitions to the active state and a power-up done (*[PWRUP\\_DONE\\_\\*](#page-96-0)*) interrupt is generated. When the device is in the active state and global enable (*[EN](#page-168-0)*) is set low, the device transitions to the software shutdown state and a power-down done (*[PWRDN\\_DONE\\_\\*](#page-96-1)*) interrupt is generated. Additionally, the device is reset and enters software shutdown anytime the global enable bit (*[RST](#page-94-0)*) is written with a 1.

By default, the device supply configuration is PVDD and VBAT pins being supplied with voltages and in this scenario, regardless of the state of the global enable bit, the device cannot transition from the software shutdown state to the active state until PVDD and VBAT are all above their UVLO thresholds. If PVDD or VBAT supplies drop below their UVLO levels while the device is in the active state, the device is forced back into the software shutdown state.

In systems where the VBAT supply is not available, the device is configured to operate with PVDD only by connecting the VBAT pin to the VREFC pin on the PCB and setting the NOVBAT bit to 1. In this scenario, regardless of the state of the global enable bit, the device cannot transition from the software shutdown state to the active state until PVDD is above its UVLO thresholds. If PVDD supply drops below its UVLO levels while the device is in the active state, the device is forced back into the software shutdown state.

While in the software shutdown state, the PVDD and VBAT (if applicable) supplies can be powered down safely.

#### **Recovery from Software Shutdown due to Supply Faults**

The device provides two forms of fault recovery in the event that either VBAT or PVDD drop below their UVLO thresholds while the device is in its active state. Based on the setting of the *[VBAT\\_AUTORESTART\\_EN](#page-168-1)* and *[PVDD\\_AUTORESTART\\_EN](#page-168-2)* bits, the individual supply fault recovery is either in manual mode or auto restart mode.

If the bit is set low, then the supply UVLO fault recovery is in manual mode. In manual mode, when the supply drops below its UVLO threshold, the device transitions into the software shutdown state (sets *[EN](#page-168-0)* = 0) and generates the appropriate UVLO fault shutdown interrupt (*[VBAT\\_UVLO\\_SHDN\\_\\*](#page-96-2)* or *[PVDD\\_UVLO\\_SHDN\\_\\*](#page-96-3)* respectively). Even once the supply recovers (when voltage levels exceed the UVLO thresholds), the device remains in the software shutdown state until the global enable bit (*[EN](#page-168-0)*) is set high by the host software.

If the bit is instead set high, then the supply UVLO fault recovery is in auto restart mode. In auto restart mode, when the supply drops below its UVLO threshold, the device is internally forced into software shutdown (*[EN](#page-168-0)* state is preserved and remains high) and generates the appropriate UVLO fault shutdown interrupt (*[VBAT\\_UVLO\\_SHDN\\_\\*](#page-96-2)* or *[PVDD\\_UVLO\\_SHDN\\_\\*](#page-96-3)* respectively). Once the supply recovers (voltage levels exceed the UVLO thresholds), the device is no longer held in software shutdown and automatically restarts back into the active state (if all other conditions are met). These recovery modes do not apply when the AVDD, DVDD, or DVDDIO supplies cause a UVLO fault while the device is in the active state. If AVDD, DVDD, or DVDDIO drop below their UVLO thresholds, the device is reset and is placed into hardware shutdown.

### **Active State**

The device always enters the active state through a transition from the software shutdown state. In the active state, all enabled device blocks are active and speaker amplifier playback is possible. In the active state, only dynamic register settings (or those restricted to disabled blocks) can be programmed safely.

The only non-fault state transitions to or from the active state are those initiated through the global enable bit (*[EN](#page-168-0)*). All other transitions to or from the active state are the result of fault events, and can result in audible glitches if they occur during active playback.

### **Device Sequencing**

[Table 1](#page-39-0) and [Table 2](#page-39-1) show the recommended typical device power-up and power-down sequences.

## <span id="page-39-0"></span>**Table 1. Typical Power-Up Sequence**



## <span id="page-39-1"></span>**Table 2. Typical Power-Down Sequence**



## **Table 2. Typical Power-Down Sequence (continued)**



## <span id="page-40-1"></span>**PCM Interface**

The flexible PCM slave interface supports common audio playback sample rates from 16kHz to 192kHz and I/V sense ADC sample rates from 8kHz to 192kHz. The PCM interface also supports standard I2S, left-justified, and TDM data formats. The PCM interface is disabled and powered down when both the PCM data input (DIN) and PCM data output (DOUT) are disabled.

### **PCM Clock Configuration**

The PCM slave interface requires the host to supply both BCLK and LRCLK. To configure the PCM interface clock inputs, the host must program both the device interface sample rate (*[PCM\\_SR](#page-121-0)*) and BCLK to LRCLK (*[PCM\\_BSEL](#page-120-0)*) ratio. The PCM interface sample rate must be configured to match the frequency of the frame clock (LRCLK) using the *[PCM\\_SR](#page-121-0)*  registers. The speaker path sample rate is also set by the *[PCM\\_SR](#page-121-0)* setting. However, the I/V sense ADC path sample rate (*[IVADC\\_SR\)](#page-121-1)* can be set to the same rate or lower rate than the speaker path sample rate (*[PCM\\_SR](#page-121-0)*) according to the restrictions in [Table 3.](#page-40-0) When the I/V sense ADC path is set to a lower rate than the speaker amplifier path, the output data contains repeated samples.



## <span id="page-40-0"></span>**Table 3. Sample Rate Selection for I/V Sense**

The device supports a range of BCLK to LRCLK clock ratios (*[PCM\\_BSEL](#page-120-0)*) ranging from 32 to 512. However, based on the selected PCM interface sample rate (LRCLK frequency), the configured clock ratio cannot result in a BLCK frequency that exceeds 24.576MHz.

## **PCM Data Format Configuration**

The device supports the standard I<sup>2</sup>S, left-justified, and TDM data formats. The operating mode is configured using the *[PCM\\_FORMAT](#page-119-0)* bit field.

## **I 2S/Left-Justified Mode**

I 2S and left-justified formats support two channels that can be 16-, 24-, or 32-bits in length. The BCLK to LRCLK ratio (*[PCM\\_BSEL](#page-120-0)*) must be configured to be twice the desired channel length. The audio data word size is configurable to 16-, 24-, or 32-bits in length (*[PCM\\_CHANSZ](#page-119-1)*), but must be programmed to be less than or equal to the channel length. If

the resulting channel length exceeds the configured data word size then the data input LSBs are truncated and the data output LSBs are padded with either zero or Hi-Z data based on the *[PCM\\_TX\\_EXTRA\\_HIZ](#page-120-1)* register bit setting.

## **Table 4. Supported I2S/Left-Justified Mode Configurations**



With the default PCM settings, falling LRCLK indicates the left channel data (Channel 0) and the start of a new frame while rising LRCLK indicates the right channel data (Channel 1). In I<sup>2</sup>S mode, the MSB of the audio word is latched on the second active BCLK edge after an LRCLK transition. In left-justified mode, the MSB of the audio word is latched on the first active BCLK edge after an LRCLK transition.

The [PCM\\_BCLKEDGE](#page-120-2) register bit selects either the rising or falling edge of BCLK as the active edge that is used for data capture (DIN) and data output (DOUT). The *[PCM\\_CHANSEL](#page-119-2)* bit configures which LRCLK edge indicates the start of a new frame (Channel 0), and LRCLK transitions always align with the inactive BCLK edge. The data output is valid on the same active BCLK edge as the data input. The data output also transitions on the same edge as the data input.



*Figure 1. Standard I2S Mode* 



*Figure 2. Left-Justified Mode* 



### *Figure 3. Left-Justified Mode (LRCLK Inverted)*



*Figure 4. Left-Justified Mode (BCLK Inverted)* 

### **TDM Modes**

The provided TDM modes support timing for up to 16 digital audio input channels (DIN), each containing 16-, 24-, or 32-bits of data. The digital audio output (DOUT) is structured into 8-bit slots, and the timing can support up to a maximum of 128 data output slots. The number of TDM input channels and output slots is determined by both the selected BCLK to LRCLK ratio (*[PCM\\_BSEL](#page-120-0)*) and the selected data word and channel length (*[PCM\\_CHANSZ](#page-119-1)*).

For a given valid configuration, the number of available data input channels per frame is calculated as follows:

Number of Available Data Input Channels = BCLK to LRCLK Ratio/Channel Length

For a given valid configuration, the number of available 8-bit data output slots per frame is calculated as follows:

Number of Available Data Output Slots = BCLK to LRCLK Ratio/8

[Table 5](#page-42-0) shows the supported TDM mode configurations for each combination of input data channels and output data slots. In some configurations, the maximum PCM interface and speaker amplifier playback sample rate is limited to less than 96kHz to avoid violating the BCLK frequency limit of 24.576MHz.

## <span id="page-42-0"></span>**Table 5. Supported TDM Mode Configurations**





## **Table 5. Supported TDM Mode Configurations (continued)**

With the default PCM interface settings in TDM mode, a rising frame clock (LRCLK) edge acts as the frame sync pulse and indicates the start of a new frame. The frame sync pulse width must be equal to at least one bit clock period. However, the falling edge can occur at any time as long as it does not violate the setup time of the next frame sync pulse rising edge. The *[PCM\\_CHANSEL](#page-119-2)* bit can be used to invert the LRCLK edges (sync pulse) used to start a TDM frame.

In TDM mode, the MSB of the first audio word can be latched on the first (TDM Mode 0), second (TDM Mode 1), or third (TDM Mode 2) active BCLK edge after the sync pulse and is programmed by the *[PCM\\_FORMAT](#page-119-0)* bits. Additionally, the *[PCM\\_BCLKEDGE](#page-120-2)* register bit allows the BLCK edge (that is used for data capture and data output) to be programmed. The data output is valid on the same active BCLK edge as the data input. The data output also transitions on the same edge as the data input.



#### *Figure 5. TDM Modes*

## **PCM Data Path Configuration**

The PCM interface data input (DIN) receives the source data for the speaker amplifier path and the speaker audio processing bypass path while the data output (DOUT) transmits the data from the I/V sense ADC path. In addition, the PCM data output can also transmit internal diagnostic data such as the speaker DSP monitor path, supply measurement ADC results, device status reporting, and the DHT attenuation level.

### **PCM Data Input**



*Figure 6. PCM Data Input* 

The PCM playback path is enabled with the *[PCM\\_RX\\_EN](#page-128-0)* bit and can accept data from any valid input data channel. The device provides an input digital mono mixer that can route a single channel or mix two PCM input channels to create a mono input to the speaker playback path. The *[PCM\\_DMMIX\\_CFG](#page-126-0)* [b](#page-126-0)it is used to configure the mixer, while the *[PCM\\_DMMIX\\_CH0\\_SOURCE](#page-127-0)* and *[PCM\\_DMMIX\\_CH1\\_SOURCE](#page-126-1)* bits select which of the 16 PCM input channels are used as the input to the mono mixer. In I<sup>2</sup>S and left-justified modes, only two input data channels are available while in TDM mode up to 16 channels of input data can be available. If the PCM playback path is disabled (*[PCM\\_RX\\_EN](#page-128-0)* = 0), a zero code value is driven into the speaker amplifier path.

The device also supports an audio processing bypass path from the PCM input to speaker amplifier output that bypasses the audio processing blocks like the volume control, DHT, BPE, and thermal foldback circuits. The audio processing bypass path is enabled by setting the *[PCM\\_BYP\\_EN](#page-128-1)* bit field to 1. The PCM data input for the audio processing bypass path is selected with the *[PCM\\_BYPASS\\_SOURCE](#page-127-1)* bit field. If the PCM audio processing bypass path is disabled (*[PCM\\_BYP\\_EN](#page-128-1)* = 0), a zero code value is driven into the audio processing bypass path.

## **PCM Data Output**

The PCM interface data output (DOUT) is enabled by the *[PCM\\_TX\\_EN](#page-128-2)* bit field, and can transmit any output data type onto any valid output channel or slot. In I2S and left-justified mode, only two data output channels are available in each output transmit frame (Channel 0 and 1). In TDM mode, each output transmit frame can contain up to 64 sequential 8-bit data output slots, each of which are numbered from 0 up to a maximum of 63.

The PCM data output can transmit several different output data types. In I<sup>2</sup>S and left-justified modes, only the speaker amplifier output voltage sense, output current sense, and DSP monitor output data types are available for data output transmission. If the word size of the data output type is longer than output channel data word (*[PCM\\_CHANSZ](#page-119-1)*), the lowest trailing bits are truncated.

In TDM mode, all output data types are available and are individually assigned to data output slots. The output data types vary in word size from 3 bits to 32 bits, and as a result require from 1 to 4 data output slots to transmit in TDM mode. [Table 6](#page-46-0) shows the supported output data types and the parameters of each data type.



## <span id="page-46-0"></span>**Table 6. Supported PCM Data Output Types**

An individual enable and slot assignment bit field is provided for each output data type. In I<sup>2</sup>S and left-justified modes, use output slot 0 to assign data to Channel 0 and output slot 1 to assign data to Channel 1. In TDM mode, the slot assignment selects the slot where the output data type transmit begins for data output types requiring more than one slot to transmit (e.g., a two slot data type assigned to slot 6 would occupy slots 6 and 7).

In TDM mode, each data type can be assigned to any valid data output slot (or series of slots) with some restrictions. First, it is invalid for data types to be assigned such that the data word extends beyond the end of the data output frame. For example, data types that require two slots to transmit cannot be assigned to the last slot of the frame. Next, it is also invalid to assign a data output type to any slot that overlaps with the slot assignment of another data type (this also applies to channels in I<sup>2</sup>S and left-justified modes). Finally, it is invalid to assign a data type to any slots that do not exist in the frame structure of the current PCM interface configuration.

Any data output (DOUT) slots that exist in the current frame structure but have no output data type assigned to them are either Hi-Z or driven with a 0 code (as set by the *[PCM\\_TX\\_SLOT\\_HIZ](#page-124-1)* bit field). Likewise, if a data output type is disabled, then the assigned data output slot(s) are also either Hi-Z or driven with a 0 code (as set by the *[PCM\\_TX\\_SLOT\\_HIZ](#page-124-1)* bit field).

### **Data Output Channel-Interleaved I/V Sense Data**

In I<sup>2</sup>S and left-justified use cases, the PCM interface limits the number of available data output channels to two making it impossible to fit amplifier output current and voltage sense data from stereo devices on a single shared data output (DOUT) line. For these cases, the data output can be configured to allow the current and voltage sense data types from a single device to share a single data output channel. To enable channel-interleaved mode, set the *[PCM\\_TX\\_INTERLEAVE](#page-119-3)* bit high. Then assign the current and voltage sense data types to the same valid data channel (using *[PCM\\_VMON\\_SLOT](#page-121-2)* and *[PCM\\_IMON\\_SLOT](#page-122-0)*).

In this configuration, the current and voltage sense data types are frame interleaved on the assigned data output channel. The current and voltage sense data words are both 16-bits in length, and as a result if the channel length is longer than 16-bits the trailing padding bits are set to either Hi-Z or zero code depending on the state of the *[PCM\\_TX\\_EXTRA\\_HIZ](#page-120-1)*  bit field.

To identify the data type in channel-interleaved mode, the LSB of the 16-bit data word is dropped (truncated). The data word is then right shifted by a single bit, and the now vacant MSB is replaced with either a 0 to indicate voltage sense data or a 1 to indicate current sense data. For phase alignment, the voltage sense data for a single sampling instant is always transmitted in the assigned channel on the first frame, followed by the current sense data on the second frame. The MSB value and the transmission order allow the host to identify and phase-align the output data across frames.

Since the I/V sense data is frame interleaved, the sample rate for the PCM interface must be greater than that of the I/V sense ADCs by an integer ratio of 2. The example below shows a basic case where the sample rate of the PCM interface is twice that of the I/V sense ADCs.



*Figure 7. I/V Sense Path Data Interleaved on a Single Data Output Channel* 

## **Data Output Status Bits**

The following interrupt information is reported in the status slots:

- Bit 15: BPE level 0 begin
- Bit 14: BPE level change
- Bit 13: BPE active begin
- Bit 12: BPE active end
- Bit 11: Thermal warning 1 begin
- Bit 10: Thermal warning 1 end
- Bit 9: Thermal warning 2 begin
- Bit 8: Thermal warning 2 end
- Bit 7: Thermal foldback begin
- Bit 6: Thermal foldback end
- Bit 5: DHT active end
- Bit 4: DHT active begin
- Bit 3: Speaker overcurrent
- Bit 2: Power-up done
- Bit 1: 0
- Bit 0: 0

Each of the interrupt information above corresponds to a raw interrupt and is 1 bit wide. When a raw interrupt has a rising edge, the corresponding status bit goes high during the next LRCLK frame. The status bit goes low during the next LRCLK frame even if the raw interrupt has remained high.

### **PCM Interface Timing**

[Figure 8](#page-48-0) and [Figure 9](#page-48-1) shows timing for BCLK, LRCLK, DIN, and DOUT. See the Electrical Characteristics table for more details.

<span id="page-48-0"></span>

*Figure 8. PCM Interface Timing/Slave Mode—LRCLK, BCLK, DIN Timing Diagram* 

<span id="page-48-1"></span>

*Figure 9. PCM Interface Timing/DOUT Timing Diagram* 

### **Interrupts**

The device supports individually enabled status interrupts for sending feedback to the host about events that have occurred on-chip. When enabled, interrupts are transmitted on the IRQ output.

### **Interrupt Bit Field Composition**

Each interrupt source has five individual bit field components. The function of each component is detailed as follows, and the corresponding bit fields for each source can be identified by the appended suffix (shown in parentheses).

- **Raw Status (RAW):** Each interrupt source has a read-only bit to indicate the real-time raw status of the interrupt source.
- **State (STATE):** Each interrupt source has a read-only state bit that is set whenever a rising edge occurs on the associated raw status bit. The state bit is set regardless of the setting of the source enable bit.
- **Flag (FLAG):** Each interrupt source has a read-only flag bit. If the source enable bit is set, then the flag bit is set and an interrupt can be generated whenever the source state bit is set.
- **Enable (EN):** Each interrupt source has a dynamic read/write enable bit. When the enable bit is set, the associated flag bit is set and an interrupt can be generated whenever the source state bit is set.
- **Clear (CLR):** Each interrupt source has a dynamic write-only clear bit. Writing a 1 to a clear bit resets the associated state and flag bits to 0. Writing a 0 to a clear bit has no effect. In I<sup>2</sup>C control mode, the IRQ output is deasserted if all flag bits are 0.

### **Interrupt Output Configuration**

The device allows the user to configure the drive mode, drive strength, and polarity of the IRQ output. The *[IRQ\\_MODE](#page-111-0)*  bit controls the drive mode. If *[IRQ\\_MODE](#page-111-0)* is 0, the pin is configured as an open-drained output and requires an external pullup resistor. If *[IRQ\\_MODE](#page-111-0)* is 1, then IRQ is configured as a push-pull CMOS output.

Additionally, when IRQ is configured as a push-pull CMOS output, the drive strength control (*[IRQ\\_DRV](#page-119-4)*) bits set the drive strength of the IRQ output. Four different CMOS drive strengths are available.

The *[IRQ\\_POL](#page-111-1)* bit controls the polarity of the IRQ bus. Interrupt events (a flag bit is set high) assert the IRQ bus low if *[IRQ\\_POL](#page-111-1)* = 0 and high if *[IRQ\\_POL](#page-111-1)* = 1. The IRQ bus deasserts if all flag bits are cleared (set low).

#### **Interrupt Sources**

## **Table 7. Interrupt Sources**





## **Table 7. Interrupt Sources (continued)**

## **Speaker Path**

### **Speaker Path Block Diagram**



*Figure 10. Speaker Signal Path Diagram* 

### **Speaker Playback Path**

The source input data to the speaker amplifier path is routed from either the PCM interface or the tone generator. The data is then routed through digital filters, signal processing, and volume/gain control blocks before reaching the Class-DG speaker amplifier.

### **Speaker Path Noise Gate**

The speaker path noise gate function is enabled when the device is in the active state and the noise gate enable (*[NOISEGATE\\_EN](#page-115-0)*) is set to 1. The noise gate enable can be programmed dynamically. However, if the noise gate function is disabled (*[NOISEGATE\\_EN](#page-115-0)* is set to 0) while the noise gate is active (speaker path actively muted), the noise gate function remains active until after it deactivates normally (unmutes the speaker path).

When the noise gate is enabled, the noise gate activates whenever the amplitude of the input audio data to the speaker path (from the PCM interface) is below the configured mute threshold (*[MUTE\\_THRESH](#page-114-0)*) for more than 1024 consecutive data samples. When the noise gate is active, the amplifier path is muted, the current and voltage sense ADC paths output zero code data, and the device idles in a reduced power state.

The noise gate deactivates immediately if the amplitude of a single sample from the input audio data exceeds the configured unmute threshold (*[UNMUTE\\_THRESH](#page-114-1)*). When the noise gate deactivates, the speaker path is unmuted and returns to normal operation before the input audio data (that triggered deactivation) reaches the speaker output. Once noise gate deactivation is complete, the current and voltage sense ADC paths resume operation and output data normally.

The noise gate mute and unmute threshold settings are selected in terms of the number of bits (starting from the LSB) that must be toggling (or active) in order for the input signal amplitude to exceed the thresholds. It is invalid to set the noise gate unmute threshold (*[UNMUTE\\_THRESH](#page-114-1)*) such that it is less than the configured mute threshold (*[MUTE\\_THRESH](#page-114-0)*). The location of the audio data LSB within the PCM input data channel is determined by the configured PCM data word size ([PCM\\_CHANSZ](#page-119-1)). The supported combinations are shown in [Table 8](#page-52-0).

## <span id="page-52-0"></span>**Table 8. Noise Gate/Idle Mode Threshold LSB Location by Input Data Configuration**



It is not valid to enable the speaker path noise gate function when the tone generator is enabled or when the speaker idle mode is enabled.

#### **Speaker Path Dither**

The input data to the speaker path can optionally have dither (±1LSB peak-to-peak) applied if *[SPK\\_DITH\\_EN](#page-133-0)* is set to 1. No dither is applied when *[SPK\\_DITH\\_EN](#page-133-0)* is set to 0.

#### **Speaker Path Data Inversion**

The input data to the speaker path can optionally be inverted by setting the *[SPK\\_INVERT](#page-133-1)* bit to 1. The input data to the speaker path can only be inverted when the speaker DC blocking filter is also enabled by setting *[SPK\\_DCBLK\\_EN](#page-134-0)* to 1.

#### **Speaker Path DC Blocking Filter**

A DC blocking filter can be enabled on the speaker path by setting the *[SPK\\_DCBLK\\_EN](#page-134-0)* bit to 1.

### <span id="page-52-1"></span>**Speaker Path Digital Volume Control**

The device has a dynamically programmable speaker path digital volume control. The digital volume control provides an attenuation range of 0dB to -63dB in 0.5dB steps that is configured with the *[SPK\\_VOL](#page-132-0)* bit field. A digital mute is also provided, and is enabled when *[SPK\\_VOL](#page-132-0)* is set to 0x7F.

Digital volume ramping during speaker path start up, speaker path shutdown, and digital mute (*[SPK\\_VOL](#page-52-1)* = 0x7F) is disabled by default. However, both the volume ramp up and ramp down can be individually enabled with the *[SPK\\_VOL\\_RMPUP\\_BYPASS](#page-133-2)* and *[SPK\\_VOL\\_RMPDN\\_BYPASS](#page-133-3)* bit fields respectively. When volume ramp up or ramp down is enabled, the device turn-on and turn-off times are longer.

### <span id="page-52-2"></span>**Speaker Path Digital Gain Control**

The device provides a programmable speaker path digital gain control. The digital gain control provides a range of 0dB to +6dB in 0.5dB steps that is configured with the *[SPK\\_GAIN](#page-138-0)* bit field. Unlike the digital volume control, the digital gain setting cannot be dynamically changed.

#### <span id="page-52-3"></span>**Speaker Path DSP Data Feedback Path**

The speaker path DSP data can be routed from just before the DAC input back to the PCM interface, and can be assigned to any valid data output channel. The speaker path DSP data feedback path is enabled with the *[SPK\\_FB\\_EN](#page-139-0)* bit.

#### **Speaker Safe Mode**

The device provides a safe mode bit (*[SPK\\_SAFE\\_EN](#page-133-4)*) which applies a -18dB attenuation to the input signal when set to 1. By default, speaker safe mode is enabled to protect any speaker connected to the device on power-up. While speaker safe mode is enabled, the digital volume control (*[SPK\\_VOL](#page-132-0)*) and speaker digital gain control (*[SPK\\_GAIN](#page-52-2)*) settings are ignored.

#### **Speaker Audio Processing Bypass Path**

In applications where the audio processing in the main speaker path is not desired, the device provides a bypass path. The bypass path is selected with the *[PCM\\_BYPASS\\_EN](#page-128-1)* bit field. The PCM data input channel for the speaker audio processing bypass path is selected with the *[PCM\\_BYPASS\\_SOURCE](#page-127-1)* bit field.

### **Bypass Path Data Inversion**

The input data to the audio processing bypass path can optionally be inverted by setting the *[BYP\\_INVERT](#page-139-1)* bit to 1. The input data to the audio processing bypass path can only be inverted when the DC blocking filter is also enabled by setting *[SPK\\_DCBLK\\_EN](#page-134-0)* to 1.

### **Bypass Path Dither**

The input data to the audio processing bypass path can optionally have dither (±1LSB peak-to-peak) applied if *[SPK\\_DITH\\_EN](#page-133-0)* is set to 1. No dither is applied when *[SPK\\_DITH\\_EN](#page-133-0)* is set to 0. The *[SPK\\_DITH\\_EN](#page-133-0)* bit also controls the dither applied to the audio playback path.

### **Bypass Path DC Blocking Filter**

A DC blocking filter can be enabled on the audio processing bypass path by setting the *[SPK\\_DCBLK\\_EN](#page-134-0)* bit to 1. The *[SPK\\_DCBLK\\_EN](#page-134-0)* also controls the DC blocking filter in the audio playback path.

### <span id="page-53-0"></span>**Speaker Maximum Peak Output Voltage Scaling**

The device operates over a large PVDD supply voltage range, and as a result the full-scale speaker amplifier output amplitude level is configurable to allow it to be scaled. As a baseline, the full-scale output of the speaker path DAC is 3.68dBV (typ). The speaker path no-load maximum peak output voltage level ( $V_{MPO}$ ) is then programmable relative to this baseline level. The peak output scaling range is from +4dB to +21dB, and is set with the *SPK GAIN MAX* bit field.

The speaker output signal level (in dBV) for a given digital input signal level (in dBFS) is calculated as follows:

Output Signal Level (dBV) = Input Signal Level (dBFS) + 3.68 (dBV) + SPK\_GAIN\_MAX (dB)

(0dBFS is referenced to 0dBV)

The peak output voltage scaling is applied to the signal path using a combination of digital gain and analog gain adjustments.

## **Dynamic-Headroom Tracking (DHT)**

The device features dynamic-headroom tracking that can preserve consistent signal distortion and listening levels in the presence of a varying supply level. The DHT block provides both a dynamic range compressor (DRC) and limiter. The limiter can operate either as a signal distortion limiter (SDL) or a standard signal level limiter (SLL). Each of these three functions can be used independently (modes 1 through 3), and the SLL and DRC can be used simultaneously (mode 4). The DHT block is enabled with the *[DHT\\_EN](#page-150-0)* bit. Prior to enabling the DHT, the measurement ADC, PVDD, and VBAT channels should be configured and enabled as required based on the amplifier mode of operation. The DHT block uses the measured supply levels and the current signal level to calculate the attenuation (if any) that is applied to the signal path. Also, the DHT block should not be disabled by setting *[DHT\\_EN](#page-150-0)* bit to 0 when the DHT is active (i.e., attenuation is being applied).



*Figure 11. Simplified Dynamic-Headroom Tracking System Block Diagram* 

### **DHT Supply Tracking and Headroom**

The DHT block uses three parameters to track the target peak output level  $(V<sub>TPO</sub>)$  relative to the maximum peak output voltage ( $V_{MPO}$ ) as the active speaker amplifier supply level varies.

The first is the speaker amplifier full-scale gain setting (*[SPK\\_GAIN\\_MAX](#page-53-0)* bit field). This control selects the maximum (noload) peak output voltage level ( $V_{\text{MPO}}$ ) that is output by the Class-DG amplifier with a full-scale input signal (0dBFS). Most DHT thresholds and parameters are calculated relative to the full-scale  $V_{\text{MPO}}$ .

The second parameter is the measured speaker amplifier supply voltage level  $(V_{\text{SIIP}})$ . The measurement ADC provides the DHT block with the current supply voltage levels ( $V_{PVDD}$  and  $V_{VBAT}$ ). The DHT block decides which supply voltage to use for calculations based on the currently active speaker amplifier supply.

The third parameter is the speaker amplifier supply headroom (SUP $_{HR}$ ). The supply headroom is a positive or negative percentage offset relative to the measured V<sub>SUP</sub> conversion result. It is configured using the *[DHT\\_HR](#page-147-0)* bit field, and can be set from  $+20\%$  to -20% of  $V_{\text{SUP}}$  in 2.5% step sizes.

The DHT target peak output voltage level ( $V_{\text{TPO}}$ ) is equal to the measured supply voltage ( $V_{\text{SIIP}}$ ) scaled to include the selected supply headroom percentage and is actively calculated with the following equation:

 $V_{TPO}$  =  $V_{SUPO}$  x (100% – SUP<sub>HR</sub>) The target peak output attenuation (or ratio) from  $V_{TPO}$  to  $V_{MPO}$  is calculated as follows:

 $A_{TPO}$  = 20 x log ( $V_{TPO}/V_{MPO}$ )

If  $A_{TPO}$  exceeds 0dB ( $V_{SUP}$  with headroom >  $V_{MPO}$ ), then the DHT block assumes that there is sufficient supply voltage to reproduce the audio signals as configured without attenuation. In this case,  $A_{TPO} = 0$ dB is used for all further calculations. This is important as the DHT functions only ever apply attenuation and do not apply positive gain. Once the calculated V<sub>TPO</sub> drops below V<sub>MPO</sub>, the calculated target peak output attenuation (A<sub>TPO</sub>) is less than 0dB, and the DHT functions are applied appropriately as the input signal level changes.

For example, if V<sub>MPO</sub> = 13.63V, V<sub>SUP</sub> = 8.04V, and SUP<sub>HR</sub> = -20%, then solving for V<sub>TPO</sub> yields a target peak output level of approximately 9.65V. Next, solving for the target peak output attenuation  $(A_{TPO})$  yields approximately -3dB.

[Figure 12](#page-55-0) shows the default transfer function (with no DHT attenuation applied), where the current target peak output level (V<sub>TPO</sub>) is based on the current V<sub>SUP</sub> and the supply headroom settings. The tracked V<sub>TPO</sub> and the resulting peak output attenuation ( $A_{TPO}$ ) are then used in the attenuation calculations for the DHT functions. Note that this and all subsequent figures are not drawn to precise scale, and that the x-axis is input signal level (dBFS) on a linear scale, while the y-axis is peak output voltage level on a log scale.

<span id="page-55-0"></span>

*Figure 12. VTPO and ATPO Calculation Example* 

### **DHT Mode 1 – Signal Distortion Limiter**

The DHT signal distortion limiter (SDL) maintains a consistent level of signal distortion at the amplifier output as the supply voltage (V<sub>SUP</sub>) changes. To use DHT mode 1 (just the signal distortion limiter active), set the *[DHT\\_LIM\\_MODE](#page-148-0)* bit low (default) to place the limiter function in supply tracking mode (SDL), and set the dynamic range compressor rotation point (*[DHT\\_VROT\\_PNT](#page-147-1)*) to 0dBFS (effectively disabling the DRC). The signal distortion limiter function is a compressor with a ratio of infinity to one that actively sets its threshold  $(V_{SDL}$  in voltage) equal to the calculated target peak output voltage level (V<sub>TPO</sub>). The output referred SDL threshold (SDL<sub>THR</sub>) and the input referred SDL knee or rotation point (SDL<sub>RP</sub>) are equal in mode 1, and can be calculated relative to full-scale (in dBFS) as a ratio of V<sub>TPO</sub> to V<sub>MPO</sub>:

 $SDL_{RP}$  =  $SDL_{THR}$  = 20 x log( $A_{TPO}$ ) = 20 x log( $V_{TPO}$  /  $V_{MPO}$ )

The transfer function for input signal levels below the SDL rotation point (SDL<sub>RP</sub>) is unchanged. When the input signal level exceeds SDL<sub>RP</sub>, the signal distortion limiter function is applied to the signal path. As the input signal level increases, the distortion limiter attenuation continues to increase as well and can be calculated for a given input signal level  $(A_{\text{INPI TT}})$ in dBFS) as follows:

### SDL ATTENUATION = SDLRP – AINPUT

By actively recalculating SDL<sub>RP</sub> (or SDL<sub>THR</sub>) as the target peak output level ( $V<sub>TPO</sub>$ ) changes, the DHT SDL maintains a consistent limit and level of amplifier output distortion relative to available supply voltage ( $V_{\text{SUP}}$ ).

When the target peak output voltage (V<sub>TPO</sub>) exceeds the amplifier maximum peak output voltage (V<sub>MPO</sub>) there is sufficient headroom and no SDL attenuation is applied. However, as soon as  $V<sub>TPO</sub>$  falls below  $V<sub>MPO</sub>$ , it is possible for the input signal amplitude to exceed the calculated SDL<sub>RP</sub>. The following examples show the transfer function when  $V_{\text{SUP}}$ ≥ V<sub>MPO</sub> with the minimum (-20%), without (0%), and maximum (+20%) supply headroom (SUP<sub>HR</sub>) settings. Note that in the case with positive headroom (+20%), the SDL<sub>RP</sub> falls below the input signal full-scale level even though V<sub>SUP</sub> =  $V_{\text{MPO}}$ .



*Figure 13. Signal Distortion Limiter with VMPO ≤ VSUP and +20% Headroom (SUPHR)* 



*Figure 14. Signal Distortion Limiter with V<sub>MPO</sub> ≤ V<sub>SUP</sub> and 0% Headroom (SUP<sub>HR</sub>)* 



*Figure 15. Signal Distortion Limiter with VMPO ≤ VSUP and -20% Headroom (SUPHR)* 

As the supply voltage (V<sub>SUP</sub>) drops further below the maximum peak output voltage (V<sub>MPO</sub>), the DHT target peak out voltage (V<sub>TPO</sub>) proportionally scales down. In cases with zero or positive amplifier supply headroom settings (+20% ≥ SUP<sub>HR</sub> ≥ 0%), the input signal level can exceed the SDL rotation point (SDL<sub>RP</sub>) before the peak output exceeds V<sub>SUP</sub>. In this case, amplifier output clipping can be prevented.



*Figure 16. Signal Distortion Limiter with VMPO > VSUP and +20% Headroom (SUPHR)* 



*Figure 17. Signal Distortion Limiter with VMPO > VSUP and 0% Headroom (SUPHR)* 

In cases with a negative supply headroom setting (0% > SUP<sub>HR</sub> ≥ -20%), the input signal does not exceed the SDL<sub>RP</sub> until after the peak output reaches V<sub>SUP</sub>. As a result, clipping occurs at the amplifier output. However, once the input signal level exceeds the SDL<sub>RP</sub>, the audio signal level is digitally limited by the SDL preventing the amplifier output clipping from worsening further.



*Figure 18. Signal Distortion Limiter with V<sub>MPO</sub> > V<sub>SUP</sub> and -20% Headroom (SUPHR)* 

## **DHT Mode 2 – Signal Level Limiter**

In DHT mode 2, the limiter is configured as a fixed threshold signal level limiter (SLL). Set the *[DHT\\_LIM\\_MODE](#page-148-0)* bit high to place the limiter function in SLL mode, and set the dynamic range compressor rotation point to 0dBFS (effectively disabling the DRC).

Like the signal distortion limiter, the signal level limiter function is a compressor with a ratio of infinity to 1. However, unlike the SDL, the SLL output referred threshold (SLL<sub>THR</sub>) is configured to a set level. The SLL<sub>THR</sub> is selected with the *[DHT\\_LIM\\_THRESH](#page-148-1)* bit field from a range of 0dBFS to -15dBFS. The SLL threshold can also expressed as an input referred knee or rotation point (SLL<sub>RP</sub>) which is equal to SLL<sub>THR</sub> in mode 2. The SLL amplifier peak output voltage limit (V<sub>SLL</sub>) is calculated from the selected SLL threshold (SLL<sub>THR</sub>) and maximum peak output voltage (V<sub>MPO</sub>) with the following equation:

SLL PEAK OUTPUT VOLTAGE LIMIT = V<sub>SLL</sub> = V<sub>MPO</sub> x 10<sup>(SLL</sup>THR <sup>/ 20)</sup>

The transfer function for signal levels below the SLL threshold ( $SLL<sub>THR</sub>$ ) is unchanged. When the signal level exceeds the  $SLL<sub>THR</sub>$ , the signal level limiter function is applied to the signal path. As the input signal level increases, the limiter attenuation continues to increase as well and can be calculated for a given input signal level  $(A_{\text{INPUT}})$  in dBFS) relative to  $SLL$ <sub>RP</sub> (=  $SLL$ <sub>THR</sub>) as follows:

SLL ATTENUATION = SLLRP - AINPUT

When  $V<sub>TPO</sub>$  is greater than  $V<sub>SLL</sub>$ , the amplifier peak output level is limited to  $V<sub>SLL</sub>$  whenever the signal amplitude exceeds the SLL threshold (SLL $THR$ ). As a result of the fixed SLL threshold and rotation point, the transfer function is identical for any  $V_{\text{SUP}}$  level and corresponding  $V_{\text{TD}}$  that is greater than  $V_{\text{SI}}$ .

This is illustrated in [Figure 19](#page-60-0) for decreasing V<sub>SUP</sub> and V<sub>TPO</sub> levels. As V<sub>SUP</sub> decreases, V<sub>TPO</sub> is recalculated and decreases as well. Three different progressively lower V<sub>TPO</sub> levels are shown (V<sub>TPO1</sub>, V<sub>TPO2</sub>, and V<sub>TPO3</sub>). Due to the fixed SLL threshold,  $V_{SLL}$  is the same in all three cases. Since all three  $V_{TPO}$  values are greater than  $V_{SLL}$ , the transfer function for each case is identical and is limited at  $V_{SLL}$ .

<span id="page-60-0"></span>

*Figure 19. Signal Level Limiter with V<sub>TPO</sub> > V<sub>SLL</sub> as V<sub>SUP</sub> Decreases* 

When V<sub>TPO</sub> is less than V<sub>SLL</sub>, the amplifier output can clip before the input signal amplitude exceeds the SLL rotation point (SLL<sub>RP</sub> = SLL<sub>THR</sub>). As the input signal level continues to increase and exceed SLL<sub>RP</sub>, the signal level is digitally limited which prevents the amplifier output clipping from worsening further. Because both the SLL threshold and rotation point are fixed relative to full-scale, the clipping at the amplifier output grows progressively worse prior to the input signal exceeding SLL<sub>RP</sub> (= SLL<sub>THR</sub>) as  $V_{\text{SUP}}$  continues to decrease.

The following [[Signal Level Limiter with V<sub>TPO</sub>< V<sub>SLL</sub> Showing Amplifier Output Clipping]] has the same SLL settings as [Figure 19](#page-60-0) (same SLL<sub>THR</sub>). For simplicity, V<sub>TPO</sub> = V<sub>SUP</sub> (SUP<sub>HR</sub> = 0%), and V<sub>TPO</sub> has decreased further and is now less than V<sub>SLL</sub>. As a result, the amplifier output clips before the SLL digitally limits the signal level.



*Figure 20. Signal Level Limiter with VTPO < VSLL Showing Amplifier Output Clipping* 

## **DHT Mode 3: Dynamic Range Compressor**

The DHT dynamic range compressor (DRC) is configured by setting the input referred rotation point (DRC<sub>RP</sub> in dBFS). The DRC<sub>RP</sub> can be selected from a range from 0dBFS to -15dBFS with the *[DHT\\_VROT\\_PNT](#page-147-1)* bit field. To calculate the DRC output referred voltage threshold ( $V_{\text{DRC}}$ ), use the following equation:

$$
V_{DRC} = V_{MPO} \times 10^{(DRC_{RP}/20)}
$$

For mode 3 operation, set the DRC rotation point (DRC<sub>RP</sub>) to any level lower than 0dBFS. Next, to disable limiter functions, place DHT into signal level limiter mode (*[DHT\\_LIM\\_MODE](#page-148-0)* = 1) and set the fixed SLL threshold (SLL<sub>THR</sub>) to 0dBFS (using the *[DHT\\_LIM\\_THRESH](#page-148-1)* bit field).

Once configured, the dynamic range compressor rotation point (DRC<sub>RP</sub>) is fixed at the selected level (or ratio) relative to the input full-scale. As  $V_{\rm SID}$  and  $V_{\rm TPO}$  change, the DRC compression ratio for input signals that exceed DRC<sub>RP</sub> changes as well. However, the transfer function remains unchanged for input signals below  $DRC_{RP}$ . If the amplifier is operating in class-G mode, it is recommended that the DRC rotation point (DRC<sub>RP</sub>) be set such that V<sub>DRC</sub> exceeds the maximum possible VBAT voltage level. This ensures that DRC compression is only applied when PVDD is the active amplifier supply, and that the DHT cannot rapidly switch between two different ratios if the active amplifier supply toggles quickly.

The DHT tracks the target peak output voltage ( $V<sub>TPO</sub>$ ) and attenuation ( $A<sub>TPO</sub>$ ). As they change, the adaptive DRC compression ratio smoothly scales the listening level of the amplifier for any input signals that exceed DRCRP. The DRC compression ratio is actively calculated with the following formula:

DRC COMPRESSION RATIO = DRCRP/(ATPO – DRCRP)

The DRC attenuation for a given input signal level  $(A_{\text{INPUT}}$  in dBFS) is calculated as follows:

DRC ATTENUATION = ATPO – AINPUT x (ATPO / DRCRP)

The following example shows the DRC transfer function with SUP<sub>HR</sub>  $\geq$  0% as V<sub>SUP</sub> (and thus V<sub>TPO</sub>) decreases. As the  $V<sub>TPO</sub>$  level decreases (from  $V<sub>TPO1</sub>$  to  $V<sub>TPO2</sub>$  to  $V<sub>TPO3</sub>$ ), the DRC compression ratio increases.



*Figure 21. Dynamic Range Compression with Decreasing VSUP and SUPHR ≥ 0%* 

[Figure 22](#page-63-0) shows the DRC transfer function with SUP<sub>HR</sub> < 0%. Due to the negative supply headroom, V<sub>TPO</sub> is greater than  $V_{\text{SUP}}$  and the amplifier output clips before the input signal reaches full-scale.

<span id="page-63-0"></span>

*Figure 22. Dynamic Range Compressor with SUPHR < 0% and Output Clipping* 

### **DHT Mode 4: Dynamic Range Compressor with Signal Level Limiter**

In DHT mode 4, the dynamic range compressor (DRC) and signal level limiter (SLL) are both enabled. The DRC rotation point (DRC<sub>RP</sub>) must be less than 0dBFS to enable mode 4. In addition, the DHT limiter function must be configured for SLL mode (*[DHT\\_LIM\\_MODE](#page-148-0)* = 1) with an SLL threshold (SLL<sub>THR</sub>) less than 0dBFS. Finally, to create a DHT response curve with both DRC and SLL inflection points, the SLL threshold ( $V_{SLL}$ ) must be greater than the DRC voltage threshold ( $V_{\text{DRC}}$ ). This insures that the resulting SLL<sub>RP</sub> is always greater than the DRC<sub>RP</sub>; otherwise, the SLL limits the signal level before the DRC rotation point is ever reached.

[Figure 23](#page-64-0) shows three mode 4 transfer functions for three progressively lower  $V_{\text{SIIP}}$  levels. The supply headroom is configured for SUP<sub>HR</sub> > 0% (positive supply headroom), and the calculated V<sub>TPO</sub> value is falling such that V<sub>TPO1</sub> >  $V<sub>TPO2</sub> > V<sub>TPO3</sub>$ . The DRC rotation point and SLL threshold are constant in all three cases, and SLL<sub>THR</sub> is selected such that as  $V<sub>TPO</sub>$  falls the SLL knee (SLL<sub>RP</sub>) is greater than the DRC<sub>RP.</sub>

In the first two cases (for  $V_{TPO1}$  and  $V_{TPO2}$ ), the calculated SLL output voltage limit ( $V_{SLL}$ ) is less than  $V_{TPO}$ . As the signal level increases, it is first compressed by the DRC function then limited once the output level reaches  $V_{SLL}$ . In the third case, the SLL function is never applied since  $V_{SLL}$  is greater than  $V_{TPO3}$  and the signal level, while still compressed by the DRC, reaches full-scale before exceeding  $V_{SLL}$ .

<span id="page-64-0"></span>

*Figure 23. DHT DRC and SLL with Decreasing VSUP (VTPO), and SUPHR ≥ 0%* 

[Figure 24](#page-65-0) shows a mode 4 transfer function where the supply headroom is negative (SUP<sub>HR</sub> < 0%). As before, the SLL threshold (SLL<sub>THR</sub>) is programmed so that the resulting SLL<sub>RP</sub> is greater than the DRC<sub>RP</sub>. This also insures that the resulting  $V_{SLL}$  is greater than  $V_{DRC}$  and less than  $V_{TPO}$ . As the audio signal level increases, it is first compressed by the DRC function, then limited once the digital output signal level reaches V<sub>SLL</sub>. However, due to the negative headroom, the amplifier output clips before the SLL function digitally limits the signal level.

<span id="page-65-0"></span>

*Figure 24. DHT DRC and SLL with Decreasing VSUP (VTPO), and SUPHR < 0%* 

### **DHT Attenuation**

An interrupt is generated (*[DHT\\_ACTIVE\\_BGN\\_\\*](#page-96-7)*) when the DHT block first applies attenuation. When the DHT block fully releases all applied attenuation (i.e., DHT is inactive), an interrupt is generated (*[DHT\\_ACTIVE\\_END\\_\\*](#page-96-8)*). Interrupts are not generated when DHT is actively adjusting the level of attenuation.

The maximum attenuation (A<sub>MAX</sub>) applied to the audio signal by the DHT functions is selected with the *DHT\_MAX\_ATN* bit field. The maximum attenuation can be set from -1dB to -15dB with a 1dB step size. The configured DHT functions stop further attenuation of the audio signal once the calculated attenuation (relative to the un-attenuated input signal level) reaches the selected maximum attenuation (A<sub>MAX</sub>). If the calculated attenuation (based on input signal level and measured V<sub>SUP</sub>) exceeds the selected maximum attenuation (A<sub>MAX</sub>), the applied attenuation is set equal to (limited at)  $A_{MAX}$ . This can occur anytime when the target peak output ( $V_{TPO}$ ) to maximum peak output ( $V_{MPO}$ ) ratio or peak output attenuation (denoted  $A_{TPO}$ ) is less than (or has a larger absolute value than)  $A_{MAX}$ .

All previous examples show cases where the peak output attenuation  $(A_{TPO})$  did not exceed the selected maximum attenuation (A<sub>MAX</sub>). The following figures show signal distortion limiter use cases where V<sub>SUP</sub> has decreased until A<sub>TPO</sub>  $<$  A<sub>MAX</sub> (the DHT DRC function DRC<sub>RP</sub> is set to 0dbFS as in use case 1).

In [Figure 25](#page-66-0), the SUP<sub>HR</sub> is set to -20%. Since A<sub>TPO</sub> < A<sub>MAX</sub>, the attenuation applied by the distortion limiter reaches the programmed maximum attenuation level before the input signal reaches full-scale. For input signals past the point where calculated attenuation is equal to  $A_{MAX}$ , the attenuation stops increasing and is now fixed at  $A_{MAX}$ . As a result, the audio signal (in the digital domain) begins to increase past this point. This results in the distortion increasing at the amplifier output (which was already clipping at the limited level of distortion).

<span id="page-66-0"></span>

*Figure 25. Distortion Limiter Case with -20% Headroom and AMAX Exceeded* 

In Figure  $26$ , the supply headroom is set to  $+20%$ . As before, the attenuation applied by the SDL reaches the selected maximum attenuation (A<sub>MAX</sub>) before the input signal reaches full-scale. The audio signal (in the digital domain) begins increasing past this point, and the signal level (and any distortion) at the amplifier output increases as well. In this case, the amplifier output was not clipping until after  $A_{MAX}$  was exceeded.

<span id="page-67-0"></span>

*Figure 26. Distortion Limiter Case with +20% Headroom and AMAX Exceeded* 

## **DHT Attenuation Reporting**

In TDM mode, the current level of DHT attenuation is reported on the PCM data output (DOUT) when the DHT attenuation transmit enable bit is set high (*[PCM\\_DHT\\_ATN\\_EN](#page-127-2)* = 1). The DHT attenuation level output is transmitted as a 14-bit unsigned binary attenuation level:

### DOUT CURRENT DHT ATTENUATION (dB) = 20 x log(14-bit DOUT Value/16383)

If enabled, the DHT attenuation target (in dB) is also shared between devices on the interchip communication (ICC) bus. In this case, the DHT attenuation level output requires two ICC output slots (8 bits each) and is transmitted as a 10-bit unsigned binary attenuation level (DHT\_ATN) followed by 6 bits of zero padding.

The current DHT attenuation (in dB) is calculated from the 10-bit value (DHT\_ATN) with the following equation:

$$
ICC CURRENT DHT ATTENUATION (dB) = -(DHT_ATN[9:0] \times 0.015625dB)
$$

The current DHT attenuation level cannot exceed the selected DHT maximum attenuation (A<sub>MAX</sub>). Additionally, when the DHT is inactive, the reported attenuation is 0x0.

### **DHT Ballistics**

When the signal level exceeds the rotation point or threshold for a configured DHT function (SDL, SLL, and/or DRC) or continues to increase beyond this point, the appropriate level of attenuation is applied to the signal level at the programmed attack rate. The DHT attack rate is selected with the *[DHT\\_ATK\\_RATE](#page-149-0)* bit field.

The change in input signal level is detected by a peak detect circuit which has a fixed 3.5ms release time. When the signal level decreases or drops below the rotation point or threshold for a configured DHT function (SDL, SLL, and/or DRC), the appropriate level of applied attenuation is released. The DHT release rate is selected with the *[DHT\\_RLS\\_RATE](#page-149-1)* bit field. However, due to the 3.5ms/dB peak detector, the 2ms/dB release rate is effectively 3.5ms/dB. All other release rates have a fixed delta of 3.5ms compared to the programmed release rate.

The attack and release behavior is slightly different when triggered by a change in active amplifier supply level. When the supply level decreases and triggers a DHT function attack, the attenuation is applied quickly at the configured attack rate. Likewise, as the supply level increases, the attenuation is released at the configured release rate. However, if DHT supply hysteresis is enabled (*[DHT\\_SUPPLY\\_HYST\\_EN](#page-150-1)* = 1), then as the supply increases the applied DHT function does not release attenuation until the increase in the supply level exceeds the programmed DHT supply hysteresis level (*[DHT\\_SUPPLY\\_HYST](#page-150-2)*). Once the supply increase exceeds the hysteresis, the appropriate level of applied attenuation is released at the configured release rate.

## **Speaker Amplifier**

The device features a Class-DG speaker amplifier output stage. The speaker amplifier playback path is enabled and disabled using the *[SPK\\_EN](#page-139-2)* bit. The Class-DG multilevel amplifier generates a rail-to-rail output, pulse-width modulated (PWM) signal. By varying the PWM duty cycle, the amplifier modulates the output with the audio input signal. Because the switching frequency of the amplifier is 472kHz (typ) when the output signal is filtered by the speaker, only the audio component remains. Rail-to-rail operation ensures that power dissipation at the output is dominated by the on resistance  $(R<sub>ON</sub>)$  of the power output MOSFETs brief saturation current draw as the output switches as well as the current draw necessary to charge the output stage gates.

### **Speaker Amplifier Operating Modes**

The speaker amplifier can operate both in Class-DG and standard Class-D modes. In Class-DG mode, the amplifier output supply rail is switched between  $V_{PVDD}$  and  $V_{VBAT}$  based on the signal level. If Class-DG operation is disabled, the amplifier operates as a fixed supply Class-D amplifier and can be configured to use either  $V_{PVDD}$  or  $V_{VBAT}$  as the output supply rail. The speaker amplifier operating mode is selected with the *[SPK\\_MODE](#page-136-0)* bit field.

#### **Class-DG Mode Enabled**

Class-DG is the default speaker amplifier mode of operation (*[SPK\\_MODE](#page-136-0)* = 0x0). In this mode, the amplifier switches the supply rail between PVDD and VBAT as needed to efficiently supply the required output power.

Additionally, if V<sub>VBAT</sub> drops below the programmed threshold level *[VBATLOW\\_OK\\_LVL](#page-136-1)* bit field, the amplifier operates from PVDD supply rail regardless of signal level.

The Class-DG signal level threshold (VDG THR) at which the amplifier switches between the supply rails is programmable. The method used to program the signal level threshold is selected with the *[SPK\\_DG\\_SEL](#page-135-0)* bit field. When *[SPK\\_DG\\_SEL](#page-135-0)* is set to 0x0, the threshold (VDG\_THR) is set to a fixed peak voltage level with the *[SPK\\_DG\\_THRES](#page-134-1)*  bit field. When *SPK DG SEL* is set to 0x1 (default), the threshold (VDG THR) is variable relative to the current VBAT voltage (measurement ADC result). The peak voltage headroom relative to  $V_{VBAT}$  is configured with the *[SPK\\_DG\\_HEADROOM](#page-135-1)* bit field. Finally, if *[SPK\\_DG\\_SEL](#page-135-0)* is set to 0x2, the threshold (VDG\_THR) is set based on whichever setting (*[SPK\\_DG\\_THRES](#page-134-1)* or *[SPK\\_DG\\_HEADROOM](#page-135-1)*) results in the lowest threshold for the current V<sub>VBAT</sub> voltage level. As a result, as  $V_{VBAT}$  decreases, VDG\_THR might transition from a fixed threshold to a lower  $V_{VBAT}$ headroom based variable threshold.

The Class-DG mode hold time is configured with the *[SPK\\_DG\\_HOLD\\_TIME](#page-135-2)* bit field. VBAT is selected as the active amplifier supply to save power when the signal level drops below the threshold for longer than hold time. The amplifier switches to the VBAT supply only at signal zero cross, so the measured hold time is the register configured hold time plus the time taken for a zero cross event to occur. When VBAT is the active amplifier output supply, the LV\_EN output asserts high. When the signal level rises above the threshold, the amplifier supply quickly switches to PVDD to provide higher output voltage swing and to avoid clipping. When PVDD is the active amplifier output supply, the LV\_EN output asserts low.

### **Delay for DG Mode**

When the amplifier is operating in the automatic Class-DG mode, to avoid the potential for clipping the output signal as the PVDD supply rises, there is a programmable delay in the signal path controlled by *[SPK\\_DG\\_DELAY](#page-136-2)*. This allows the PVDD supply time to increase the output voltage before it is required to output larger signals.

### **Class-DG Mode Disabled**

When Class-DG mode is disabled, the speaker amplifier operates in standard Class-D mode. In this case, the active amplifier output supply is configured to either PVDD (*[SPK\\_MODE](#page-136-0)* = 0x1) or VBAT (*[SPK\\_MODE](#page-136-0)* = 0x2).

If the active amplifier output supply is configured to VBAT (*SPK MODE* = 0x2), the amplifier operates from VBAT regardless of signal level. In this mode, the LV\_EN pin is asserted high.

When the active amplifier supply is set to PVDD, the amplifier always operates from PVDD regardless of the signal and supply levels. Furthermore, PVDD can be actively regulated between any levels within its standard operating range (3V to 20V) to save power. In this mode, LV\_EN pin is always asserted low.

#### **NOVBAT Mode**

By default, the PVDD and VBAT pins must be supplied with external supplies for the amplifier to operate. In systems where the VBAT pin cannot be supplied by an external supply, the device can still operate from PVDD supply only by programming the *[NOVBAT](#page-139-3)* bit field to 1. Additionally, the VBAT pin must be shorted to the VREFC pin on the PCB. When *[NOVBAT](#page-139-3)* bit field is set to 1, the amplifier automatically operates in PVDD mode and the *[SPK\\_MODE](#page-136-0)* bits have no effect.

#### **IDLE Mode**

In customer systems where an external LC filter is used for EMI reduction, the quiescent power consumption and power consumption in Class-DG mode for low signal levels can be reduced by setting the IDLE\_MODE\_EN bit field to 1. The idle mode setting is not intended for operation in PVDD mode (with VBAT pin supplied externally) or NOVBAT mode (when the VBAT pin is not supplied externally) and therefore should not be enabled.

When the idle mode is enabled, the idle mode activates whenever the amplitude of the input audio data to the speaker path (from the PCM interface) is below the configured mute threshold (*[MUTE\\_THRESH](#page-114-0)*) for more than 1024 consecutive data samples. When the idle mode is active, the current and voltage sense ADC paths output zero code data. The idle mode deactivates immediately if the amplitude of a single sample from the input audio data exceeds the configured unmute threshold (*[UNMUTE\\_THRESH](#page-114-1)*). When the idle mode deactivates, the current and voltage sense ADC paths resume operation and output data normally.

The idle mode mute and unmute threshold settings are selected in terms of the number of bits (starting from the LSB) that must be toggling (or active) in order for the input signal amplitude to exceed the thresholds. It is invalid to set the noise gate unmute threshold (*[MUTE\\_THRESH](#page-114-1)*) such that it is less than the configured mute threshold (*[MUTE\\_THRESH](#page-114-0)*). The location of the audio data LSB within the PCM input data channel is determined by the configured PCM data word size (*[PCM\\_CHANSZ](#page-119-1)*). The supported combinations are shown in [[Noise Gate/Idle Mode Threshold LSB Location By Input Data Configuration]]. It is not valid to enable the idle mode and noise gate function simultaneously.

#### **Speaker Amplifier Ultra-Low EMI Filterless Operation**

Traditional Class-D amplifiers require the use of external LC filters or shielding to meet electromagnetic-interference (EMI) regulation standards. However, the device features emissions limiting circuitry that limits the output switching harmonics that can directly contribute to EMI and radiated emissions.

The programmable speaker amplifier edge rate control bits are used to adjust the switching edge rate to help tune EMI performance. As the edge rate increases, the efficiency improves slightly, while as the edge rate is decreased, the efficiency drops slightly. The speaker amplifier edge rate is configured with the *[SPK\\_SL\\_RATE\\_GMODE](#page-137-0)*, *[SPK\\_SL\\_RATE\\_LS](#page-138-1)*, and *[SPK\\_SL\\_RATE\\_HS](#page-138-2)* bit fields.

The speaker amplifier output also supports spread-spectrum modulation (SSM). SSM is enabled by default to optimize the suppression and control of the output switching harmonics that can contribute to EMI and radiated emissions. The modulation index in spread-spectrum mode is controlled by the *[SPK\\_SSM\\_MOD\\_INDEX](#page-134-2)* bit field, and the maximum modulation index (MMI) varies accordingly. Higher percentage settings of the modulation index result in the switching frequency of the amplifier being modulated by a wider range, spreading out-of-band energy across a wider bandwidth. Above 10MHz, the wideband spectrum looks like noise for EMI purposes.

### **Speaker Amplifier Overcurrent Protection**

The device features amplifier current limit protection that protects the amplifier output from both high current and short circuit events. If the *[OVC\\_AUTORESTART\\_EN](#page-167-0)* bit is set to 1 and the speaker amplifier output current exceeds the current limit threshold (6.2A min), the device generates an interrupt and disables the amplifier output. After approximately 20ms, the amplifier output is re-enabled. If the overcurrent condition still exists, the device continues to disable and reenable the amplifier output automatically until the fault condition is removed.

If the *[OVC\\_AUTORESTART\\_EN](#page-167-0)* bit is set to 0, the device still generates an interrupt and disables the amplifier output when a speaker amplifier overcurrent event occurs. However, in this case the device is placed into software shutdown and the software enable (*[EN](#page-168-0)*) bit is set to 0. As a result, the host must manually re-enable the device after an overcurrent event.

## **Speaker Current and Voltage Sense ADC Path**

The device provides two separate 16-bit ADCs to monitor the speaker amplifier output current and voltage (the I/V sense ADC path). The current and voltage ADC paths are independently enabled with the *[IVADC\\_I\\_EN](#page-151-0)* and *[IVADC\\_V\\_EN](#page-151-1)* bits, respectively.

The voltage and current digital data output are routed to the host through the PCM interface data output (DOUT), which is enabled by the PCM\_TX\_EN bit field. Both I and V data can be formatted in 16-/24-/32-bit 2's complement format and has a voltage sense range of ±22V, and a current sense range of ±6.5A.

When configured in 16-bit mode for Vsense, if MSB is 0 then the voltage would be 11/(2^15) x DIG CODE.

For example, a Vsense reading of 0111 1111 1111 1111 translates to a voltage of +22V and a reading of 1000 0000 0000 0000 would be -22V.

For 16-bit Isense, if MSB is 0, then the current would be 3/(2^15) x DIG CODE.

For example, an Isense reading of 0111 1111 1111 1111 translates to a voltage of +6.5A and a reading of 1000 0000 0000 0000 would be -6.5A.

See the *[PCM Interface](#page-40-1)* section for details on configuring I/V sense-data output on DOUT. Both the current and voltage sense ADC output data can optionally have dither applied (±1 LSB peak-to-peak) by setting the *[IVADC\\_DITH\\_EN](#page-150-3)* bit field to 1. No dither is applied when *[IVADC\\_DITH\\_EN](#page-150-3)* is set to 0.

The I/V sense ADC path provides separate optional DC blocking filters (first-order highpass) in the current and voltage sense paths. The current and voltage path filters are enabled by setting the *IVADC I DCBLK EN* and *[IVADC\\_V\\_DCBLK\\_EN](#page-150-5)* bit fields to 1, respectively.

To ensure phase alignment, the current and voltage sense ADCs should be enabled either with a single write to the *[IVADC\\_I\\_EN](#page-151-0)* and *[IVADC\\_V\\_EN](#page-151-1)* bits (*[EN](#page-168-0)* = 1) or by setting both bits high before exiting software shutdown. Additionally, all feedback channels (VMON, IMON, and *[DSP feedback data](#page-52-3)*) are nominally in phase except for LRCLK sample rates of 88.2kHz/176.4kHz. For 88.2kHz/176.4kHz sample rates, VMON/IMON data is 180 degrees out of phase with the *[DSP](#page-52-3)  [feedback data.](#page-52-3)* 

When laying out the PCB, the OUTPSNS and OUTNSNS pins should be Kelvin connected as close as possible to the load connected between OUTP and OUTN for accurate voltage measurements. If a filter comprised of a ferrite bead and capacitor is installed between the speaker amplifier output pins and the load, then the sense lines should be connected between the filter and the load and as close to the load as possible. If an LC filter is installed between the amplifier output pins and the load, the OUTPSNS and OUTNSNS lines should be connected to the OUTP and OUTN lines before the filter. The OUTPSNS and OUTNSNS pins are not intended to be driven by an external source. The speaker amplifier current is measured internally and requires no external connections.

### **Brownout-Prevention Engine**

The brownout-prevention engine (BPE) allows the device to reduce its contribution to the overall system power consumption by attenuating the amplifier output when the supply drops below a set of programmable thresholds. The BPE is enabled and disabled using the *[BPE\\_EN](#page-167-1)* bit. The BPE can be enabled at any time by setting the *[BPE\\_EN](#page-167-1)* bit

high. However, the BPE must not be disabled when it is active (in critical supply levels 0 through 3). The BPE can be disabled safely at any time that it is inactive. The input to the BPE controller is selected using the *[BPE\\_SRC\\_SEL](#page-166-0)* bit. By default, the input to the BPE controller is the measurement ADC PVDD channel. If the selected measurement ADC channel is not already active, enabling the BPE will automatically enables it. The sample rate and filter settings for the measurement ADC determine the speed at which the BPE updates.



*Figure 27. BPE Block Diagram* 

## **BPE State Controller and Level Thresholds**

There are a total of four BPE critical supply levels, each with individually programmable thresholds. The thresholds for each level are configured with the *[BPE\\_L0\\_VTHRESH](#page-153-0)* to *[BPE\\_L3\\_VTHRESH](#page-151-2)* bit fields respectively. The BPE state controller monitors the measurement ADC channel results and automatically makes state changes.

## **Table 9. Brownout-Prevention Engine Levels**



The brownout engine supports hysteresis on the levels. This behaves as follows:

- When in level N, transition to level N + 1 when  $V_{\text{SUPPLY}}$  stays above (level N threshold) + (hysteresis)
- When in level N, transition to level N 1 when V<sub>SUPPLY</sub> falls below level N threshold

The amount of hysteresis is defined by the *[BPE\\_VTHRESH\\_HYST](#page-165-0)* register. The hysteresis is only applied to the thresholds when supply voltage is increasing. The amount of hysteresis can be defined as larger than the distance between two levels.

Thresholds must be configured so that the level N threshold is greater than the sum of the level N - 1 threshold and the hysteresis. For example, if the level 2 threshold is set to 3.15V and hysteresis is set to 25mV, then the level 3 threshold must be set to 3.1875V or higher.

The current level that the BPE is in can be read-back using the *[BPE\\_STATE](#page-151-3)* register. The *[BPE\\_LOWEST](#page-166-1)* register contains the lowest BPE level that the controller has visited since the last time the *[BPE\\_LOWEST](#page-166-1)* register was read.
### **BPE Level Configuration Options**

For a given BPE level, the following options are configurable to reduce the overall device current draw:

Gain Attenuation Function

Limiter Function

Each of these configuration options are individually configurable for each BPE level.

### **BPE Gain Attenuation Function**

The speaker gain attenuation block reduces the overall current draw by the device at low supply voltages by applying smooth digital gain changes to the signal path. The maximum attenuation that can be applied can be independently configured for each BPE level. The maximum attenuation that can be applied for each level is programmable from 0 to -31dB in 1dB steps and is configured using the *[BPE\\_Lx\\_MAXATTN](#page-158-0)* bits.

When the V<sub>SUPPLY</sub> voltage level falls below the programmed level, the brownout controller waits for a time equal to the programmed dwell time for the level before applying attenuation at the programmed attack rate. The brownout controller then enters the hold time phase when the  $V_{\text{SUPPI}}$  y voltage increases and causes the brownout controller to enter the next level. After the programmed hold time for a BPE level expires, the controller enters the release phase where the controller releases the attenuation at the programmed release rate. Additionally, each BPE level has independent programmable settings for dwell time (*[BPE\\_Lx\\_DWELL](#page-154-0)*), hold time (*[BPE\\_Lx\\_HOLD](#page-154-1)*), attack and release step size (*[BPE\\_Lx\\_STEP](#page-156-0)*), attack rate (*[BPE\\_Lx\\_GAIN\\_ATK](#page-159-0)*), and release rate (*[BPE\\_Lx\\_GAIN\\_RLS](#page-159-1)*). While the attack and release rates are independently configurable for the gain attenuation and limiter block, the attack and release step size settings are common.

The *[BPE\\_LOWEST\\_GAIN](#page-166-0)* register contains the lowest gain (highest attenuation) applied the brownout controller. The register is updated upon reading to show the current attenuation applied by the BPE.

### **BPE Limiter Function**

The BPE limiter function allows the device to reduce the overall current draw at low supply levels by quickly attenuating (15μs) input signals that exceed a programmed threshold. When the BPE limiter is enabled (*[BPE\\_LIM\\_EN](#page-167-0)* = 1), the device ignores the Signal Distortion Limiter and Signal Level Limiter settings in the DHT. In this state, the limiter knee threshold is determined solely by the BPE limiter setting of the current BPE level. Each BPE level has an individually configured limiter threshold (set by *[BPE\\_Ln\\_LIM](#page-161-0)*) that is programmable from 0dBFS and -15dBFS in 1dB steps.

Input signals that exceed the limiter knee threshold are attenuated, while input signals below the threshold are not. Each BPE level has an individually configured attack and release step size (*[BPE\\_Lx\\_STEP](#page-156-0)*), attack rates (*[BPE\\_Lx\\_LIM\\_ATK](#page-163-0)*), and release rates (*[BPE\\_Lx\\_LIM\\_RLS](#page-163-1)*). While the attack and release rates are independently configurable for the gain attenuation and limiter block, the attack and release step size settings are common.

The *[BPE\\_LOWEST\\_LIMIT](#page-167-1)* register contains the lowest limiter setting applied the brownout controller. The register is updated upon reading to show the current limiter setting applied by the BPE.

### **Brownout Interrupts**

The BPE can generate interrupts triggered by the following conditions:

- BPE controller enters level 0 (*[BPE\\_L0\\_\\*](#page-97-0)*)
- BPE controller changes from one level to another (*[BPE\\_LEVEL\\_\\*](#page-97-1)*)
- BPE controller is active (*[BPE\\_ACTIVE\\_BGN\\_\\*](#page-97-2)*)
- BPE controller is no longer active (*[BPE\\_ACTIVE\\_END\\_\\*](#page-97-3)*)

See the *[Interrupts](#page-48-0)* section for more information.

### **Measurement ADC**

The device features a configurable 9-bit measurement ADC. The measurement ADC has three channels, one for die temperature measurement (measurement ADC thermal channel), one for PVDD supply voltage measurement (measurement ADC PVDD channel), and one for VBAT supply voltage measurement (measurement ADC VBAT channel). Enabled channels are measured sequentially and continuously. The programmable measurement ADC sample rate can be set independently for each channels. Each channel separately provides an optional programmable lowpass IIR filter.

### **Measurement ADC Thermal Channel**

When the device is clocked in the active state  $(EN = 1)$ , the measurement ADC thermal channel automatically activates. When active, it continuously measures and reports the device die temperature over the range from -29°C to +150°C.

The output of the thermal ADC channel can be readback through the *[MEAS\\_ADC\\_THERM\\_DATA](#page-144-0)* bit field and is the input to both the thermal protection and thermal foldback blocks. By default (*[MEAS\\_ADC\\_THERM\\_RD\\_MODE](#page-142-0)* = 0), the thermal readback value is automatically updated after each conversion is completed. Setting *[MEAS\\_ADC\\_THERM\\_RD\\_MODE](#page-142-0)* to 1 places thermal readback into manual mode. In manual mode, the thermal readback result is updated manually when 1 is written to the *[MEAS\\_ADC\\_THERM\\_UPD](#page-142-1)* bit field. The ADC thermal channel data readback in manual mode and the data streamed through the PCM interface are 9 bits. In automatic mode, since the 9-bit data readback is from two registers, the LSB register is not guaranteed to be synchronous with the MSB register and can result in higher noise.

The thermal ADC channel also provides an optional lowpass IIR filter with a programmable bandwidth that scales relative to the measurement ADC sample rate. The filter is enabled with the *[MEAS\\_ADC\\_TEMP\\_FILT\\_EN](#page-141-0)* bit field, and the bandwidth is set with the *[MEAS\\_ADC\\_TEMP\\_FILT\\_COEFF](#page-141-1)* bit field.

### **Measurement ADC PVDD Channel**

When the device is clocked and in the active state ( $EN = 1$  $EN = 1$ ), the measurement ADC PVDD channel can be enabled. The PVDD channel is manually enabled by setting the MEAS\_ADC\_PVDD\_EN bit to 1 and must be manually enabled for the DHT to operate. When the channel is enabled, it continuously measures and reports the PVDD supply voltage level over the range of 2.5V to 20V.

The output of the measurement ADC PVDD channel can be read back through the *[MEAS\\_ADC\\_PVDD\\_DATA](#page-143-0)* bit field and is routed to the DHT. By default (*[MEAS\\_ADC\\_PVDD\\_RD\\_MODE](#page-142-2)* = 0), the PVDD readback value is automatically updated after each conversion is completed. Setting *[MEAS\\_ADC\\_PVDD\\_RD\\_MODE](#page-142-2)* to 1 places PVDD readback into manual mode. In manual mode, the readback result is updated when 1 is written to the *[MEAS\\_ADC\\_PVDD\\_RD\\_UPD](#page-142-3)*  bit field. The ADC PVDD channel data readback in manual mode and the data streamed through the PCM interface are 9 bits. In automatic mode, since the 9-bit data readback is from two registers, the LSB register is not guaranteed to be synchronous with the MSB register and can result in higher noise.

The lowest PVDD measurement is readback through the *[LOWEST\\_PVDD\\_DATA\\_MSB](#page-145-0)* and *[LOWEST\\_PVDD\\_DATA\\_LSB](#page-145-1)* bit fields. These bit fields both automatically clear and are reset to the value of the current measurement result immediately after LSB readback is completed.

The PVDD channel also provides an optional lowpass IIR filter with a programmable bandwidth that scales relative to the measurement ADC sample rate. The filter is enabled with the *[MEAS\\_ADC\\_PVDD\\_FILT\\_EN](#page-140-0)* bit and the bandwidth is set with the *[MEAS\\_ADC\\_PVDD\\_FILT\\_COEFF](#page-140-1)* bit field.

### **Measurement ADC VBAT Channel**

When the device is clocked in the active state ( $EN = 1$ ), the measurement ADC VBAT channel can be enabled. The VBAT channel is manually enabled by setting the *[MEAS\\_ADC\\_VBAT\\_EN](#page-146-0)* bit to 1 and must be manually enabled for the DHT to operate. When the channel is enabled, it continuously measures and reports the VBAT supply voltage level over the range of 2.5V to 5.5V.

The output of the measurement ADC VBAT channel can be read back through the *[MEAS\\_ADC\\_VBAT\\_DATA](#page-145-2)* bit field. By default (*[MEAS\\_ADC\\_VBAT\\_RD\\_MODE](#page-142-4)* = 0), the VBAT readback value is automatically updated after each conversion

is completed. Setting **[MEAS\\_ADC\\_VBAT\\_RD\\_MODE](#page-142-4)** to 1 places VBAT readback into manual mode. In manual mode, the readback result is updated when 1 is written to the *[MEAS\\_ADC\\_VBAT\\_RD\\_UPD](#page-142-4)* bit field. The ADC VBAT channel data readback in manual mode and the data streamed through the PCM interface are 9 bits. In automatic mode, since the 9-bit data readback is from two registers, the LSB register is not guaranteed to be synchronous with the MSB register and can result in higher noise.

The lowest measured VBAT measurement result is readback through the *[LOWEST\\_VBAT\\_DATA\\_MSB](#page-145-2)* and *[LOWEST\\_VBAT\\_DATA\\_LSB](#page-146-1)* bit fields. These bit fields both automatically clear and are reset to the value of the current measurement result immediately after LSB readback is completed.

The VBAT channel also provides an optional lowpass IIR filter with a programmable bandwidth that scales relative to the measurement ADC sample rate. The filter is enabled with the *[MEAS\\_ADC\\_VBAT\\_FILT\\_EN](#page-141-2)* bit and the bandwidth is set with the *[MEAS\\_ADC\\_VBAT\\_FILT\\_COEFF](#page-141-3)* bit field.

### **Clock and Data Monitors**

The device has input data and external clock monitors that detect host and system level faults. The data monitor detects persistent stuck and high amplitude input signals while the clock monitor detects external clock failures and invalid clock configurations. Upon fault detection, these monitors automatically place the device into software shutdown to stop glitches and unwanted signals at the amplifier output and speaker load.

### **Input Data Monitor**

The device has an optional input data monitor that is enabled by setting *[DMON\\_MAG\\_EN](#page-118-0)* to 1 for the data magnitude monitor or *[DMON\\_STUCK\\_EN](#page-118-1)* to 1 (for the data stuck monitor). Once the data monitor is enabled, it actively monitors the selected input data (from DIN) to the speaker amplifier path anytime the device exits software shutdown (*[EN](#page-168-0)* = 1) and the amplifier is enabled (*[SPK\\_EN](#page-139-0)* = 1). When the tone generator is enabled, the data monitor is automatically disabled.

When active, the block monitors the selected input data for the enabled data error types (data magnitude, data stuck, or both). The *[DMON\\_DURATION](#page-117-0)* bit field selects the duration that a data stuck or magnitude error must persist before a data error is detected. Once a data error is detected, the data monitor automatically places the device into software shutdown (sets EN to 0) and generates a data monitor error interrupt (*[DMON\\_ERR\\_\\*](#page-96-0)*).

A data stuck error is detected if the input signal repeats a fixed value with a magnitude (positive or negative) that is beyond the data stuck threshold (*[DMON\\_STUCK\\_THRES](#page-116-0)*) for longer than the data error duration (set by *[DMON\\_DURATION](#page-117-0)*). If the input signal repeats a fixed value for any duration with a magnitude that is within the data stuck threshold limits (such as a zero or near zero code), no data stuck error is detected.



*Figure 28. Data Monitor Error Generation due to Input Data Stuck Error Detection* 

A data magnitude error is detected if the input signal magnitude (positive or negative) is beyond the data magnitude threshold (set by *[DMON\\_MAG\\_THRES](#page-116-1)*) for longer than the data error duration (set by *[DMON\\_DURATION](#page-117-0)*).



*Figure 29. Monitor Error Generation due to Input Data Magnitude Error Detection* 

### **Clock Monitor**

The device provides an optional clock monitor that is enabled by setting *[CMON\\_EN](#page-118-2)* to 1. Once enabled, it actively monitors the input BCLK and LRCLK anytime the device exits software shutdown (*[EN](#page-168-0)* = 1). When the tone generator is enabled, the clock monitor is automatically disabled. When active, the clock monitor detects clock activity, clock frequency, and frame timing (clock ratio). If faults are detected, the clock monitor automatically places the device into software shutdown and generates a clock error interrupt (*[CLK\\_ERR\\_\\*](#page-96-1)*).

The clock monitor operates in automatic mode when *[CMON\\_AUTORESTART\\_EN](#page-116-2)* = 1 and manual mode when *[CMON\\_AUTORESTART\\_EN](#page-116-2)* = 0. In automatic mode, when a clock error places the device into software shutdown, the global enable bit (EN) is not changed (remains 1) and the device automatically recovers from all clock errors. In automatic mode, both clock error (*[CLK\\_ERR\\_\\*](#page-96-1)*) and clock recovery (*[CLK\\_RECOVER\\_\\*](#page-96-2)*) interrupts are generated in pairs (a clock recovery interrupt is not possible until after a clock error has occurred).

In manual mode, when a clock error places the device into software shutdown, the global enable bit (*[EN](#page-168-0)*) is set to 0. Clock recovery (*[CLK\\_RECOVER\\_\\*](#page-96-2)*) interrupts are never generated in manual mode, and the device remains in software shutdown until the host sets *[EN](#page-168-0)* back to 1. Once the device is re-enabled (*[EN](#page-168-0)* set to 1), the clock monitor is active and detects any new (or persisting) clock errors. If a clock error is detected, the device returns to software shutdown (*[EN](#page-168-0)* = 0), and a new clock error interrupt (*[CLK\\_ERR\\_\\*](#page-96-1)*) is generated.

Clock errors are fault conditions, and audible glitches may occur on clock monitor based transitions into and out of software shutdown. When the clock monitor is enabled, no false clock error or clock recovery interrupts are generated when the host software transitions the device normally into and out of software shutdown.

### **Clock Activity and Frequency Detection**

When the clock monitor is enabled, the bit clock (BCLK) and frame clock (LRCLK) frequencies are monitored. The expected LRCLK frequency is equal to the PCM sample rate (*[PCM\\_SR](#page-121-0)*). The expected BCLK frequency is based on the BCLK to LRCLK ratio (*[PCM\\_BSEL](#page-120-0)*) relative to the PCM sample rate (*[PCM\\_SR](#page-121-0)*).

The current frequency of each clock is measured relative to (and once per interval of) the programmed frame period (as set by [PCM\\_SR](#page-121-0)). A clock frequency error is detected when the measured clock frequencies differ from programmed clock frequencies (faster or slower) by more than the frequency error threshold (45% typ). If either clock stops high or low, the frequency measurement result allows detection of the clock stop event.

The *[CMON\\_ERRTOL](#page-116-3)* bit field sets the clock frequency error tolerance. The tolerance is the required number of consecutive frame clock periods (*[PCM\\_SR](#page-121-0)*) with an incorrect clock frequency before a clock error is generated. If the error persists for the selected number of frame periods, a clock error interrupt (*[CLK\\_ERR\\_\\*](#page-96-1)*) is generated and the device is placed into software shutdown.

In automatic mode, the *[CMON\\_ERRTOL](#page-116-3)* bit field also sets the number of consecutive frame clock periods without clock frequency errors (LRCLK or BCLK) that are required for automatic restart to occur. Once the selected number of

consecutive error-free frames are detected, a clock recovery interrupt (*[CLK\\_RECOVER\\_\\*](#page-96-2)*) is generated and the device automatically exits software shutdown.

In manual mode, no clock recovery interrupts are generated. The clock monitor remains disabled and the device remains in software shutdown until the host software sets *[EN](#page-168-0)* back to 1.



*Figure 30. Clock Monitor LRCLK Rate Error Example With CMON\_ERRTOL = 0x1* 

### **Clock Frame Error Detection**

When the clock monitor is enabled, the bit clock (BCLK) to frame clock (LRCLK) ratio is monitored. The clock monitor counts the number of BCLK periods per frame (LRCLK period) and then compares the count to the configured clock ratio (*[PCM\\_BSEL](#page-120-0)*). In addition, in I<sup>2</sup>S and left-justified modes the clock monitor verifies the LRCLK duty cycle by checking that the number of BCLK periods per channel is equal. In TDM mode, data transport is synchronized to the active frame clock (LRCLK) edge, so no duty cycle restrictions are enforced.

A frame error is detected in each frame where the monitored clock ratio (and duty cycle in I2S and left-justified modes) differs from the configured settings. The *[CMON\\_BSELTOL](#page-115-0)* bit field sets the number of consecutive frames with frame errors that are required before a clock error interrupt is generated (*CLK ERR\_\**) and the device is placed into software shutdown.

In automatic mode, the *[CMON\\_BSELTOL](#page-115-0)* bit field also sets the number of consecutive frames without frame errors that are required for automatic restart to occur. Once the selected number of consecutive error-free frames are detected, a clock recover interrupt (*[CLK\\_RECOVER\\_\\*](#page-96-2)*) is generated and the device automatically exits software shutdown.

In manual mode, no clock recovery interrupts are generated; the clock monitor remains disabled and the device remains in software shutdown until the host software sets *[EN](#page-168-0)* back to 1.



*Figure 31. Clock Monitor Framing Error Example In TDM Mode With PCM\_BSEL = 0x6 and CMON\_BSELTOL = 0x0* 

### **Speaker Monitor**

The speaker monitor is a circuit that is designed to protect the speaker against amplifier signals that could damage it. The speaker monitor is enabled by default and can be disabled by setting the *[SPKMON\\_EN](#page-118-3)* bit to zero. The circuit monitors the amplifier's PWM signal and shuts down the amplifier output when the signal goes above a programmed speaker

monitor threshold (set by *[SPKMON\\_THRESH](#page-117-1)*) for a programmed amount of time (set by *[SPKMON\\_DURATION](#page-118-4)*). Additionally, the device also generates an *[INT\\_SPKMON\\_ERR](#page-95-0)* interrupt.

The speaker monitor circuit uses PWM signals of the amplifier, and in the case of DC signals, the amplifier accurately detects signals above the threshold. However, for a sine wave with a DC offset, the average DC detected by the circuit is lower because of the zeroes presented from the sine signal. In this case, the speaker monitor threshold (set by *[SPKMON\\_THRESH](#page-117-1)*) and/or the speaker monitor duration (set by *[SPKMON\\_DURATION](#page-118-4)*) can be adjusted to protect the speaker against the DC present in the signal.

### **Thermal Protection**

When the device is active, the measurement ADC thermal channel is automatically enabled and monitors die temperature to ensure that it does not exceed the configured thermal thresholds. Interrupt registers are provided so that the device can notify the host when the die temperature crosses either the thermal warning or thermal-shutdown threshold, or when thermal foldback starts and stops.

### **Thermal Warning and Thermal Shutdown Configuration**

The device features two thermal-warning thresholds. The thermal-warning thresholds are configured by *[THERMWARN1\\_THRESH\[6:0\]](#page-112-0)* and *[THERMWARN2\\_THRESH\[6:0\]](#page-112-1)* bit fields and the thermal shutdown threshold is configured by the **THERMSHDN THRESH[5:0]** bit field. The thermal-warning2 threshold should always be set to a temperature higher than or equal to thermal-warning1 temperature. Additionally, the thermal-warning2 threshold temperature should never be configured higher than the thermal shutdown threshold temperature minus hysteresis. When the die temperature is decreasing, hysteresis is applied to thermal-shutdown, thermal-warning1, and thermalwarning2 thresholds. The temperature hysteresis is configured with *[THERM\\_HYST](#page-113-0)* bit field.

### **Thermal Shutdown Recovery Configuration**

The device thermal-shutdown recovery behavior is determined by the state of *[THERM\\_AUTORESTART\\_EN](#page-167-2)* bit. When the *[THERM\\_AUTORESTART\\_EN](#page-167-2)* bit is set to 0, the thermal-shutdown recovery is in manual mode. In manual mode, when the die temperature exceeds the thermal shutdown threshold, an interrupt is generated and the amplifier output is automatically disabled. Once the die temperature drops below both of the thermal-shutdown threshold minus the hysteresis and the thermal-warning2 threshold minus the hysteresis, the appropriate interrupts are generated to notify the host. In addition, once the die temperature drops below the thermal-warning2 threshold minus the hysteresis, the device is placed into software shutdown (*[EN](#page-168-0)* is set to 0) and remains in that state until the host manually re-enables the device. When the *[THERM\\_AUTORESTART\\_EN](#page-167-2)* bit is set to 1, the thermal-shutdown recovery is in automatic mode. In automatic mode, when the die temperature exceeds the thermal-shutdown threshold, an interrupt is generated and the amplifier is automatically disabled. Once the die temperature drops below the thermal shutdown threshold minus the hysteresis, an interrupt is generated but the amplifier remains disabled. When the temperature drops below the thermalwarning2 threshold minus the hysteresis, another interrupt is generated and (unlike manual mode) the amplifier is then automatically re-enabled.

### **Thermal Foldback**

The device features thermal foldback to allow for a smoother audio response to high temperature events. Thermal foldback is enabled by setting the *[THERMFB\\_EN](#page-114-0)* bit to 1. The device provides two thermal-warning thresholds that are configured by the *[THERMWARN1\\_THRESH\[6:0\]](#page-112-0)* and *[THERMWARN2\\_THRESH\[6:0\]](#page-112-1)* bit fields. They should be set such that thermal-warning2 threshold temperature is higher than or equal to thermal-warning1 threshold temperature. Additionally, the thermal-warning2 threshold temperature should never be configured higher than the thermal shutdown threshold temperature minus hysteresis.

Once enabled, thermal foldback begins when the die temperature exceeds the configured thermal-warning1 threshold, interrupts are generated (*[THERMFB\\_BGN\\_\\*](#page-95-1)* and *[THERMWARN1\\_BGN\\_\\*](#page-95-2)*), and attenuation is applied to the speaker amplifier path. The slope of the thermal foldback attenuation is programmed with the *[THERMFB\\_SLOPE1](#page-113-1)* bit field.

If the die temperature continues to increase and exceeds the configured thermal-warning2 threshold, an interrupt (*[THERMWARN2\\_BGN\\_\\*](#page-95-3)*) is generated and attenuation continues to be applied to the speaker amplifier path. The slope of the attenuation applied to the speaker amplifier path is programmed with the *[THERMFB\\_SLOPE2](#page-113-2)* bit field. As the die

temperature increases, the level of attenuation also increases proportionally up to a maximum level of -12dB (unless the thermal shutdown threshold is exceeded first). When thermal foldback is active, the attack time for a gain change can be a maximum of two samples. Additionally, there is a sample delay in the signal path attenuation due to the group delay of the amplifier.

When the die temperature starts to decrease and drops below the thermal-warning2 threshold minus the programmed hysteresis level and remains there for longer than the programmed hold time (set with the *[THERMFB\\_HOLD](#page-113-3)* bit field), the attenuation starts to release and an interrupt (*[THERMWARN2\\_END\\_\\*](#page-95-4)*) is generated. If the die temperature continues to reduce and drops below the thermal-warning1 threshold temperature minus the programmed hysteresis level, thermal foldback ends and interrupts (*[THERMFB\\_END\\_\\*](#page-95-5)* and *[THERMWARN1\\_END\\_\\*](#page-95-6)*) are generated. The attenuation release rate (for decreasing temperature) is programmable and configured with the *[THERMFB\\_RLS](#page-113-4)* bit field.

### **Tone Generator**

The device includes a tone generator which is enabled using the *[TONE\\_EN](#page-132-0)* bit field and replaces the PCM interface as the input source to the speaker playback path. When the tone generator is enabled, both it and the speaker playback path operate without the need for any external clocks.

The tone generator output is configured to generate sine wave or DC tones (using the *[TONE\\_CONFIG](#page-131-0)* bit field).

The tone generator can create sine waves with either a 1kHz fixed frequency or a variable sample rate dependent frequency. When a sample rate based sinewave output is selected, the tone generator output frequency is set by the playback sample rate setting (*[PCM\\_SR](#page-121-0)*) divided by the selected ratio (as set by *[TONE\\_CONFIG](#page-131-0)*). For the playback sample rate of 44.1kHz and its multiples, the tone generator output frequency can vary by up to 9%. The tone generator supports all available sample rate settings (*[PCM\\_SR](#page-121-0)*)*.* The amplitude of the output sine wave relative to full-scale is selected with the *[TONE\\_AMPLITUDE](#page-131-1)* bit field.

The tone generator can output either a fixed or a programmable DC output level (as set by *[TONE\\_CONFIG](#page-131-0)*). Fixed DC output levels of zero code, positive half-scale, and negative half-scale are provided for quick configuration. If the programmable DC output level is selected (*[TONE\\_CONFIG](#page-131-0)*), the DC level is configured as a signed two's complement value with the *[TONE\\_DC](#page-131-2)* bit field.

### **Pink Noise Generator**

The device includes a pink noise generator, which is enabled using the PINK\_NOISE\_EN bit field and replaces the PCM interface as the input source to the speaker playback path. The pink noise generator and the tone generator cannot be enabled at the same time. When the pink noise generator is enabled, both it and the speaker playback path operate without the need for any external clocks. The output level of the generator is fixed, so the amplifier gain and speaker volume must be used to adjust the level.

### **Interchip Communication**

The device features an interchip communication (ICC) interface that uses a shared data bus to facilitate synchronized speaker amplifier path attenuation adjustments across groups of devices. Depending on the configuration, the ICC interface can synchronize brownout-prevention engine (BPE), DHT, and thermal foldback. Each device receives the data of the other grouped devices and reacts accordingly.

### **ICC Operation and Data Format**

The bidirectional ICC bus is used to synchronize the responses of grouped devices. To create the ICC bus, the ICC interface pins of each device are externally connected together (whether or not the devices are in the same group). The ICC bus operates with the same clock sources and data format configuration as the PCM interface data input (DIN), and can support a maximum of 16 channels. For a given valid PCM interface configuration, the number of available ICC data channels per frame is calculated as follows (based on the *[PCM\\_CHANSZ](#page-119-0)*, *[PCM\\_BSEL](#page-120-0)*, and *[PCM\\_FORMAT](#page-119-1)* settings):

Number of Available Data Input Channels = BCLK to LRCLK Ratio / Channel Length

The ICC interface is disabled when both the ICC data transmit output (*[ICC\\_TX\\_EN](#page-130-0)*) and the ICC data link (*[ICC\\_LINK\\_EN](#page-130-1)*) are disabled. To enable the ICC interface, both *[ICC\\_TX\\_EN](#page-130-0)* and *[ICC\\_LINK\\_EN](#page-130-1)* must be set to

1. It is invalid to set these controls to different values, and both must always be set to the same state (either enabled or disabled). Once the ICC link and data transmit are enabled, the ICC data output channel is assigned with the *ICC* TX DEST bit field. The ICC pin is Hi-Z during all other data channels, and can be configured (with the *[ICC\\_RX\\_CHn\\_EN](#page-129-0)* bits) to accept data from the output data channels of grouped devices.

The transmitted ICC data is always the same size as the configured PCM data input word size (set by *[PCM\\_CHANSZ](#page-119-0)*). When a 16-bit data word is selected, the ICC data word is not long enough to synchronize BPE, DHT, and thermal foldback. In this case, the *[ICC\\_DATA\\_SEL](#page-130-3)* bit is used to choose whether the DHT function or thermal foldback function is synchronized in addition to BPE state. When a 24-bit or 32-bit data word size is selected, ICC can synchronize BPE, DHT, and thermal foldback across a given group. In these cases, the *[ICC\\_DATA\\_SEL](#page-130-3)* bit has no effect. Active ICC data channels always contain ICC data words followed by zero padding bit up to the ICC data channel length (which is equal to PCM input data channel length). If BPE, DHT, or thermal foldback is disabled for any given group, then the transmitted ICC data for the disabled function(s) is always zero code.

### **Multiamplifier Grouping**

The ICC interface allows multiple devices to be grouped so that BPE, DHT, and thermal foldback behavior can be synchronized. The receive channel enables (*[ICC\\_RX\\_CHn\\_EN](#page-129-1)*) are used to define groups. A given device monitors all selected channels (when *[ICC\\_RX\\_CHn\\_EN](#page-129-1)* = 1, and n denotes the channel number) on the ICC data bus. The configured set of receive channels must also include the assigned transmit channel (as set by *[ICC\\_TX\\_DEST](#page-130-2)*) for any given device. Each device in a given group must have identical settings for all ICC receive channel enables (*[ICC\\_RX\\_CHn\\_EN](#page-129-1)*). Furthermore, all devices in the same group must have identical DHT, thermal protection, and thermal foldback settings to achieve a synchronized response across the group. The behavior of a group as a whole is undefined if any given device in a group has different settings.

The ICC bus can support a maximum of 16 data channels. The minimum size of a group is two devices, and as a result the maximum number of concurrent groups on a single ICC bus is eight. A group can contain as many as 16 devices, but then only a single group is possible on a single ICC bus.

### **ICC Multi-Group Example**

Consider a system design that includes four devices that require DHT synchronization, and that two distinct groups of two devices each (with different DHT settings) must share single ICC bus. The PCM interface (and thus ICC bus) is configured in TDM mode 1 with four 16-bit data channels available. One possible configuration (among many) is to assign devices 1 and 3 to the first group (denoted group A), and to assign devices 2 and 4 to the second group (denoted group B).

To configure group A, both devices 1 and 3 are set to monitor channels 0 and 2 by programming *[ICC\\_RX\\_CH0\\_EN](#page-129-0)* = 1 and *[ICC\\_RX\\_CH2\\_EN](#page-129-2)* = 1 on both devices (all other ICC receive bit fields are 0). Device 1 transmits on channel 0 (*[ICC\\_TX\\_DEST](#page-130-2)* = 0x0) and device 3 transmits on channel 2 (*[ICC\\_TX\\_DEST](#page-130-2)* = 0x2).

To configure group B, both devices 2 and 4 are set to monitor channels 1 and 3 by programming *[ICC\\_RX\\_CH1\\_EN](#page-129-1)* = 1 and *[ICC\\_RX\\_CH3\\_EN](#page-129-3)* = 1 on both devices (all other ICC receive bit fields are 0). Device 2 transmits on channel 1 (*[ICC\\_TX\\_DEST](#page-130-2)*= 0x1) and device 4 transmits on channel 3 (*[ICC\\_TX\\_DEST](#page-130-2)* = 0x3).

Since the ICC channel length and data word size is limited to 16 bits, the *[ICC\\_DATA\\_SEL](#page-130-3)* bit field in all four devices must be set to 0 to select DHT target attenuation synchronization. Finally, on all four devices set *ICC\_LINK\_EN* = 1 and *[ICC\\_TX\\_EN](#page-130-0)* = 1 to enable the ICC interfaces.



*Figure 32. ICC Multi-Group Example with 2 Groups and 4 Total Devices* 

## **I 2C Serial Interface**

The  $12C$  serial control interface is activated when the device detects a valid  $12C$  start condition at the I2C1 and I2C2 pins. The I2C1 and I2C2 pins can each act as either SCL or SDA respectively, and the start condition configures the device address and state of each pin. After the first  $12C$  transaction, the  $12C$  interface configuration should remain fixed.

### **Slave Address**

The slave address is defined as the seven most significant bits (MSBs) followed by the read/write bit. The seven most significant bits are programmable through the ADDR connection, and the required I2C1 and I2C2 connections for each address are shown in [Table 10](#page-80-0). The device does not communicate if ADDR is floating. Setting the read/write bit to 1 configures the device for read mode. Setting the read/write bit to 0 configures the device for write mode. The address is the first byte of information sent to the IC after the START condition.



## <span id="page-80-0"></span>**Table 10. I2C Slave Address**

The IC features an I2C/SMBus™-compatible, 2-wire serial interface consisting of a serial data line (SDA) and a serial clock line (SCL). SDA and SCL facilitate communication between the IC and the master at clock rates up to 1MHz. [Figure](#page-81-0)  [33](#page-81-0) shows the 2-wire interface timing diagram. The master generates SCL and initiates data transfer on the bus. The master device writes data to the IC by transmitting the proper slave address followed by two register address bytes (most significant byte first) and then the data word. Each transmit sequence is framed by a START (S) or REPEATED START (Sr) condition and a STOP (P) condition. Each word transmitted to the IC is 8 bits long and is followed by an acknowledge clock pulse. A master reading data from the IC transmits the proper slave address followed by a series of nine SCL pulses. The IC transmits data on SDA in sync with the master-generated SCL pulses. The master acknowledges receipt of each byte of data. Each read sequence is framed by a START (S) or REPEATED START (Sr) condition, a not acknowledge, and a STOP (P) condition. SDA operates as both an input and an open-drain output. A pullup resistor, typically greater than 500Ω, is required on SDA. SCL operates only as an input. A pullup resistor, typically greater than 500Ω, is required on SCL if there are multiple masters on the bus, or if the single master has an open-drain SCL output. Series resistors in line with SDA and SCL are optional. Series resistors protect the digital inputs of the IC from high voltage spikes on the bus lines, and minimize crosstalk and undershoot of the bus signals.

<span id="page-81-0"></span>

*Figure 33. I2C Interface Timing Diagram* 

### **Bit Transfer**

One data bit is transferred during each SCL cycle. The data on SDA must remain stable during the high period of the SCL pulse. Changes in SDA while SCL is high are control signals (see the *[START and STOP Conditions](#page-81-1)* section).

### <span id="page-81-1"></span>**START and STOP Conditions**

SDA and SCL idle high when the bus is not in use. A master initiates communication by issuing a START condition. A START condition is a high-to-low transition on SDA with SCL high. A STOP condition is a low-to-high transition on SDA while SCL is high. A START condition from the master signals the beginning of a transmission to the IC. The master terminates transmission and frees the bus by issuing a STOP condition. The bus remains active if a REPEATED START condition is generated instead of a STOP condition.



*Figure 34. I2C START, STOP, and REPEATED START Conditions* 

### **Early STOP Conditions**

The IC recognizes a STOP condition at any point during data transmission except if the STOP condition occurs in the same high pulse as a START condition. For proper operation, do not send a STOP condition during the same SCL high pulse as the START condition.

### **Acknowledge**

The acknowledge bit (ACK) is a clocked 9th bit that the IC uses to handshake receipt each byte of data when in write mode. The IC pulls down SDA during the entire master-generated 9th clock pulse if the previous byte is successfully received. Monitoring ACK allows for detection of unsuccessful data transfers. An unsuccessful data transfer occurs if a receiving device is busy or if a system fault has occurred. In the event of an unsuccessful data transfer, the bus master retries communication. The master pulls down SDA during the 9th clock cycle to acknowledge receipt of data when the IC is in read mode. An acknowledge is sent by the master after each read byte to allow data transfer to continue. A notacknowledge is sent when the master reads the final byte of data from the IC, followed by a STOP condition.



*Figure 35. I2C Acknowledge* 

### **Write Data Format**

A write to the IC includes transmission of a START condition, the slave address with the R/W bit set to 0, two bytes of data to configure the internal register address pointer, one or more bytes of data, and a STOP condition.

The slave address with the R/W bit set to 0 indicates that the master intends to write data to the IC. The IC acknowledges receipt of the address byte during the master-generated 9th SCL pulse.

The second and third bytes transmitted from the master configure the ICs internal register address pointer. The pointer tells the IC where to write the next byte of data. An acknowledge pulse is sent by the IC upon receipt of the address pointer data.

The third byte sent to the IC contains the data that is written to the chosen register. An acknowledge pulse from the IC signals receipt of the data byte. The address pointer auto-increments to the next register address after each received data byte. This auto-increment feature allows the master to write to sequential registers within one continuous frame. The master signals the end of transmission by issuing a STOP condition.



*Figure 36. I2C Writing One Byte of Data to the Slave* 



*Figure 37. I2C Writing n-Bytes of Data to the Slave* 

### **Read Data Format**

Initiate a read operation by sending the slave address with the R/W bit set to 1. The IC acknowledges receipt of its slave address by pulling SDA low during the 9th SCL clock pulse. A START command followed by a read command resets the address pointer to register 0x2000.

The first byte transmitted from the IC is the content of register 0x00. Transmitted data is valid on the rising edge of SCL. The address pointer auto-increments after each read data byte. This auto-increment feature allows all registers to be read sequentially within one continuous frame. A STOP condition can be issued after any number of read data bytes. If a STOP condition is issued followed by another read operation, the first data byte to be read is from register 0x00.

The address pointer can be preset to a specific register before a read command is issued. The master presets the address pointer by first sending the ICs slave address with the R/W bit set to 0 followed by the two byte register address. A REPEATED START condition is then sent followed by the slave address with the R/W bit set to 1. The IC then transmits the contents of the specified register. The address pointer auto-increments after transmitting the first byte.

The master acknowledges receipt of each read byte during the acknowledge clock pulse. The master must acknowledge

all correctly received bytes except the last byte. The final byte must be followed by a not acknowledge from the master and then a STOP condition.



*Figure 38. I2C Reading One Byte of Data from the Slave* 



*Figure 39. I2C Reading n-Bytes of Data from the Slave* 

## **I 2C Register Map**

### **Control Bit Field Types and Write Access Restrictions**

The device control bit fields fall into one of three basic types: read, write, or read and write. There are no read restrictions, and any read enabled bit field can be read back anytime the  $12C$  control interface is active. However, there are write restrictions, and every write enabled bit field falls into one of two write access subtypes.

The first write access subtype is dynamic. Dynamic bit fields effectively have no write access restrictions, and can be safely changed (written) in any device state where the I<sup>2</sup>C control interface is active. The second bit field access subtype is restricted. Restricted bit fields should only be changed (written) when the related functional block (as shown in [Table](#page-84-0)  [11\)](#page-84-0) is powered down.

If the write access is restricted to the global enable (restrictions EN and ENL), then the restricted bit field should only be changed (written) when the device is in software-shutdown. As a form of system protection, write access to some critical global enable restricted bit fields (ENL) is locked out by the hardware when the device is not in the software-shutdown state. Attempting to change (write to) these locked restricted bit fields when the device is not in the software-shutdown state has no effect (read access is still allowed).

The bit field type and write access subtype is provided for every bit field in the detailed register description tables. For all bit fields with the restricted subtype, the dependency is also denoted in the "RES" column.

[Table 11](#page-84-0) provides a detailed description of all device register types, access subtypes, and restriction dependencies that are used by this device. The write access restrictions describe the specific device condition(s) that should be met (and the corresponding bit field settings) before the system attempts to change (write to) bit fields with that restriction type.

## <span id="page-84-0"></span>**Table 11. Control Bit Types and Write Access Restrictions**





# **Table 11. Control Bit Types and Write Access Restrictions (continued)**

# **Register Map**

## <span id="page-86-0"></span>**Register Map**



















## **Register Details**

## <span id="page-94-0"></span>**[Software Reset \(0x2000\)](#page-86-0)**



## <span id="page-94-1"></span>**[Interrupt Raw 1 \(0x2001\)](#page-86-0)**



<span id="page-95-6"></span><span id="page-95-2"></span>

### <span id="page-95-7"></span><span id="page-95-5"></span><span id="page-95-1"></span>**[Interrupt Raw 2 \(0x2002\)](#page-86-0)**

<span id="page-95-4"></span><span id="page-95-3"></span><span id="page-95-0"></span>

<span id="page-96-2"></span><span id="page-96-1"></span>

### <span id="page-96-3"></span><span id="page-96-0"></span>**[Interrupt Raw 3 \(0x2003\)](#page-86-0)**



## <span id="page-96-4"></span>**[Interrupt Raw 4 \(0x2004\)](#page-86-0)**



<span id="page-97-2"></span><span id="page-97-1"></span><span id="page-97-0"></span>

## <span id="page-97-4"></span><span id="page-97-3"></span>**[Interrupt State 1 \(0x2006\)](#page-86-0)**





## <span id="page-98-0"></span>**[Interrupt State 2 \(0x2007\)](#page-86-0)**





### <span id="page-99-0"></span>**[Interrupt State 3 \(0x2008\)](#page-86-0)**



## <span id="page-100-0"></span>**[Interrupt State 4 \(0x2009\)](#page-86-0)**



## <span id="page-100-1"></span>**[Interrupt Flag 1 \(0x200B\)](#page-86-0)**





## <span id="page-102-0"></span>**[Interrupt Flag 2 \(0x200C\)](#page-86-0)**





### <span id="page-103-0"></span>**[Interrupt Flag 3 \(0x200D\)](#page-86-0)**





## <span id="page-104-0"></span>**[Interrupt Flag 4 \(0x200E\)](#page-86-0)**



## <span id="page-105-0"></span>**[Interrupt Enable 1 \(0x2010\)](#page-86-0)**



## <span id="page-106-0"></span>**[Interrupt Enable 2 \(0x2011\)](#page-86-0)**



## <span id="page-107-0"></span>**[Interrupt Enable 3 \(0x2012\)](#page-86-0)**


### **[Interrupt Enable 4 \(0x2013\)](#page-86-0)**



### **[Interrupt Flag Clear 1 \(0x2015\)](#page-86-0)**





## **[Interrupt Flag Clear 2 \(0x2016\)](#page-86-0)**





## **[Interrupt Flag Clear 3 \(0x2017\)](#page-86-0)**



## **[Interrupt Flag Clear 4 \(0x2018\)](#page-86-0)**



## **[IRQ Control \(0x201F\)](#page-86-0)**



## **[Thermal Warning Threshhold \(0x2020\)](#page-86-0)**



## **[Warning Threshold 2 \(0x2021\)](#page-86-0)**



### **[Thermal Shutdown Threshold \(0x2022\)](#page-86-0)**



### **[Thermal Hysteresis \(0x2023\)](#page-86-0)**



### **[Thermal Foldback Settings \(0x2024\)](#page-86-0)**



### **[Thermal Foldback Enable \(0x2027\)](#page-86-0)**



### **[Noise Gate/Idle Mode Control \(0x2030\)](#page-86-0)**



### **[Noise Gate/Idle Mode Enables \(0x2033\)](#page-86-0)**



### **[Clock Monitor Control \(0x2038\)](#page-86-0)**





## **[Data Monitor Control \(0x2039\)](#page-86-0)**





## **[Speaker Monitor Threshold \(0x203A\)](#page-86-0)**



## **[Speaker Monitor Duration \(0x203B\)](#page-86-0)**





## **[Enable Controls \(0x203F\)](#page-86-0)**



### **[Pin Config \(0x2040\)](#page-86-0)**





## **[PCM Mode Config \(0x2041\)](#page-86-0)**



0x4: 64 0x5: 96 0x6: 128



### **[PCM Clock Setup \(0x2042\)](#page-86-0)**





## **[PCM Sample Rate Setup 1 \(0x2043\)](#page-86-0)**





## **[PCM TX Control 1 \(0x2044\)](#page-86-0)**



### **[PCM TX Control 2 \(0x2045\)](#page-86-0)**





### **[PCM TX Control 3 \(0x2046\)](#page-86-0)**



### **[PCM TX Control 4 \(0x2047\)](#page-86-0)**



### **[PCM TX Control 5 \(0x2048\)](#page-86-0)**





### **[PCM TX Control 6 \(0x2049\)](#page-86-0)**



## **[PCM TX Control 7 \(0x204A\)](#page-86-0)**



### **[PCM TX Control 8 \(0x204B\)](#page-86-0)**





### **[PCM Tx HiZ Control 1 \(0x204C\)](#page-86-0)**



### **[PCM Tx HiZ Control 2 \(0x204D\)](#page-86-0)**



### **[PCM Tx HiZ Control 3 \(0x204E\)](#page-86-0)**



### **[PCM Tx HiZ Control 4 \(0x204F\)](#page-86-0)**



### **[PCM Tx HiZ Control 5 \(0x2050\)](#page-86-0)**



### **[PCM Tx HiZ Control6 \(0x2051\)](#page-86-0)**



### **[PCM Tx HiZ Control 7 \(0x2052\)](#page-86-0)**





### **[PCM Tx HiZ Control 8 \(0x2053\)](#page-86-0)**



### **[PCM RX Source 1 \(0x2055\)](#page-86-0)**



## **[PCM RX Source 2 \(0x2056\)](#page-86-0)**





### **[PCM Bypass Source \(0x2058\)](#page-86-0)**



### **[PCM TX Source Enables \(0x205D\)](#page-86-0)**





### **[PCM Rx Enables \(0x205E\)](#page-86-0)**



### **[PCM Tx Enables \(0x205F\)](#page-86-0)**



### **[ICC Rx Enables A \(0x2070\)](#page-86-0)**



### **[ICC Rx Enables B \(0x2071\)](#page-86-0)**





### **[ICC Tx Control \(0x2072\)](#page-86-0)**



## **[ICC Enables \(0x207F\)](#page-86-0)**



### **[Tone Generator and DC Config \(0x2083\)](#page-86-0)**



### **[Tone Generator DC Level 1 \(0x2084\)](#page-86-0)**



### **[Tone Generator DC Level 2 \(0x2085\)](#page-86-0)**



### **[Tone Generator DC Level 3 \(0x2086\)](#page-86-0)**



## **[Tone Generator Enable \(0x208F\)](#page-86-0)**



### **[AMP volume control \(0x2090\)](#page-86-0)**



### **[AMP Path Gain \(0x2091\)](#page-86-0)**





### **[AMP DSP Config \(0x2092\)](#page-86-0)**



X

# MAX98396 20V Digital Input Class-DG Amplifier with I/V Sense and Brownout Prevention

0x3: MMI \* 3/6 0x4: MMI \* 2/6 0x5: MMI \* 1/6 0x6 to 0x7: Reserved



### **[SSM Configuration \(0x2093\)](#page-86-0)**



clocks. The modulation index can be

varied as follows:

### **[SPK Class DG Threshold \(0x2094\)](#page-86-0)**



### **[SPK Class DG Headroom \(0x2095\)](#page-86-0)**





## **[SPK Class DG Hold Time \(0x2096\)](#page-86-0)**



### **[SPK Class DG Delay \(0x2097\)](#page-86-0)**



### **[SPK Class DG Mode \(0x2098\)](#page-86-0)**



### **[SPK Class DG VBAT Level \(0x2099\)](#page-86-0)**



### **[SPK Edge Control \(0x209A\)](#page-86-0)**



### **[SPK Path Wideband Only Enable \(0x209B\)](#page-86-0)**



### **[SPK Edge Control 1 \(0x209C\)](#page-86-0)**



### **[SPK Edge Control 2 \(0x209D\)](#page-86-0)**



### **[Amp Clip Gain \(0x209E\)](#page-86-0)**



## **[Bypass Path Config \(0x209F\)](#page-86-0)**



### **[Amplifier Supply Control \(0x20A0\)](#page-86-0)**



### **[AMP enables \(0x20AF\)](#page-86-0)**



## **[Meas ADC Sample Rate \(0x20B0\)](#page-86-0)**



### **[Meas ADC PVDD Config \(0x20B1\)](#page-86-0)**



### **[Meas ADC VBAT Config \(0x20B2\)](#page-86-0)**



## **[Meas ADC Thermal Config \(0x20B3\)](#page-86-0)**



### **[Meas ADC Readback Control 1 \(0x20B4\)](#page-86-0)**



### **[Meas ADC Readback Control 2 \(0x20B5\)](#page-86-0)**



### **[Meas ADC PVDD Readback MSB \(0x20B6\)](#page-86-0)**



### **[Meas ADC PVDD Readback LSB \(0x20B7\)](#page-86-0)**



### **[Meas ADC VBAT Readback MSB \(0x20B8\)](#page-86-0)**


### **[Meas ADC VBAT Readback LSB \(0x20B9\)](#page-86-0)**



### **[Meas ADC Temp Readback MSB \(0x20BA\)](#page-86-0)**



### **[Meas ADC Temp Readback LSB \(0x20BB\)](#page-86-0)**



### **[Meas ADC Lowest PVDD Readback MSB \(0x20BC\)](#page-86-0)**



### **[Meas ADC Lowest PVDD Readback LSB \(0x20BD\)](#page-86-0)**



### **[Meas ADC Lowest VBAT Readback MSB \(0x20BE\)](#page-86-0)**



### **[Meas ADC Lowest VBAT Readback LSB \(0x20BF\)](#page-86-0)**



## **[Meas ADC Config \(0x20C7\)](#page-86-0)**



## **[DHT Configuration 1 \(0x20D0\)](#page-86-0)**





### **[Limiter Configuration 1 \(0x20D1\)](#page-86-0)**





## **[Limiter Configuration 2 \(0x20D2\)](#page-86-0)**





### **[DHT Configuration 2 \(0x20D3\)](#page-86-0)**



## **[DHT Configuration 3 \(0x20D4\)](#page-86-0)**





### **[DHT Configuration 4 \(0x20D5\)](#page-86-0)**





## **[DHT Supply Hysteresis Configuration \(0x20D6\)](#page-86-0)**





### **[DHT Enable \(0x20DF\)](#page-86-0)**



### **[I\\_V Sense Path Config \(0x20E0\)](#page-86-0)**



## **[I\\_V Sense Path Enables \(0x20E4\)](#page-86-0)**



### **[BPE State \(0x20E5\)](#page-86-0)**



## **[BPE L3 Threshold MSB \(0x20E6\)](#page-86-0)**



### **[BPE L3 Threshold LSB \(0x20E7\)](#page-86-0)**



### **[BPE L2 Threshold MSB \(0x20E8\)](#page-86-0)**



### **[BPE L2 Threshold LSB \(0x20E9\)](#page-86-0)**



## **[BPE L1 Threshold MSB \(0x20EA\)](#page-86-0)**





### **[BPE L1 Threshold LSB \(0x20EB\)](#page-86-0)**



### **[BPE L0 Threshold MSB \(0x20EC\)](#page-86-0)**



### **[BPE L0 Threshold LSB \(0x20ED\)](#page-86-0)**



## **[BPE L3 Dwell and Hold Time \(0x20EE\)](#page-86-0)**



### **[BPE L2 Dwell and Hold Time \(0x20EF\)](#page-86-0)**



## **[BPE L1 Dwell and Hold Time \(0x20F0\)](#page-86-0)**



### **[BPE L0 Hold Time \(0x20F1\)](#page-86-0)**



### **[BPE L3 Attack and Release Step \(0x20F2\)](#page-86-0)**





## **[BPE L2 Attack and Release Step \(0x20F3\)](#page-86-0)**





### **[BPE L1 Attack and Release Step \(0x20F4\)](#page-86-0)**





## **[BPE L0 Attack and Release Step \(0x20F5\)](#page-86-0)**





## **[BPE L3 Max Gain Attn \(0x20F6\)](#page-86-0)**





### **[BPE L2 Max Gain Attn \(0x20F7\)](#page-86-0)**



### **[BPE L1 Max Gain Attn \(0x20F8\)](#page-86-0)**



## **[BPE L0 Max Gain Attn \(0x20F9\)](#page-86-0)**



### **[BPE L3 Gain Attack and Rls Rates \(0x20FA\)](#page-86-0)**



### **[BPE L2 Gain Attack and Rls Rates \(0x20FB\)](#page-86-0)**





### **[BPE L1 Gain Attack and Rls Rates \(0x20FC\)](#page-86-0)**



### **[BPE L0 Gain Attack and Rls Rates \(0x20FD\)](#page-86-0)**





### **[BPE L3 Limiter Config \(0x20FE\)](#page-86-0)**



### **[BPE L2 Limiter Config \(0x20FF\)](#page-86-0)**





### **[BPE L1 Limiter Config \(0x2100\)](#page-86-0)**





## **[BPE L0 Limiter Config \(0x2101\)](#page-86-0)**





## **[BPE L3 Limiter Attack and Release Rates \(0x2102\)](#page-86-0)**



### **[BPE L2 Limiter Attack and Release Rates \(0x2103\)](#page-86-0)**





### **[BPE L1 Limiter Attack and Release Rates \(0x2104\)](#page-86-0)**



### **[BPE L0 Limiter Attack and Release Rates \(0x2105\)](#page-86-0)**





### **[BPE Threshold Hysteresis \(0x2106\)](#page-86-0)**



### **[BPE Infinite Hold Clear \(0x2107\)](#page-86-0)**



### **[BPE Supply Source \(0x2108\)](#page-86-0)**



### **[BPE Lowest State \(0x2109\)](#page-86-0)**



### **[BPE Lowest Gain \(0x210A\)](#page-86-0)**



### **[BPE Lowest Limiter \(0x210B\)](#page-86-0)**



### **[BPE Enable \(0x210D\)](#page-86-0)**



### **[Auto-Restart Behavior \(0x210E\)](#page-86-0)**





## **[Global Enable \(0x210F\)](#page-86-0)**



### **[Revision ID \(0x21FF\)](#page-86-0)**



## **Applications Information**

### **Layout and Grounding**

Proper layout and grounding are essential for optimal performance; use at least four PCB layers and add thermal vias to the ground/power plane close to the device to ensure good thermal performance and high-end output power. Good grounding improves audio performance and prevents switching noise from coupling into the audio signal. Ground the power signals and the analog signals of the IC separately at the system ground plane to prevent switching interference from corrupting sensitive analog signals. Place the recommended supply decoupling capacitors as close as possible to the IC. The PVDD to PGND connection must be kept short and should have minimal trace length and loop area to ensure optimal performance. Use wide, low-resistance output, supply, and ground traces. As load impedance decreases, the current drawn from the device outputs increase. At higher current, the resistance of the output traces decreases the power delivered to the load. For example, if 2W is delivered from the speaker output to a 4Ω load through a 100mΩ trace, 49mW is consumed in the trace. If power is delivered through a 10mΩ trace, only 5mW is consumed in the trace. Wide output, supply, and ground traces also improve the power dissipation of the device. The device is inherently designed for excellent RF immunity. For best performance, add ground fills around all signal traces on the top and bottom PCB planes. It is advisable to follow the layout of the MAX98396 EV kit as closely as possible in the application. Thermal and performance measurements shown in this data sheet were measured with a 6-layer board ( $\theta_{JA}$  = 33°C/W). As a result, the EV kit performance is likely better than what can be achieved with a JEDEC standard board.

### **Recommended External Components**

[Table 12](#page-169-0) shows the recommended external components. See the *[Typical Application Circuits](#page-170-0)* for more details.

<span id="page-169-0"></span>

## <span id="page-170-0"></span>**Typical Application Circuits**

## **Typical Application Circuit**



## **Ordering Information**



*+Denotes a lead(Pb)-free/RoHA-compliant package.* 

*T = Tape and reel.* 

## **Revision History**



For pricing, delivery, and ordering information, please visit Maxim Integrated's online storefront at https://www.maximintegrated.com/en/storefront/storefront.html.

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