DATASHEET

PORTABLE CONSUMER CODEC

LOW-POWER, HIGH-FIDELITY INTEGRATED CODEC

TSCS42xx

DESCRIPTION

The TSCS42xx is a low-power, high-fidelity integrated CODEC with 32 bit stereo playback stereo record functionality. In addition to a high-fidelity low-power CODEC, the device integrates the true cap-less headphone amplifier.

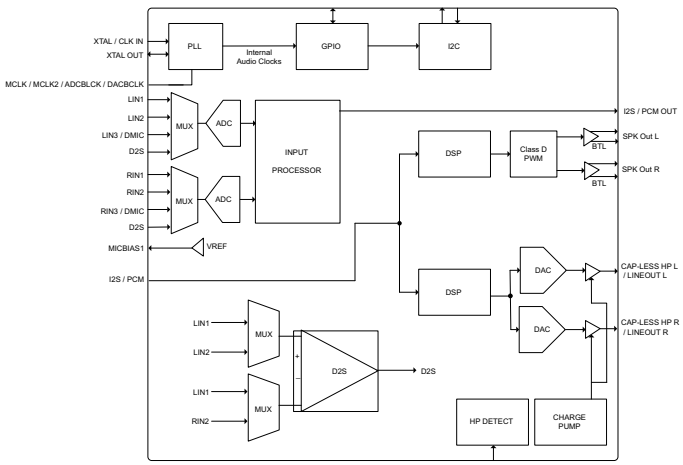
The digital audio data ports work in both master or slave modes and supports all common Linear PCM delivery formats (I2S / LJ / RJ / TDM) as well as direct Bluetooth® PCM mode.

Beyond high-fidelity for portable systems, the device offers an enriched “audio presence” through built-in audio output processing DSP engine (AOP) which is easily tunable by the designer using the Windows®-based Tempo ASC GUI.

TARGET APPLICATIONS

- Bluetooth / Wi-Fi Speakers
- Portable Navigation Devices
- Portable Gaming Devices
- Personal Media Players
- Multimedia Handsets
- Tablets

BLOCK DIAGRAM



FEATURES

- **High-Fidelity CODEC**
  - Stereo 32-bit DACs with 124dB SNR
  - Stereo 32-bit ADCs with 90dB SNR
  - Supports Sample Rates from 8kHz up to 96kHz
- **Audio Output Processing DSP Engine**
  - 3D Stereo Enhancement
  - 12-band Parametric Equalizers available per channel
  - Wideband Dynamic Range Compressor
  - Pro-Style Multi-band Compressor / Limiter / Expander
  - Psychoacoustic Bass Enhancement
  - Compressed Audio High-Frequency Restoration
- **DDX™ Digital Speaker Driver**
  - 3W/channel 4Ω (1.5W/channel 8Ω)
  - TSI DDX™ Class-D technology achieves low EMI and >90% efficiency
  - Spread spectrum support for reduced EMI
  - Constant Output power mode
  - Anti-Pop circuitry
  - Filterless architecture reduces BOM cost
- **On-chip True Capless Stereo Headphone Driver**
  - 35mW output power (16Ω), < 1% THD+N
  - 29mW output power (32Ω), < 1% THD+N
  - Charge-pump allows true ground centered outputs
  - Headphone detection logic
- **Microphone / Line-In Interface**
  - Analog microphone or line-in inputs
  - Automatic level control
  - 1 stereo DMIC
- **Low-Power with Integrated Power Management**
  - 1.7V CODEC supports 1Vrms
  - Very low standby and no-signal power consumption
  - 1.8V digital / 1.7V analog supply for low power
- **2-wire (I^2C compatible) control interface**
- **Flexible Digital Audio Data Interface**
  - Supports Bluetooth PCM
    -Adjustable Frame Length
    -Selectable Frame Sync
    - Flexible word length (16, 24, 32-bit)
  - I2S, Left-Justified, Right-Justified
    - Flexible word length (16, 24, 32-bit)
- **Package**
  - 48-Lead 7x7mm QFN

1. OVERVIEW

1.1. Block Diagram

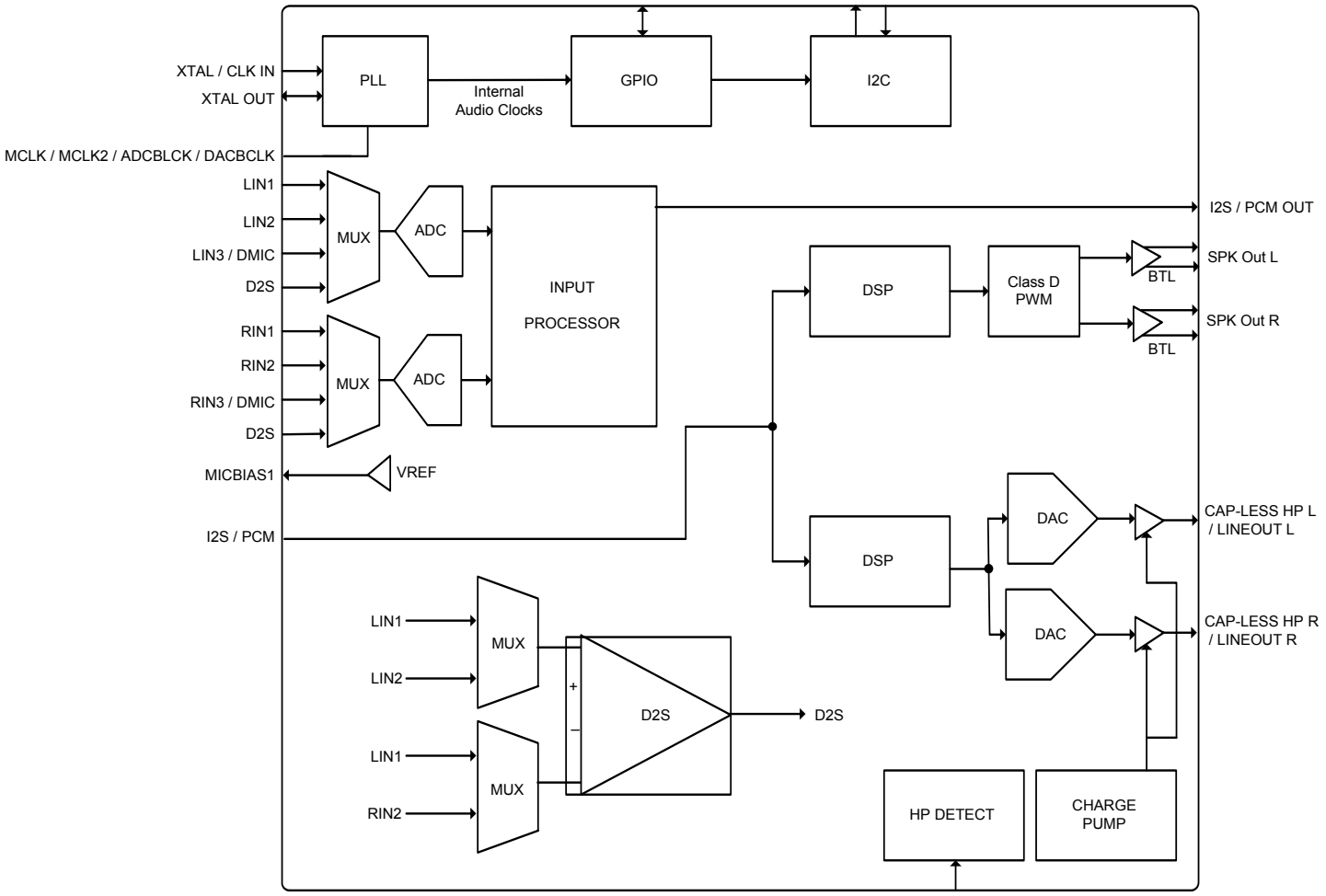


Figure 1. TSCS42xx Block Diagram

1.2. Audio Outputs

The TSCS42xx provides multiple analog audio outputs. These outputs include:

- A stereo 3W/channel (4W) or a 1.5W/channel (8W) filter-less DDX™ Class D amplifier. This amplifier is capable of driving the speakers typically found in portable equipment, providing high fidelity, high efficiency, and excellent sound quality.
- Constant output power mode maintains output volume with dropping battery supply voltage
- A line-out/cap less stereo headphone port with ground referenced outputs, capable of driving headphones without requiring an external DC blocking capacitor.

Each endpoint features independent volume controls, including a soft-mute capability which can slowly ramp up or down the volume changes to avoid unwanted audio artifacts.

The TSCS42xx output signal paths consist of digital filters, DACs and output drivers. The digital filters and DACs are enabled when the TSCS42xx is in ‘playback only’ or ‘record and playback’ mode. The output drivers can be separately enabled by individual control bits.

The digital filter and audio processing block processes the data to provide volume control and numerous sound enhancement algorithms. Two high performance sigma-delta audio DACs convert the digital data into analog.
The digital audio data is converted to over sampled bit streams using 24-bit digital interpolation filters, which then enters sigma-delta DACs, and become converted to high quality analog audio signals.

To enhance the sound available from the small, low-power speakers typically found in a portable device, the TSCS42xx provides numerous audio enhancement capabilities. The TSCS42xx features 12 independent, programmable left/right equalization, allowing the system designer to provide an advanced system equalizer to accommodate the specific speakers and enclosure design. A multiband compressor/limiter features programmable attack and release thresholds, enabling the system designer to attenuate loud noise excursions to avoid speaker artifacts, thus allowing the underlying content to be played at a louder volume without distortion. For compressed audio, a programmable expander is available to help restore the dynamic range of the original content. A stereo depth enhancement algorithm allows common left/right content (e.g. dialog) to be attenuated separately from other content, providing a perceived depth separation between background and foreground audio. Psychoacoustic bass and treble enhancement algorithms 3D sound achieve a rich, full tone even from originally compressed content, and even with speakers generally unable to play low-frequency sounds.

1.3. Audio Inputs

The TSCS42xx provides multiple digital and analog audio inputs. These inputs include:

- One digital audio input
  - supports all I2S / LJ / RJ formats in support of Linear PCM as well as direct Bluetooth PCM mode
- Three* mux selectable stereo analog line / microphone inputs with selectable differential input option *(TSCS42A1 only)
- The TSCS42A2 offers a stereo digital microphone input, in stead of a third mux selectable stereo analog line / microphone input

The device provides input gain control, separate volume controls, automatic leveling capability, and programmable microphone boost to smooth input recording. A programmable silence “floor” or “threshold” can be set to minimize background noise.

2. POWER MANAGEMENT

2.1. Control Registers

The TSCS42xx has control registers to enable system software to control which functions are active. To minimize power consumption, unused functions should be disabled. To avoid audio artifacts, it is important to enable or disable functions in the correct order.

2.1.1. Power Management Register 1

Register Address	Bit	Label	Type	Default	Description
R26(1Ah) PWRM1	7	BSTL	RW	0	Analog in Boost Left 0 = Power down 1 = Power up
	6	BSTR	RW	0	Analog in Boost Right 0 = Power down 1 = Power up
	5	PGAL	RW	0	Analog in PGA Left 0 = Power down 1 = Power up
	4	PGAR	RW	0	Analog in PGA Right 0 = Power down 1 = Power up
	3	ADCL	RW	0	ADC Left 0 = Power down 1 = Power up
	2	ADCR	RW	0	ADC Right 0 = Power down 1 = Power up
	1	MICB	RW	0	MICBIAS 0 = Power down 1 = Power up
	0	DIGENB	RW	0	Master clock disable 0: master clock enabled 1: master clock disabled

Table 1. PWRM1 Register

2.1.2. Power Management 2 Register

Register Address	Bit	Label	Type	Default	Description
R27(1Bh) PWRM2	7	D2S	RW	0	Analog in D2S AMP 0 = Power down 1 = Power up
	6	HPL	RW	0	LHP Output Buffer + DAC 0 = Power down 1 = Power up
	5	HPR	RW	0	RHP Output Buffer + DAC 0 = Power down 1 = Power up
	4	SPKL	RW	0	LSPK Output Buffer 0 = Power down 1 = Power up
	3	SPKR	RW	0	RSPK Output Buffer 0 = Power down 1 = Power up
	2	RSVD	RW	0	Reserved(bit implemented but unused)
	1	RSVD	RW	0	Reserved (bit implemented bur unused)
	0	VREF	RW	0	VREF (necessary for all other functions) 0 = Power down 1 = Power up

Table 2. PWRM2 Register

2.2. Stopping the Master Clock

In order to minimize digital core power consumption, the master clock may be stopped in Standby and OFF modes by setting the DIGENB bit (R26, bit 0).

Register Address	Bit	Label	Type	Default	Description
R26(1Ah) PWRM1	0	DIGENB	RW	0	Master clock disable 0: master clock enabled 1: master clock disabled

Table 3. Stopping the Master Clock

Note: Before DIGENB can be set, the control bits ADCL, ADCR, HPL, HPR, SPKL, and SPKR must be set to zero and a waiting time of 100ms must be observed to allow port ramping/gain fading to complete. Any failure to follow this procedure may cause pops or, if less than 1mS, may prevent the DACs and ADCs from re-starting correctly.

3. OUTPUT AUDIO PROCESSING

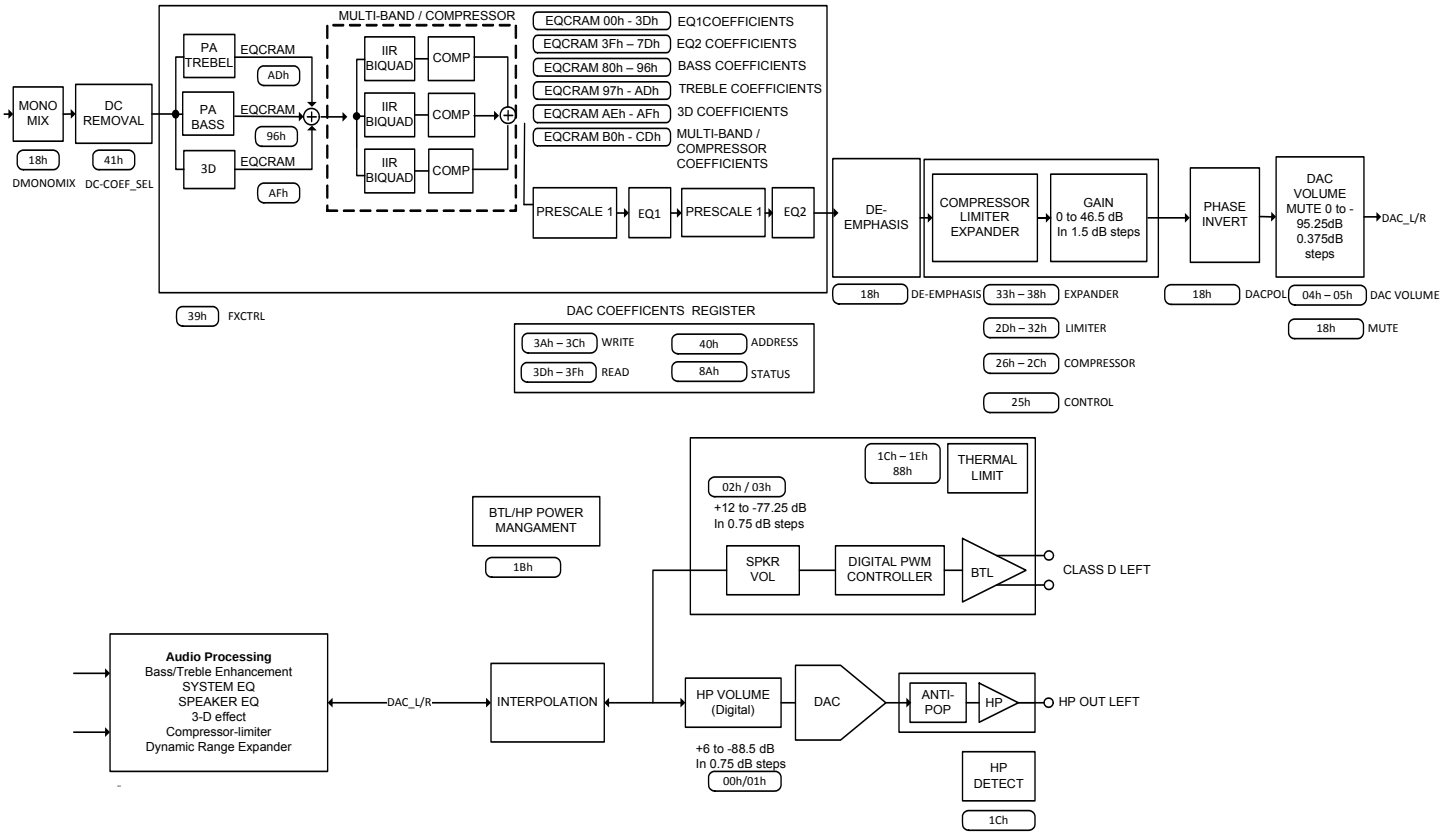


Figure 2. Output Audio Processing

3.1. DC Removal

Before processing, a DC removal filter removes the DC component from the incoming audio data. The DC removal filter is programmable, and can be bypassed by setting dc_bypass bit (R31 CONFIG0, bit1).

Register Address	Bit	Label	Type	Default	Description
R65 (41h) DCOFSEL	7:3	–	R	0	Reserved for future use.

	2:0	-	RW	5	0: dc_coef = 24'h008000; //2^^-8
					1: dc_coef = 24'h004000; //2^^-9
					2: dc_coef = 24'h002000; //2^^-10
					3: dc_coef = 24'h001000; //2^^-11
					4: dc_coef = 24'h000800; //2^^-12
					5: dc_coef = 24'h000400; //2^^-13
					6: dc_coef = 24'h000200; //2^^-14
					7: dc_coef = 24'h000100; //2^^-15

Table 4. DCOFSEL Register

Register Address	Bit	Label	Type	Default	Description
R31 (1Fh) CONFIG0	1	dc_bypass	RW	0	1 = bypass DC removal filter

Table 5. DC removal filter bypass

3.2. Volume Control

The signal volume can be controlled digitally, across a gain and attenuation range of -95.25dB to 0dB (0.375dB steps). The level of attenuation is specified by an eight-bit code, 'DACVOL_x', where 'x' is L, or R. The value "00000000" indicates mute; other values select the number of 0.375dB steps above -95.625dB for the volume level.

The Volume Update bits control the updating of volume control data; when a bit is written as '0', the Left Volume control associated with that bit is updated when ever the left volume register is written and the Right Volume control is updated when ever the right volume register is written. When a bit is written as '1', the left volume data is placed into an internal holding register when the left volume register is written and both the left and right volumes are updated when the right volume register is written. This enables a simultaneous left and right volume update.

Register Address	Bit	Label	Type	Default	Description
R4 (04h) DACVOLL	7:0	DACVOL_L [7:0]	RW	FF (0dB)	Left DAC Volume Level 0000 0000 = Digital Mute 0000 0001 = -95.25dB 0000 0010 = -94.875dB ... 0.375dB steps up to 1111 1111 = 0dB Note: If DACVOLU is set, this setting will take effect after the next write to the Right Input Volume register.
R5 (05h) DACVOLRI	7:0	DACVOL_R [7:0]	RW	FF (0dB)	Right DAC Digital Volume Level 0000 0000 = Digital Mute 0000 0001 = -95.25dB 0000 0010 = -94.875dB ... 0.375dB steps up to 1111 1111 = 0dB

Table 6. DACVOLL/DACVOLR Register

3.2.1. Volume Control Registers

Register Address	Bit	Label	Type	Default	Description
R10 (0Ah) VUCTL	7	ADCFade	RW	1	1 = volume fades between old/new value 0 = volume/mute changes immediately
	6	DACFade	RW	1	1 = volume fades between old/new value 0 = volume/mute changes immediately
	5	RSVD	R	0	Reserved for future use.
	4	INVOLU	RW	0	0 = Left input volume updated immediately 1 = Left input volume held until right input volume register written.
	3	ADCVOLU	RW	0	0 = Left ADC volume updated immediately 1 = Left ADC volume held until right ADC volume register written.
	2	DACVOLU	RW	0	0 = Left DAC volume updated immediately 1 = Left DAC volume held until right DAC volume register written.
	1	SPKVOLU	RW	0	0 = Left speaker volume updated immediately 1 = Left speaker volume held until right speaker volume register written.
	0	HPVOLU	RW	0	0 = Left headphone volume updated immediately 1 = Left headphone volume held until right headphone volume register written.

Table 7. VUCTL Register

The output path may be muted automatically when a long string of zero data is received. The length of zeros is programmable and a detection flag indicates when a stream of zero data has been detected.

Register Address	Bit	Label	Type	Default	Description
R33 (21h) GAINCTL	7	zerodet_flag	R	0	1 = zero detect length exceeded.
	6	RSVD	R	0	Reserved for future use.
	5:4	zerodetlen	RW	2	Enable mute if input consecutive zeros exceeds this length. 0 = 512, 1 = 1k, 2 = 2k, 3 = 4k samples
	3	auto_pwr	R	0	power down when mute detected
	2	auto_mute	RW	1	1 = auto mute if detect long string of zeros on input
	1	RSVD	R	0	Reserved for future use.
	0	RSVD	R	0	Reserved for future use.

Table 8. GAINCTL Register

3.3. Parametric Equalizer

The TSCS42xx has a 12-band digital parametric equalizer (a dual 6-band parametric equalizer: EQ1 and EQ2) to enable fine tuning of the audio response and preferences for a given system. Each EQ may be enabled or disabled independently. Typically one EQ will be used for speaker compensation and disabled when only headphones are in use while the other EQ is used to alter the audio to make it more pleasing to the listener.This function operates on the digital audio data before it is converted back to analog by the audio DACs.

3.3.1. Prescaler & Equalizer Filter

The dual 6-band parametric equalizer consists of a Prescaler and 6 cascaded 6-tap IIR Filters. The Prescaler allows the input to be attenuated prior to the EQ filters in case the EQ filters introduce gain, and would thus clip if not prescaled.

TSI provides a tool to enable an audio designer to determine appropriate coefficients for the equalizer filters. The filters enable the implementation of a 6-band parametric equalizer with selectable frequency bands, gain, and filter characteristics (high, low, or bandpass)

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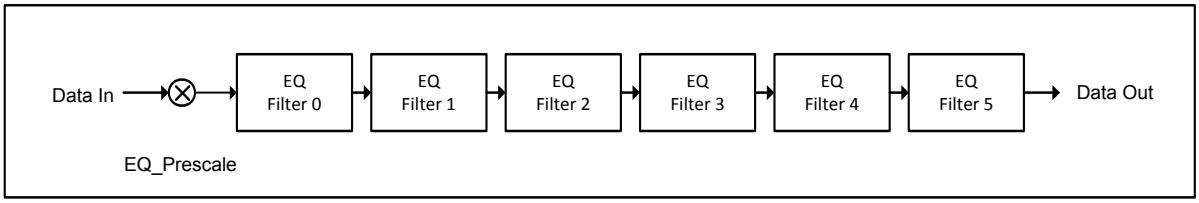

Figure 3. Prescaler & EQ Filters

The figure below shows the structure of a single EQ filter. The a(0) tap is always normalized to be equal to 1 (400000h). The remaining 5 taps are 24-bit twos compliment format programmable coefficients. (-2 < coefficient < +2)

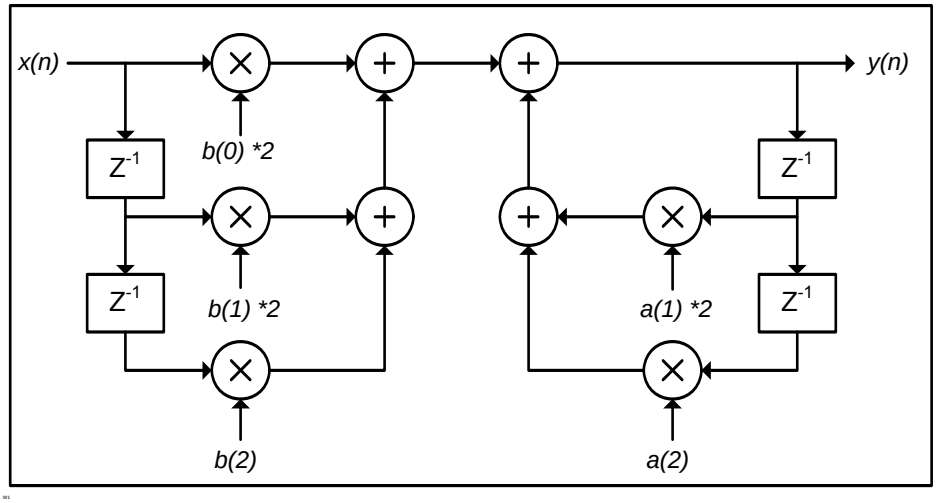

Figure 4. 6-Tap IIR Equalizer Filter

3.3.2. EQ Filter Enable Register

Register Address	Bit	Label	Type	Default	Description
R32 (20h) CONFIG1	7	EQ2_EN	R/W	0	EQ bank 2 enable 0 = second EQ bypassed 1 = second EQ enabled
	6:4	EQ2_BE[2:0]	R/W	0	EQ2 band enable. When the EQ is enabled the following EQ stages are executed. 0 - Prescale only 1 - Prescale and Filter Band 0 ... 6 - Prescale and Filter Bands 0 to 5 7 - RESERVED
	3	EQ1_EN	R/W	0	EQ bank 1 enable 0 = first EQ bypassed 1 = first EQ enabled
	2:0	EQ1_BE[2:0]	R/W	0	EQ1 band enable. When the EQ is enabled the following EQ stages are executed. 0 - Prescale only 1 - Prescale and Filter Band 0 ... 6 - Prescale and Filter Bands 0 to 5 7 - RESERVED

Table 9. CONFIG1 Registers

3.3.3. DACCRAM Write/Read Registers

Below registers provide the 24-bit data holding registers used when doing indirect writes/reads to the DAC Coefficient RAM.

3.3.3.1. DAC Coefficient Write Data Low Register

Register Address	Bit	Label	Type	Default	Description
R58 (3Ah) DACCRWRL	7:0	DACCRWD[7:0]	R/W	0	Low byte of a 24-bit data register, contains the values to be written to the DACCRAM. The address written will have be specified by the DACCRAM Address fields.

Table 10. DACCRWRL Register

3.3.3.2. DAC Coefficient Write Data Mid Register

Register Address	Bit	Label	Type	Default	Description
R59 (3Bh) DACCRWRM	7:0	DACCRWD[15:8]	R/W	0	Middle byte of a 24-bit data register, contains the values to be written to the DACCRAM. The address written will have be specified by the DACCRAM Address fields.

Table 11. DACCRWRM Register

3.3.3.3. DAC Coefficient WRITE Data High Register

Register Address	Bit	Label	Type	Default	Description
R60 (3Ch) DACCRWRH	7:0	DACCRWD[23:16]	R/W	0	High byte of a 24-bit data register, contains the values to be written to the DACCRAM. The address written will have be specified by the DACCRAM Address fields.

Table 12. DACCRWRH Register

3.3.3.4. DAC Coefficient Read Data Low Register

Register Address	Bit	Label	Type	Default	Description
R61 (3Dh) DACCRRDL	7:0	DACCRRD[7:0]	R	0	Low byte of a 24-bit data register, contains the contents of the most recent DACCRAM address read from the RAM. The address read will have been specified by the DACCRAM Address fields.

Table 13. DACCRRDL Register

3.3.3.5. DAC Coefficient Read Data Mid Register

Register Address	Bit	Label	Type	Default	Description
R62 (3Eh) DACCRRDM	7:0	DACCRRD[15:8]	R	0	Middle byte of a 24-bit data register, contains the contents of the most recent DACCRAM address read from the RAM. The address read will have been specified by the DACCRAM Address fields.

Table 14. DACCRRDM Register

3.3.3.6. DAC Coefficient Read Data High Register

Register Address	Bit	Label	Type	Default	Description
R63 (3Fh) DACCRRDH	7:0	DACCRRD[23:16]	R	0	High byte of a 24-bit data register, contains the contents of the most recent DACCRAM address read from the RAM. The address read will have been specified by the DACCRAM Address fields.

Table 15. DACCRRDH Register

3.3.4. DACCRAM Address Register

This 8-bit register provides the address to the internal RAM when doing indirect writes/reads to the DAC Coefficient RAM

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Register Address	Bit	Label	Type	Default	Description
R64 (40h) DACCRADDR	7:0	DACCRADD	R/W	0	Contains the address (between 0 and 255) of the DACCRAM to be accessed by a read or write. This is not a byte address--it is the address of the 24-bit data item to be accessed from the DACCRAM.This address is automatically incremented after writing to DACCRAM_WRITE_HI or reading from DACCRAM_READ_HI (and the 24 bit data from the next RAM location is fetched.)

Table 16. DACCRADDR Register

3.3.5. DACCRAM STATUS Register

This control register provides the write/read enable when doing indirect writes/reads to the DAC Coefficient RAM.

Register Address	Bit	Label	Type	Default	Description
R138 (8Ah) DACCRSTAT	7	DACCRAM_Busy	R	0	1 = read/write to DACCRAM in progress, cleared by HW when done.
	6:0	RSVD	R	0	Reserved

Table 17. DACCRSTAT Register

3.3.6. Equalizer, Bass, Treble Coefficient & Equalizer Prescaler RAM

The DAC Coefficient RAM is a single port 176x24 synchronous RAM. It is programmed indirectly through the Control Bus in the following manner as shown in the figure below:

1 Write target address to DACCRAM_ADDR register. (DAC Coefficient data is pre-fetched even if we don't use it)
   - **a** Start command followed by the Device Address and Write flag
   - **b** Register Address (DACCRAM_ADDR register address)
   - **c** Register Data (DACCRAM address to be held in DACCRAM_ADDR)

2 Start a multiple write cycle
   - **a** Start command followed by the Device Address and Write Flag
   - **b** Register Address of the DACCRAM_WRITE_LO register
   - **c** Write D7:0 to the DACCRAM_WRITE_LO register
   - **d** Write D15:8 to the DACCRAM_WRITE_MID register
   - **e** Write D23:16 to the DACCRAM_WRITE_HI register

3 On successful receipt of the DACCRAM_WRITE_HI data, the part will automatically start a write cycle. The DACCRAM_Busy bit will be set high to indicate that a write is in progress.

4 On completion of the internal write cycle, the DACCRAM_Busy bit will be 0 (when operating the control interface at high speeds - TBD - software must poll this bit to ensure the write cycle is complete before starting another write cycle.)

5 The bus cycle may be terminated by the host or steps 2-3 may be repeated for writes to consecutive DAC Coefficient RAM locations.

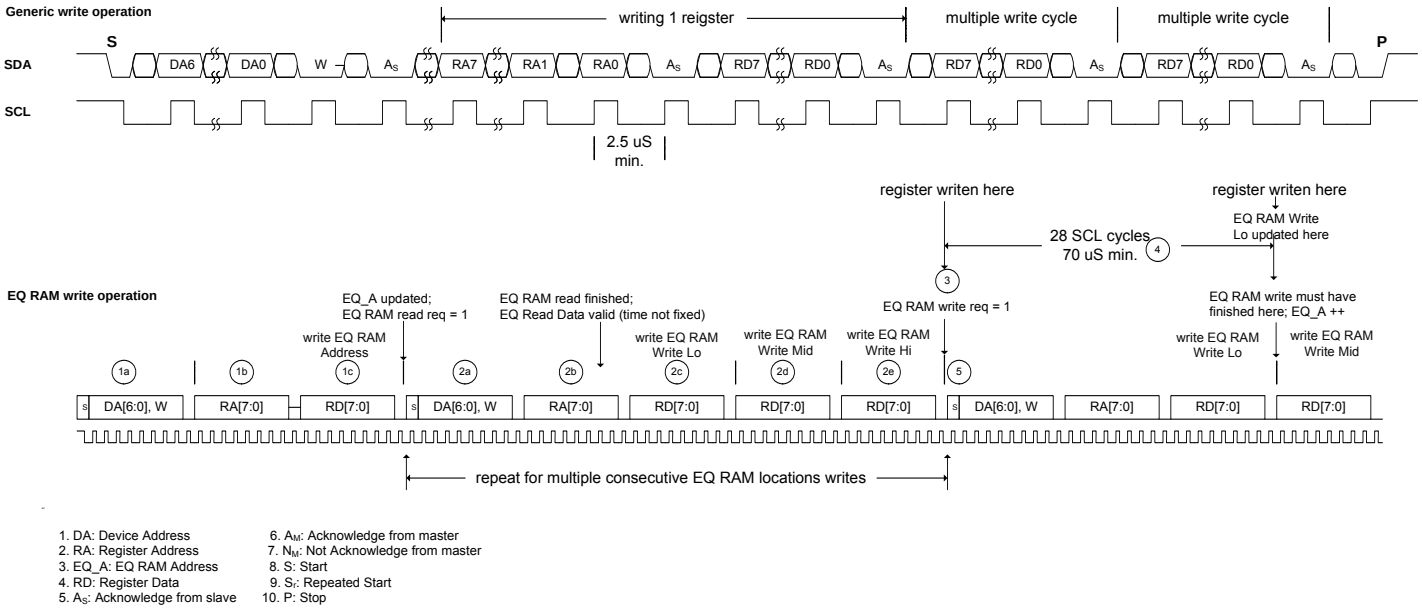

Figure 5. DAC Coefficient RAM Write Sequence

Reading back a value from the DACCRAM is done in this manner:

1 Write target address to DACCRAM_ADDR register.(DAC Coefficient data is pre-fetched for read even if we don't use it)
   a Start command followed by the Device Address and Write flag
   b Register Address (DACCRAM_ADDR register address)
   c Register Data (DACCRAM address to be held in DACCRAM_ADDR)
2 Start (or repeat start) a write cycle to DACCRAM_READ_LO and after the second byte (register address) is acknowledged, go to step 3. (Do not complete the write cycle.)
   a Start command followed by the Device Address and Write Flag
   b Register Address of the DACCRAM_READ_LO register
3 Signal a repeat start, provide the device address, and indicate a read operation
4 Read D7:0 (register address incremented after ack by host)
5 Read D15:8 (register address incremented after ack by host)
6 Read D23:16 (register address incremented and next DAC Coefficient location pre-fetched after ack by host)
7 The host stops the bus cycle

To repeat a read cycle for consecutive DAC Coefficient RAM locations:

8 Start (or repeat start instead of stopping the bus cycle in step 7) a write cycle indicating DACCRAM_RD_LO as the target address.
9 After the second byte is acknowledged, signal a repeated start.
10 Indicate a read operation
11 Read the DACCRAM_READ_LO register as described in step 4

12 Read the DACCRAM_READ_MID register as described in step 5

13 Read the DACCRAM_READ_HI register as described in step 6

14 Repeat steps 8-13 as desired

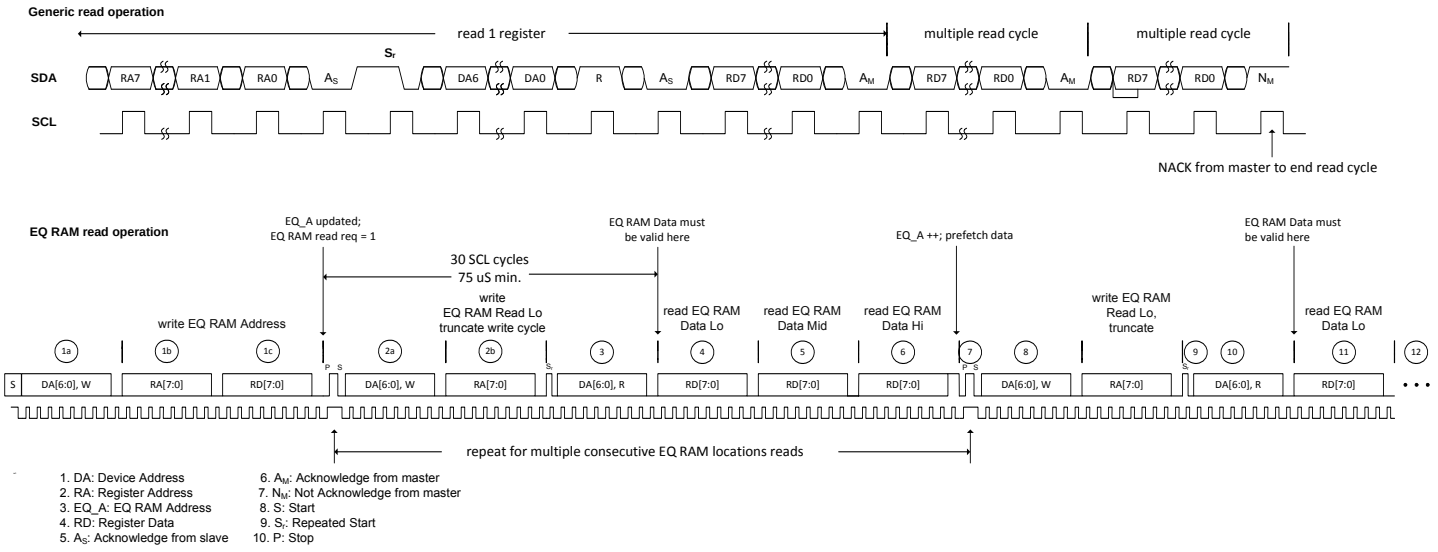

Figure 6. DAC Coefficient RAM Read Sequence

Table 18: DACCRAM EQ Addresses

EQ 1					EQ2				
Addr	**Channel 0 Coefficients**		**Addr**	**Channel 1 Coefficients**	**Addr**	**Channel 0 Coefficients**		**Addr**	**Channel 1 Coefficients**
0x00	EQ_COEF_0F0_B0		0x20	EQ_COEF_1F0_B0	0x40	EQ_COEF_2F0_B0		0x60	EQ_COEF_3F0_B0
0x01	EQ_COEF_0F0_B1		0x21	EQ_COEF_1F0_B1	0x41	EQ_COEF_2F0_B1		0x61	EQ_COEF_3F0_B1
0x02	EQ_COEF_0F0_B2		0x22	EQ_COEF_1F0_B2	0x42	EQ_COEF_2F0_B2		0x62	EQ_COEF_3F0_B2
0x03	EQ_COEF_0F0_A1		0x23	EQ_COEF_1F0_A1	0x43	EQ_COEF_2F0_A1		0x63	EQ_COEF_3F0_A1
0x04	EQ_COEF_0F0_A2		0x24	EQ_COEF_1F0_A2	0x44	EQ_COEF_2F0_A2		0x64	EQ_COEF_3F0_A2
0x05	EQ_COEF_0F1_B0		0x25	EQ_COEF_1F1_B0	0x45	EQ_COEF_2F1_B0		0x65	EQ_COEF_3F1_B0
0x06	EQ_COEF_0F1_B1		0x26	EQ_COEF_1F1_B1	0x46	EQ_COEF_2F1_B1		0x66	EQ_COEF_3F1_B1
0x07	EQ_COEF_0F1_B2		0x27	EQ_COEF_1F1_B2	0x47	EQ_COEF_2F1_B2		0x67	EQ_COEF_3F1_B2
0x08	EQ_COEF_0F1_A1		0x28	EQ_COEF_1F1_A1	0x48	EQ_COEF_2F1_A1		0x68	EQ_COEF_3F1_A1
0x09	EQ_COEF_0F1_A2		0x29	EQ_COEF_1F1_A2	0x49	EQ_COEF_2F1_A2		0x69	EQ_COEF_3F1_A2
0x0A	EQ_COEF_0F2_B0		0x2A	EQ_COEF_1F2_B0	0x4A	EQ_COEF_2F2_B0		0x6A	EQ_COEF_3F2_B0
0x0B	EQ_COEF_0F2_B1		0x2B	EQ_COEF_1F2_B1	0x4B	EQ_COEF_2F2_B1		0x6B	EQ_COEF_3F2_B1
0x0C	EQ_COEF_0F2_B2		0x2C	EQ_COEF_1F2_B2	0x4C	EQ_COEF_2F2_B2		0x6C	EQ_COEF_3F2_B2
0x0D	EQ_COEF_0F2_A1		0x2D	EQ_COEF_1F2_A1	0x4D	EQ_COEF_2F2_A1		0x6D	EQ_COEF_3F2_A1
0x0E	EQ_COEF_0F2_A2		0x2E	EQ_COEF_1F2_A2	0x4E	EQ_COEF_2F2_A2		0x6E	EQ_COEF_3F2_A2
0x0F	EQ_COEF_0F3_B0		0x2F	EQ_COEF_1F3_B0	0x4F	EQ_COEF_2F3_B0		0x6F	EQ_COEF_3F3_B0
0x10	EQ_COEF_0F3_B1		0x30	EQ_COEF_1F3_B1	0x50	EQ_COEF_2F3_B1		0x70	EQ_COEF_3F3_B1
0x11	EQ_COEF_0F3_B2		0x31	EQ_COEF_1F3_B2	0x51	EQ_COEF_2F3_B2		0x71	EQ_COEF_3F3_B2
0x12	EQ_COEF_0F3_A1		0x32	EQ_COEF_1F3_A1	0x52	EQ_COEF_2F3_A1		0x72	EQ_COEF_3F3_A1
0x13	EQ_COEF_0F3_A2		0x33	EQ_COEF_1F3_A2	0x53	EQ_COEF_2F3_A2		0x73	EQ_COEF_3F3_A2
0x14	EQ_COEF_0F4_B0		0x34	EQ_COEF_1F4_B0	0x54	EQ_COEF_2F4_B0		0x74	EQ_COEF_3F4_B0
0x15	EQ_COEF_0F4_B1		0x35	EQ_COEF_1F4_B1	0x55	EQ_COEF_2F4_B1		0x75	EQ_COEF_3F4_B1
0x16	EQ_COEF_0F4_B2		0x36	EQ_COEF_1F4_B2	0x56	EQ_COEF_2F4_B2		0x76	EQ_COEF_3F4_B2
0x17	EQ_COEF_0F4_A1		0x37	EQ_COEF_1F4_A1	0x57	EQ_COEF_2F4_A1		0x77	EQ_COEF_3F4_A1
0x18	EQ_COEF_0F4_A2		0x38	EQ_COEF_1F4_A2	0x58	EQ_COEF_2F4_A2		0x78	EQ_COEF_3F4_A2
0x19	EQ_COEF_0F5_B0		0x39	EQ_COEF_1F5_B0	0x59	EQ_COEF_2F5_B0		0x79	EQ_COEF_3F5_B0
0x1A	EQ_COEF_0F5_B1		0x3A	EQ_COEF_1F5_B1	0x5A	EQ_COEF_2F5_B1		0x7A	EQ_COEF_3F5_B1
0x1B	EQ_COEF_0F5_B2		0x3B	EQ_COEF_1F5_B2	0x5B	EQ_COEF_2F5_B2		0x7B	EQ_COEF_3F5_B2
0x1C	EQ_COEF_0F5_A1		0x3C	EQ_COEF_1F5_A1	0x5C	EQ_COEF_2F5_A1		0x7C	EQ_COEF_3F5_A1
0x1D	EQ_COEF_0F5_A2		0x3D	EQ_COEF_1F5_A2	0x5D	EQ_COEF_2F5_A2		0x7D	EQ_COEF_3F5_A2
0x1E	-		0x3E	-	0x5E	-		0x7E	-
0x1F	EQ_PRESCALE0		0x3F	EQ_PRESCALE1	0x5F	EQ_PRESCALE2		0x7F	EQ_PRESCALE3

Table 19: DACCRAM Bass/Treble/3D and multiband compressor Addresses

Addr	Bass	Addr	Treble	Addr	3D	Addr	Multiband
	Coefficients		Coefficients		Coefficients		Coefficients
0x80	BASS_COEF_EXT1_B0	0x97	TREB_COEF_EXT1_B0	0xAE	3D_COEF	0xB0	MBC1_BQ1_COEFF0
0x81	BASS_COEF_EXT1_B1	0x98	TREB_COEF_EXT1_B1	0xAF	3D_MIX	0xB1	MBC1_BQ1_COEFF1
0x82	BASS_COEF_EXT1_B2	0x99	TREB_COEF_EXT1_B2			0xB2	MBC1_BQ1_COEFF2
0x83	BASS_COEF_EXT1_A1	0x9A	TREB_COEF_EXT1_A1			0xB3	MBC1_BQ1_COEFF3
0x84	BASS_COEF_EXT1_A2	0x9B	TREB_COEF_EXT1_A2			0xB4	MBC1_BQ1_COEFF4
0x85	BASS_COEF_EXT2_B0	0x9C	TREB_COEF_EXT2_B0			0xB5	MBC1_BQ2_COEFF0
0x86	BASS_COEF_EXT2_B1	0x9D	TREB_COEF_EXT2_B1			0xB6	MBC1_BQ2_COEFF1
0x87	BASS_COEF_EXT2_B2	0x9E	TREB_COEF_EXT2_B2			0xB7	MBC1_BQ2_COEFF2
0x88	BASS_COEF_EXT2_A1	0x9F	TREB_COEF_EXT2_A1			0xB8	MBC1_BQ2_COEFF3
0x89	BASS_COEF_EXT2_A2	0xA0	TREB_COEF_EXT2_A2			0xB9	MBC1_BQ2_COEFF4
0x8A	BASS_COEF_NLF_M1	0xA1	TREB_COEF_NLF_M1			0xBA	MBC2_BQ1_COEFF0
0x8B	BASS_COEF_NLF_M2	0xA2	TREB_COEF_NLF_M2			0xBB	MBC2_BQ1_COEFF1
0x8C	BASS_COEF_LMT_B0	0xA3	TREB_COEF_LMT_B0			0xBC	MBC2_BQ1_COEFF2
0x8D	BASS_COEF_LMT_B1	0xA4	TREB_COEF_LMT_B1			0xBD	MBC2_BQ1_COEFF3
0x8E	BASS_COEF_LMT_B2	0xA5	TREB_COEF_LMT_B2			0xBE	MBC2_BQ1_COEFF4
0x8F	BASS_COEF_LMT_A1	0xA6	TREB_COEF_LMT_A1			0xBF	MBC2_BQ2_COEFF0
0x90	BASS_COEF_LMT_A2	0xA7	TREB_COEF_LMT_A2			0xC0	MBC2_BQ2_COEFF1
0x91	BASS_COEF_CTO_B0	0xA8	TREB_COEF_CTO_B0			0xC1	MBC2_BQ2_COEFF2
0x92	BASS_COEF_CTO_B1	0xA9	TREB_COEF_CTO_B1			0xC2	MBC2_BQ2_COEFF3
0x93	BASS_COEF_CTO_B2	0xAA	TREB_COEF_CTO_B2			0xC3	MBC2_BQ2_COEFF4
0x94	BASS_COEF_CTO_A1	0xAB	TREB_COEF_CTO_A1			0xC4	MBC3_BQ1_COEFF0
0x95	BASS_COEF_CTO_A2	0xAC	TREB_COEF_CTO_A2			0xC5	MBC3_BQ1_COEFF1
0x96	BASS_MIX	0xAD	TREB_MIX			0xC6	MBC3_BQ1_COEFF2
						0xC7	MBC3_BQ1_COEFF3
						0xC8	MBC3_BQ1_COEFF4
						0xC9	MBC3_BQ2_COEFF0
						0xCA	MBC3_BQ2_COEFF1
						0xCB	MBC3_BQ2_COEFF2
						0xCC	MBC3_BQ2_COEFF3
						0xCD	MBC3_BQ2_COEFF4

1. All B0 coefficients are set to unity (400000h) by default. All others, including M1 and M2, are 0 by default.
2. NLF coefficients (M1, M2) have a range defined as +/-8, with 1 sign bit, 3 integer bits, and 20 fraction bits. So, unity for these values is 100000h. This is as opposed to the rest of the coefficient RAM, which has a range defined as +/-2, with 1 sign bit, 1 integer bit, and 22 fraction bits.

3.4. Gain and Dynamic Range Control

The gain for a given channel is controlled by the DACVOL, HPVOL, SPKVOL registers. If the result of the gain multiply step would result in overflow of the output word width, the output is saturated at the max positive or negative value. In addition to simple gain control, the TSCS42xx also provides sophisticated dynamic range control including limiting, dynamic range compression, and dynamic range expansion functions.

3.5. Multi-band Compressor

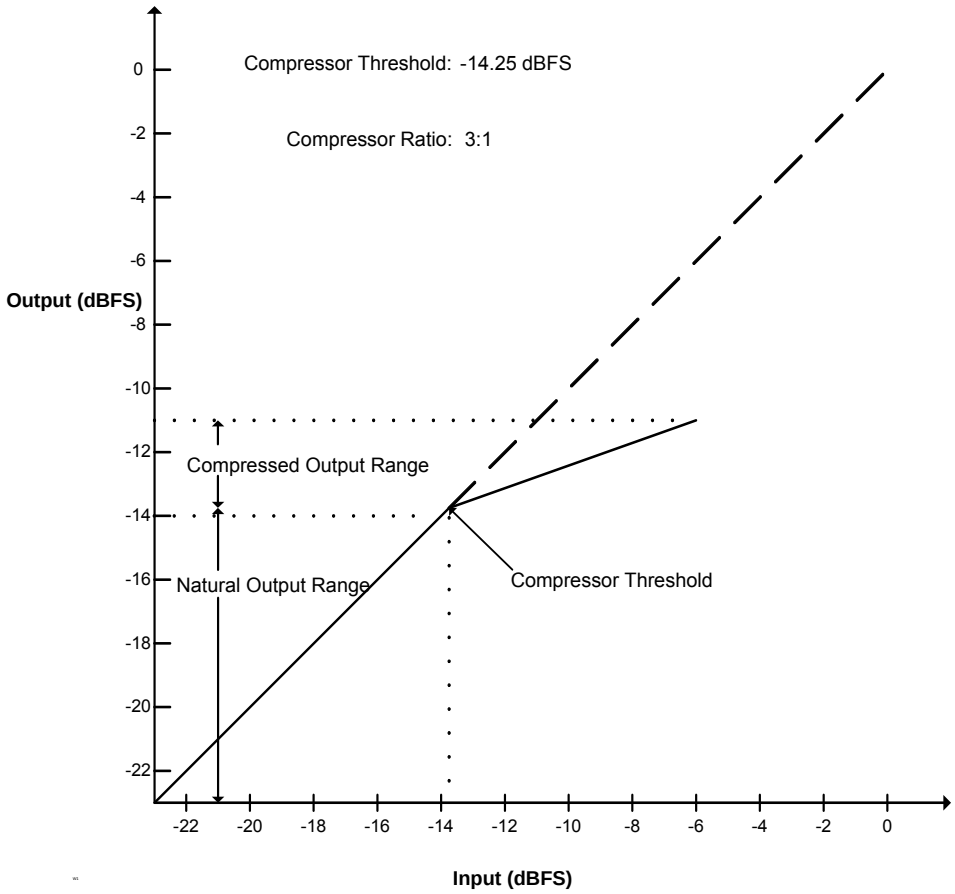

Figure 7. Gain Compressor, Output vs Input

3.5.1. Overview

The TSCS42xx output processing includes a multi-band compressor that improves sound from small loudspeakers typically used in portable devices. Three independent compressor blocks are each preceded by a Bi-quad processing block that filters the incoming audio so that each compressor operates on a select range of audio frequencies. The advantage of multiband compression over full-bandwidth (full-band, or single-band) compression is that audible gain “pumping” can be reduced. When using single band compressors high energy audio content in a narrow range of frequencies can cause the volume of the entire audio frequency band to be affected thus causing the audio signal level to audibly “pump”. This pumping of the audio signal level can be distracting. A multi-band compressor can effectively eliminate or reduce the pumping to insignificant levels.

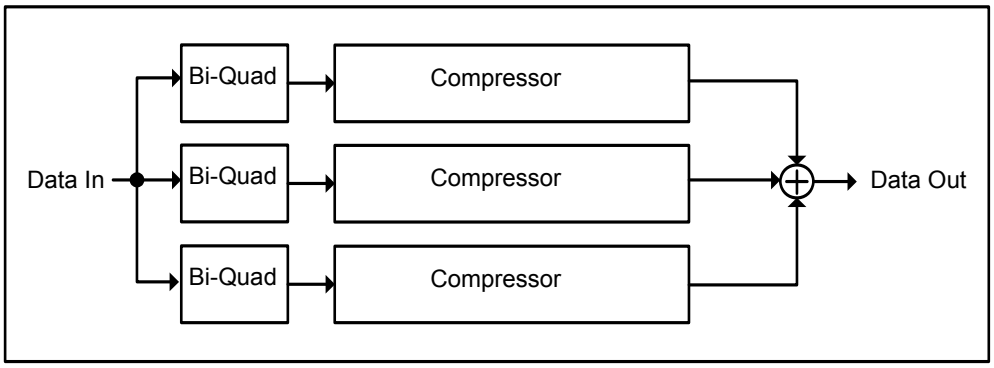

Figure 8. Block Diagram Multiband Compressor

Each band in the Multi-band Compressor is comprised of a single stage 6-tap IIR (Bi-quad) filter followed by a compressor block. The BI-quad filter coefficients are written using the Parametric Equalizer Registers. The purpose of the Bi-quad block is to provide a bandpass filter function for each Compressor band.

A basic block diagram of the compressor is shown below:

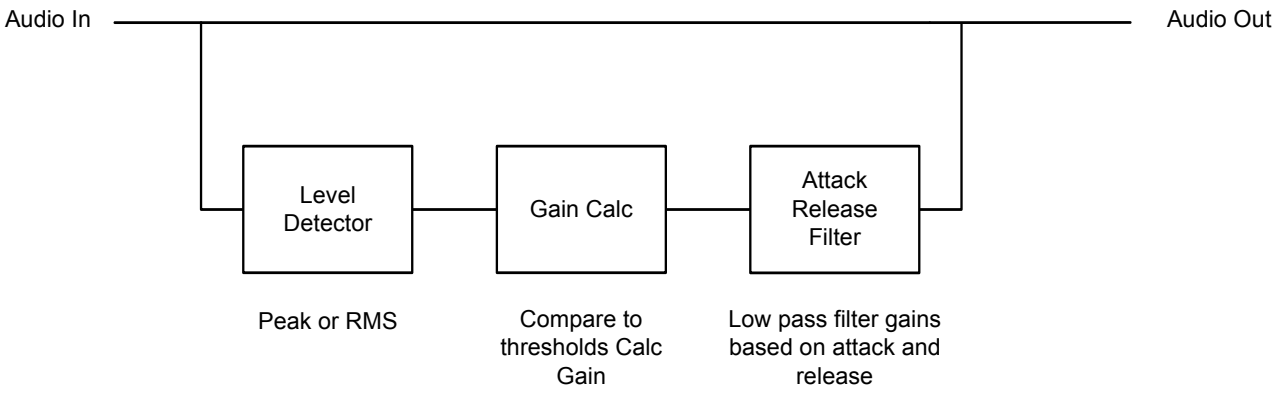

Figure 9. Compressor Block Diagram

As this diagram shows, there are 3 primary components of the compressor.

1. **Level Detector:** The level detector, detects the level of the incoming signal. Since the comp/limiter is designed to work on blocks of signals, the level detector will either find the peak value of the block of samples to be processed or the rms level of the samples within a block.
2. **Gain Calculation:** The gain calculation block is responsible for taking the output of the level detector and calculating a target gain based on that level and the compressor and expander Compression region gain calculation: In the compression region, the gain calculation is:

Atten(in db) = (1-1/ratio)(threshold(in db) – level(in db);

- For example,
  - Ratio = 4:1 compression
  - Threshold = -16db
  - Level = -4 db

The required attenuation is: 9db or a gain coefficient of 0.1259.

Translating this calculation from log space to linear yields the formula:

$Gain = (level/threshold)^{1/ratio} * (threshold/level)$

- State Transitions: In addition to calculating the new gain for the compressor, the gain calculation block will also select the filter coefficient for the attack/release filter. The rules for selecting the coefficient are as follows:

In the compression region:
- If the gain calculated is less than the last gain calculated (more compression is being applied), then the filter coefficient is the compressor attack.
- If the gain calculated is more than the last gain calculated (less compression), the filter coefficient is the compressor release.

In the linear region:
- Modify gain until a gain of 1.0 is obtained, using the compressor release.

3. **Attack/Release filter:** In order to prevent objectionable artifacts, the gain is smoothly ramped from the current value to the new value calculated by the gain calculation block. In the PC-based comp/limiter, this is achieved using a simple tracking lowpass filter to smooth out the abrupt transitions. The calculation (using the coefficient (coeff) selected by the gain block) is:

Filtered_gain = coeff*last_filtered_gain + (1.0 - coeff)*target_gain;

This creates a exponential ramp from the current gain value to the new value.

3.5.2. Multi band Compressor Registers

Register Address	Bit	Label	Type	Default	Description
Reg 199 (C7h) DACMBCEN	7:3	RSVD	R	0h	Reserved
	2	MBCEN3	RW	0	1 = enable compressor band 3
	1	MBCEN2	RW	0	1 = enable compressor band 2
	0	MBCEN1	RW	0	1 = enable compressor band 1

Table 20. DACMBCEN Register

Register Address	Bit	Label	Type	Default	Description
Reg 200 (C8h) DACMBCCTL	7:6	RSVD	R	0h	Reserved
	5	LVLMODE3	RW	0	Compressor Level Detection Mode Band 3 0 = Average 1 = Peak
	4	WINSEL3	RW	0	Window width selection for level detection Band 3 0 = equivalent of 512 samples of selected Base Rate (~10-16ms) 1 = equivalent of 64 samples of selected Base Rate (~1.3-2ms)
	3	LVLMODE2	RW	0	Compressor Level Detection Mode Band 2 0 = Average 1 = Peak
	2	WINSEL2	RW	0	Window width selection for level detection Band 2 0 = equivalent of 512 samples of selected Base Rate (~10-16ms) 1 = equivalent of 64 samples of selected Base Rate (~1.3-2ms)
	1	LVLMODE1	RW	0	Compressor Level Detection Mode Band 1 0 = Average 1 = Peak
	0	WINSEL1	RW	0	Window width selection for level detection Band1 0 = equivalent of 512 samples of selected Base Rate (~10-16ms) 1 = equivalent of 64 samples of selected Base Rate (~1.3-2ms)

Table 21. DACMBCCTL Register

Register Address	Bit	Label	Type	Default	Description
Reg 201(C9h) DACMBCMUG1	7:5	RSVD	R	0h	Reserved
	5	PHASE	RW	0h	0 = not inverted 1 = Inverted
	4:0	MUGAIN[4:0]	RW	0h	0dB...46.5dB in 1.5dB steps

Table 22. DACMBCMUG1 Register

Register Address	Bit	Label	Type	Default	Description
Reg 202(CAh) DACMBCTHR1	7:0	THRESH[7:0]	RW	00h	FFh...00h = 0dB...95.625dB in 0.375dB steps.

Table 23. DACMBCTHR1 Register

Register Address	Bit	Label	Type	Default	Description
Reg 203(CBh) DACMBCRAT1	7:5	RSVD	R	000	Reserved
	4:0	RATIO[4:0]	RW	00h	Compressor Ratio 00h = Reserved 01h = 1.5:1 02h...14h = 2:1...20:1 15h...1Fh = Reserved

Table 24. DACMBCRAT1 Register

Register Address	Bit	Label	Type	Default	Description
Reg 204(CCh) DACMBCATK1L	7:0	TCATKL	RW	0h	Compressor Attack Time Constant, Low Byte

Table 25. DACMBCATK1L Register

Register Address	Bit	Label	Type	Default	Description
Reg 205(CDh) DACMBCATK1H	7:0	TCATKH[7:0]	RW	00h	High byte of the time constant used to ramp to a new gain value during a compressor attack phase. 0000h = 0 (instantaneous) 0001h = 0.96875 + 1/(2^21) 0002h = 0.96875 + 2/(2^21) ... (step = 1/(2^21)) FFFEh = [(2^21)-2]/(2^21) FFFFh = [(2^21)-1]/(2^21)

Table 26. DACMBCATK1H Register

Register Address	Bit	Label	Type	Default	Description
Reg 206(CEh) DACMBCREL1L	7:0	TCRELL[7:0]	RW	00h	Low byte of the time constant used to ramp to a new gain value during a compressor release phase. 0000h = 0 (instantaneous) 0001h = 0.96875 + 1/(2^21) 0002h = 0.96875 + 2/(2^21) ... (step = 1/(2^21)) FFFEh = [(2^21)-2]/(2^21) FFFFh = [(2^21)-1]/(2^21)

Table 27. DACMBCREL1L Register

Register Address	Bit	Label	Type	Default	Description
Reg 207(CFh) DACMBCREL1H	7:0	TCRELH[15:8]	RW	00h	High byte of the time constant used to ramp to a new gain value during a compressor release phase. The time constant is [high byte, low byte] 0000h = 0 (instantaneous) 0001h = 0.96875 + 1/(2^21) 0002h = 0.96875 + 2/(2^21) ... (step = 1/(2^21)) FFFEh = [(2^21)-2]/(2^21) FFFFh = [(2^21)-1]/(2^21)

Table 28. DACMBCREL1H Register

Register Address	Bit	Label	Type	Default	Description
Reg 208(D0h) DACMBCMUG2	7:6	RSVD	R	0h	Reserved
	5	PHASE	RW	0h	0 = not inverted 1 = Inverted
	4:0	MUGAIN[4:0]	RW	0h	0dB...46.5dB in 1.5dB steps

Table 29. DACMBCMUG2 Register

Register Address	Bit	Label	Type	Default	Description
Reg 209(D1h) DACMBCTHR2	7:0	THRESH[7:0]	RW	00h	FFh...00h = 0dB...95.625dB in 0.375dB steps.

Table 30. DACMBCTHR2 Register

Register Address	Bit	Label	Type	Default	Description
Reg 210(D2h) DACMBCRAT2	7:5	RSVD	R	000	Reserved
	4:0	RATIO[4:0]	RW	00h	Compressor Ratio 00h = Reserved 01h = 1.5:1 02h...14h = 2:1...20:1 15h...1Fh = Reserved

Table 31. DACMBCRAT2 Register

Register Address	Bit	Label	Type	Default	Description
Reg 211(D3h) DACMBCATK2L	7:0	TCATKL[7:0]	RW	00h	Low byte of the time constant used to ramp to a new gain value during a compressor attack phase. 0000h = 0 (instantaneous) 0001h = 0.96875 + 1/(2^21) 0002h = 0.96875 + 2/(2^21) ... (step = 1/(2^21)) FFFEh = [(2^21)-2]/(2^21) FFFFh = [(2^21)-1]/(2^21)

Table 32. DACMBCATK2L Register

Register Address	Bit	Label	Type	Default	Description
Reg 212(D4h) DACMBCATK2H	7:0	TCATKH[7:0]	RW	00h	High byte of the time constant used to ramp to a new gain value during a compressor attack phase. 0000h = 0 (instantaneous) 0001h = 0.96875 + 1/(2^21) 0002h = 0.96875 + 2/(2^21) ... (step = 1/(2^21)) FFFEh = [(2^21)-2]/(2^21) FFFFh = [(2^21)-1]/(2^21)

Table 33. DACMBCATK2H Register

Register Address	Bit	Label	Type	Default	Description
Reg 213(D5h) DACMBCREL2L	7:0	TCRELL[7:0]	RW	00h	Low byte of the time constant used to ramp to a new gain value during a compressor release phase. 0000h = 0 (instantaneous) 0001h = 0.96875 + 1/(2^21) 0002h = 0.96875 + 2/(2^21) ... (step = 1/(2^21)) FFFEh = [(2^21)-2]/(2^21) FFFFh = [(2^21)-1]/(2^21)

Table 34. DACMBCREL2L Register

Register Address	Bit	Label	Type	Default	Description
Reg 214(D6h) DACMBCREL2H	7:0	TCREL[15:8]	RW	00h	High byte of the time constant used to ramp to a new gain value during a compressor release phase. The time constant is [high byte, low byte] 0000h = 0 (instantaneous) 0001h = 0.96875 + 1/(2^21) 0002h = 0.96875 + 2/(2^21) ... (step = 1/(2^21)) FFFEh = [(2^21)-2]/(2^21) FFFFh = [(2^21)-1]/(2^21)

Table 35. DACMBCREL2H Register

Register Address	Bit	Label	Type	Default	Description
Reg 215(D7h) DACMBCMUG3	7:5	RSVD	R	0h	Reserved
	5	PHASE	RW	0h	0 = not inverted 1 = Inverted
	4:0	MUGAIN[4:0]	RW	0h	0dB...46.5dB in 1.5dB steps

Table 36. DACMBCMUG3 Register

Register Address	Bit	Label	Type	Default	Description
Reg 216(D8h) DACMBCTHR3	7:0	THRESH[7:0]	RW	00h	FFh...00h = 0dB...95.625dB in 0.375dB steps.

Table 37. DACMBCTHR3 Register

Register Address	Bit	Label	Type	Default	Description
Reg 217(D9h) DACMBCRAT3	7:5	RSVD	R	000	Reserved
	4:0	RATIO[4:0]	RW	00h	Compressor Ratio 00h = Reserved 01h = 1.5:1 02h...14h = 2:1...20:1 15h...1Fh = Reserved

Table 38. DACMBCRAT3 Register

Register Address	Bit	Label	Type	Default	Description
Reg 218(DAh) DACMBCATK3L	7:0	TCATKL[7:0]	RW	00h	Low byte of the time constant used to ramp to a new gain value during a compressor attack phase. 0000h = 0 (instantaneous) 0001h = 0.96875 + 1/(2^21) 0002h = 0.96875 + 2/(2^21) ... (step = 1/(2^21)) FFFEh = [(2^21)-2]/(2^21) FFFFh = [(2^21)-1]/(2^21)

Table 39. DACMBCATK3L Register

Register Address	Bit	Label	Type	Default	Description
Reg 219(DBh) DACMBCATK3H	7:0	TCATKHH[7:0]	RW	00h	High byte of the time constant used to ramp to a new gain value during a compressor attack phase. 0000h = 0 (instantaneous) 0001h = 0.96875 + 1/(2^21) 0002h = 0.96875 + 2/(2^21) ... (step = 1/(2^21)) FFFEh = [(2^21)-2]/(2^21) FFFFh = [(2^21)-1]/(2^21)

Table 40. DACMBCATK3H Register

Register Address	Bit	Label	Type	Default	Description
Reg 220(DCh) DACMBCREL3L	7:0	TCRELL[7:0]	RW	00h	Low byte of the time constant used to ramp to a new gain value during a compressor release phase. 0000h = 0 (instantaneous) 0001h = 0.96875 + 1/(2^21) 0002h = 0.96875 + 2/(2^21) ... (step = 1/(2^21)) FFFEh = [(2^21)-2]/(2^21) FFFFh = [(2^21)-1]/(2^21)

Table 41. DACMBCREL3L Register

Register Address	Bit	Label	Type	Default	Description
Reg 221(DDh) DACMBCREL3H	7:0	TCRELH[15:8]	RW	00h	High byte of the time constant used to ramp to a new gain value during a compressor release phase. The time constant is [high byte, low byte] 0000h = 0 (instantaneous) 0001h = 0.96875 + 1/(2^21) 0002h = 0.96875 + 2/(2^21) ... (step = 1/(2^21)) FFFEh = [(2^21)-2]/(2^21) FFFFh = [(2^21)-1]/(2^21)

Table 42. DACMBCRELL3H Register

3.6. Limiter/Compressor/Expander

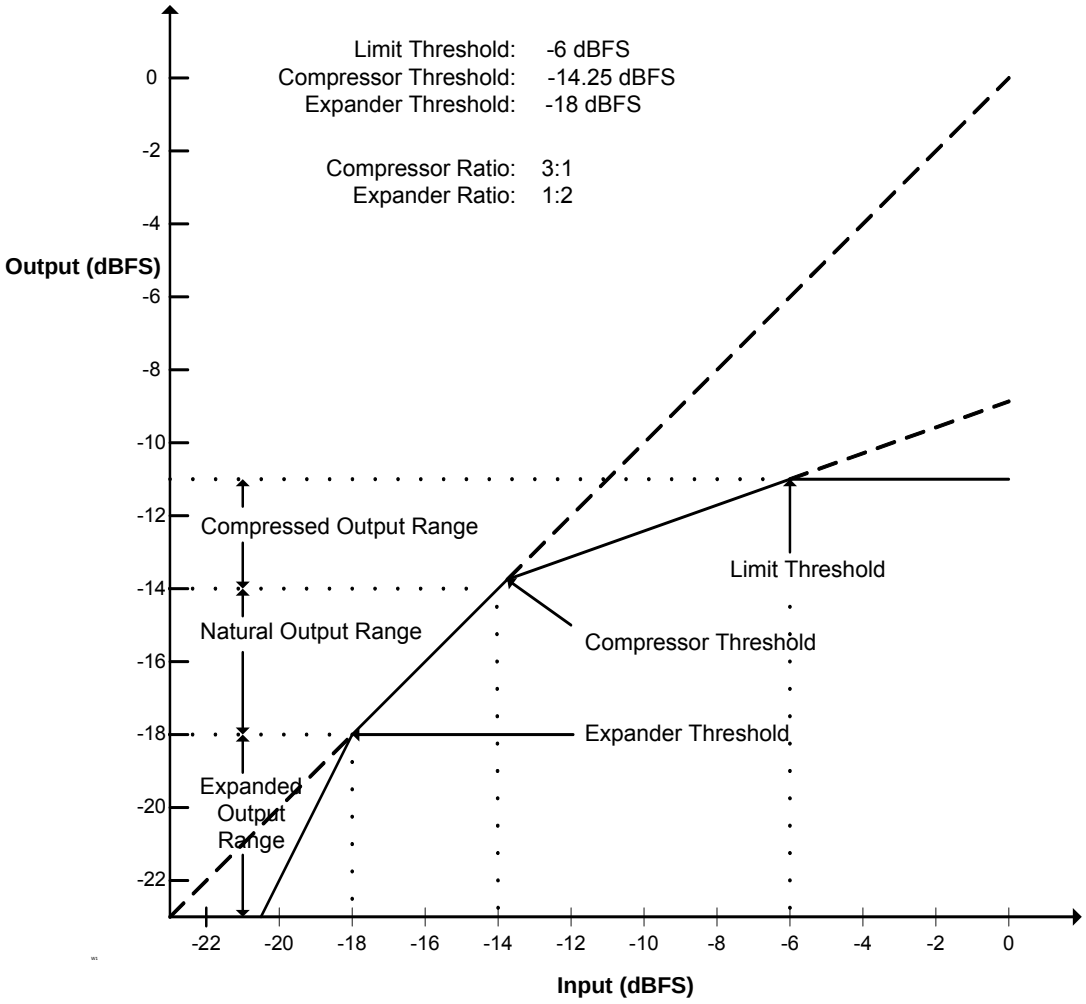

Figure 10. Gain Compressor, Output vs Input

3.6.1. Overview

The Limiter function will limit the output of the DSP module to the Class-D and DAC modules. If the signal is greater than 0dB it will saturate at 0dB as the final processing step within the DSP module.

There are times when the user may intentionally want the output Limiter to perform this saturation, for example +6dB of gain applied within the DSP gain control and then limited to 0dB when output to the Class-D module would result in a clipped signal driving the Speaker output. This clipped signal would obviously contribute to increased distortion on the Speaker output which from the user listening perception it would “sound louder”.

At other times, the system implementor may wish to protect speakers from overheating or provide hearing protection by intentionally limiting the output level before full scale is reached. A limit threshold, independent of the compressor threshold is provided for this purpose. It is expected that the limit threshold is set to a higher level than the compressor threshold.

The traditional compressor algorithm provides two functions simultaneously (depending on signal level). For higher level signals, it can provide a compression function to reduce the signal level. For lower level signals, it can provide an expansion function for either increasing dynamic range or noise gating.

The compressor monitors the signal level and, if the signal is higher than a threshold, will reduce the gain by a programmed ratio to restrict the dynamic range. Limiting is an extreme example of the compressor where, as the input signal level is increased, the gain is decreased to maintain a specific output level.

In addition to limiting the bandwidth of the compressed audio, it is common for compressed audio to also compress the dynamic range of the audio. The expansion function inTSCS42xx can help restore the original dynamics to the audio.

The expander is a close relative of the compressor. Rather than using signal dependent gain to restrict the dynamic range, the expander uses signal dependent gain to expand the dynamic range. Thus if a signal level is below a particular threshold, the expander will reduce the gain even further to extend the dynamic range of the material.

3.6.2. Configuration

This compressor limiter provides the following configurable parameters.

- Compressor/limiter
  - Threshold – The threshold above which the compressor will reduce the dynamic range of the audio in the compression region.
  - Ratio – The ratio between the input dynamic range and the output dynamic range. For example, a ratio of 3 will reduce an input dynamic range of 9db to 3db.
  - Attack Time – The amount of time that changes in gain are smoothed over during the attack phase of the compressor.
  - Release Time – The amount of time that changes in gain are smoothed over during the release phase of the compressor.
  - Makeup gain – Used to increase the overall level of the compressed audio.
- Expander
  - Threshold – The threshold below which the expander will increase the dynamic range of the audio.
  - Ratio – The ratio between the input dynamic range and the output dynamic range of the audio in the expansion range. For example a ratio of 3 will take an input dynamic range of 9db and expand it to 27db.
  - Attack Time – The amount of time that changes in gain are smoothed over during the attack phase of the expander
  - Release Time
  - - The amount of time that changes in gain are smoothed over during the release phase of the expander.
- Two level detection algorithms
  - RMS – Use an RMS measurement for the level.
  - Peak – Use a peak measurement for the level.

3.6.3. Controlling parameters

IIn order to control this processing, there are a number of configurable parameters. The parameters and their ranges are:

- Compressor/limiter
  - Threshold – -40db to 0db relative to full scale.
  - Ratio – 1 to 20
  - Attack Time – typically 0 to 500ms
  - Release Time – typically 25ms to 2 seconds
  - Makeup gain – 0 to 40db
- Expander
  - Threshold – -30 to -60 dB
  - Ratio – 1 to 6
  - Attack Time – same as above
  - Release Time – same as above.
- Two level detection algorithms
  - RMS
  - Peak

3.6.4. Limiter/Compressor/Expander Registers

3.6.4.1. General compressor/limiter/expander control Register

Register Address	Bit	Label	Type	Default	Description
R37 (25h) CLECTL	7:5	RSVD	R	0h	Reserved
	4	Lvl_Mode	RW	0	CLE Level Detection Mode 0 = Average 1 = Peak
	3	WindowSel	RW	0	Window width selection for level detection: 0 = equivalent of 512 samples of selected Base Rate (~10-16ms) 1 = equivalent of 64 samples of selected Base Rate (~1.3-2ms)
	2	Exp_en	RW	0	1 = enable expander
	1	Limit_en	RW	0	1 = enable limiter
	0	Comp_en	RW	0	1 = enable compressor

Table 43. CLECTL Register

3.6.4.2. Compressor/Limiter/Expander make-up gain Register

Register Address	Bit	Label	Type	Default	Description
R38 (26h) MUGAIN	7:5	RSVD	R	0h	Reserved
	4:0	CLEMUG[4:0]	RW	0h	0dB...46.5dB in 1.5dB steps

Table 44. MUGAIN Register

3.6.4.3. Compressor Threshold Register

Register Address	Bit	Label	Type	Default	Description
R39 (27h) COMPTH	7:0	COMPTH[7:0]	RW	00h	FFh...00h = 0dB...95.625dB in 0.375dB steps.

Table 45. COMPTH Register

3.6.4.4. Compressor ration register

Register Address	**Bit**	**Label**	**Type**	**Default**	**Description**
R40 (28h) CMPRAT	7:5	RSVD	R	000	Reserved
	4:0	CMPRAT[4:0]	RW	00h	Compressor Ratio 00h = Reserved 01h = 1.5:1 02h...14h = 2:1...20:1 15h...1Fh = Reserved

Table 46. CMPRAT Register

3.6.4.5. Compressor Attack Time Constant Register (Low)

Register Address	**Bit**	**Label**	**Type**	**Default**	**Description**
R41 (29h) CATKTCL	7:0	CATKTC[7:0]	RW	00h	Low byte of the time constant used to ramp to a new gain value during a compressor attack phase.

Table 47. CATKTCL Register

3.6.4.6. Compressor Attack Time Constant Register (High)

Register Address	Bit	Label	Type	Default	Description
R42 (2Ah) CATKTCH	7:0	CATKTC[15:8]	RW	00h	High byte of the time constant used to ramp to a new gain value during a compressor attack phase. The time constant is [high byte, low byte] 0000h = 0 (instantaneous) 0001h = 0.96875 + 1/(2^21) 0002h = 0.96875 + 2/(2^21) ... (step = 1/(2^21)) FFFEh = [(2^21)-2]/(2^21) FFFFh = [(2^21)-1]/(2^21)

Table 48. CATKTCH Register

3.6.4.7. Compressor Release Time Constant Register (Low)

Register Address	Bit	Label	Type	Default	Description
R43 (2Bh) CRELTCL	7:0	CRELTC[7:0]	RW	00h	Low byte of the time constant used to ramp to a new gain value during a compressor release phase.

Table 49. CRELTCL Register

3.6.4.8. Compressor Release Time Constant Register (High)

Register Address	Bit	Label	Type	Default	Description
R44 (2Ch) CRELTCH	7:0	CRELTC[15:8]	RW	00h	High byte of the time constant used to ramp to a new gain value during a compressor release phase. The time constant is [high byte, low byte] 0000h = 0 (instantaneous) 0001h = 0.96875 + 1/(2^21) 0002h = 0.96875 + 2/(2^21) ... (step = 1/(2^21)) FFFEh = [(2^21)-2]/(2^21) FFFFh = [(2^21)-1]/(2^21)

Table 50. CRELTCH Register

3.6.4.9. Limiter Threshold Register

Register Address	Bit	Label	Type	Default	Description
R45 (2Dh) LIMTH	7:0	LIMTH[7:0]	RW	00h	FFh...00h = 0dB...95.625dB in 0.375dB steps.

Table 51. LIMTH Register

3.6.4.10. Limiter Target Register

Register Address	Bit	Label	Type	Default	Description
R46 (2Eh) LIMTGT	7:0	LIMTGT[7:0]	RW	00h	FFh...00h = 0dB...95.625dB in 0.375dB steps.

Table 52. LIMTGT Register

3.6.4.11. Limiter Attack Time Constant Register (Low)

Register Address	Bit	Label	Type	Default	Description
R47 (2Fh) LATKTCL	7:0	LATKTC[7:0]	RW	00h	Low byte of the time constant used to ramp to a new gain value during a limiter attack phase.

Table 53. LATKTCL Register

3.6.4.12. Limiter Attack Time Constant Register (High)

Register Address	Bit	Label	Type	Default	Description
R48 (30h) LATKTCH	7:0	LATKTC[15:8]	RW	00h	High byte of the time constant used to ramp to a new gain value during a limiter attack phase. The time constant is [high byte, low byte] 0000h = 0 (instantaneous) 0001h = 0.96875 + 1/(2^21) 0002h = 0.96875 + 2/(2^21) ... (step = 1/(2^21)) FFFEh = [(2^21)-2]/(2^21) FFFFh = [(2^21)-1]/(2^21)

Table 54. LATKTCH Register

3.6.4.13. Limiter Release Time Constant Register (Low)

Register Address	Bit	Label	Type	Default	Description
R49 (31h) LRELTCL	7:0	LRELTC[7:0]	RW	00h	Low byte of the time constant used to ramp to a new gain value during a limiter release phase.

Table 55. LRELTCL Register

3.6.4.14. Limiter Release Time Constant Register (High)

Register Address	Bit	Label	Type	Default	Description
R50 (32h) LRELTCH	7:0	LRELTC[15:8]	RW	00h	High byte of the time constant used to ramp to a new gain value during a limiter release phase. The time constant is [high byte, low byte] 0000h = 0 (instantaneous) 0001h = 0.96875 + 1/(2^21) 0002h = 0.96875 + 2/(2^21) ... (step = 1/(2^21)) FFFEh = [(2^21)-2]/(2^21) FFFFh = [(2^21)-1]/(2^21)

Table 56. LRELTCH Register

3.6.4.15. Expander Threshold Register

Register Address	Bit	Label	Type	Default	Description
R51 (33h) EXPTH	7:0	EXPTH[7:0]	RW	00h	Expander threshold: 0...95.625dB in 0.375dB steps

Table 57. EXPTH Register

3.6.4.16. Expander Ratio Register

Register Address	Bit	Label	Type	Default	Description
R52 (34h) EXPRAT	7:3	RSVD	R	00h	Reserved
	2:0	EXPRAT[2:0]	RW	000	Expander Ratio 0h...1h = Reserved 2h...7h = 1:2...1:7

Table 58. EXPRAT Register

3.6.4.17. Expander Attack Time Constant Register (Low)

Register Address	Bit	Label	Type	Default	Description
R53 (35h) XATKTCL	7:0	XATKTC[7:0]	RW	00h	Low byte of the time constant used to ramp to a new gain value during a expander attack phase.

Table 59. XATKTCL Register

3.6.4.18. Expander Attack Time Constant Register (High)

Register Address	Bit	Label	Type	Default	Description
R54 (36h) XATKTCH	7:0	XATKTC[15:8]	RW	00h	High byte of the time constant used to ramp to a new gain value during a expander attack phase. The time constant is [high byte, low byte] 0000h = 0 (instantaneous) 0001h = 0.96875 + 1/(2^21) 0002h = 0.96875 + 2/(2^21) ... (step = 1/(2^21)) FFFEh = [(2^21)-2]/(2^21) FFFFh = [(2^21)-1]/(2^21)

Table 60. XATKTCH Register

3.6.4.19. Expander Release Time Constant Register (Low)

Register Address	Bit	Label	Type	Default	Description
R55 (37h) XRELTCL	7:0	XRELTC[7:0]	RW	0	Low byte of the time constant used to ramp to a new gain value during a expander release phase.

Table 61. XRELTCL Register

3.6.4.20. Expander Release Time Constant Register (High)

Register Address	Bit	Label	Type	Default	Description
R56 (38h) XRELTCH	7:0	XRELTC[15:8]	RW	0	High byte of the time constant used to ramp to a new gain value during a expander release phase. The time constant is [high byte, low byte] 0000h = 0 (instantaneous) 0001h = 0.96875 + 1/(2^21) 0002h = 0.96875 + 2/(2^21) ... (step = 1/(2^21)) FFFEh = [(2^21)-2]/(2^21) FFFFh = [(2^21)-1]/(2^21)

Table 62. XRELTCH Register

3.7. Output Effects

The TSCS42xx offers Bass enhancement, Treble enhancement, Stereo Depth enhancement. The output effects processing is outlined in the following sections.

3.7.1. FX Control Register

Register Address	Bit	Label	Type	Default	Description
R57 (39h) FXCTL	7:5	RSVD	R	000	Reserved
	4	3DEN	RW	0	3D Enhancement Enable 0 = Disabled 1 = Enabled
	3	TEEN	RW	0	Treble Enhancement Enable 0 = Disabled 1 = Enabled
	2	TNLFBYP	RW	0	Treble Non-linear Function Bypass: 0 = Enabled 1 = Bypassed
	1	BEEN	RW	0	Bass Enhancement Enable 0 = Disabled 1 = Enabled
	0	BNLFBYP	RW	0	Bass Non-linear Function Bypass: 0 = Enabled 1 = Bypassed

Table 63. FXCTL Register

3.7.2. Stereo Depth (3-D) Enhancement

The TSCS42xx has a digital depth enhancement option to artificially increase the separation between the left and right channels, by enabling the attenuation of the content common to both channels. The amount of attenuation is programmable within a range. The input is prescaled (fixed) before summation to prevent saturation.

The 3-D enhancement algorithm is a tried and true algorithm that uses two principles.

1 If the material common to the two channels is removed, then the output will sound more 3-D.

2 If the material for the opposite channel is presented to the current channel inverted, it will tend to cancel any material from the opposite channel on the current ear. For example, if the material from the right is presented to the left ear inverted, it will cancel some of the material from the right ear that is leaking into the right ear.

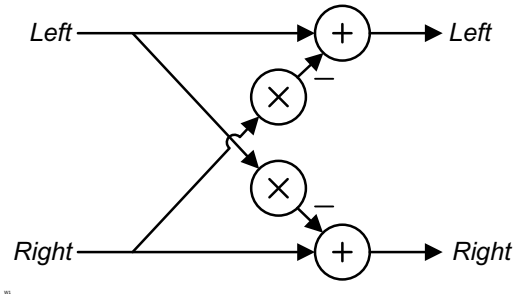

Figure 11. 3-D Channel Inversion

*Note: **.3D_Mix specifies** the amount of the common signal that is added from the left and right channels. This number is a fractional amount between -1 and 1. For proper operation, this value is typically negative.*

3.7.3. Psychoacoustic Bass Enhancement

One of the primary audio quality issues with small speaker systems is their inability to reproduce significant amounts of energy in the bass region (below 200Hz). While there is no magic mechanism to make a speaker reproduce frequencies that it is not capable of, there are mechanisms for fooling the ear into thinking that the bass material is being heard.

The psychoacoustic bass processor relies on a psychoacoustic principle called “missing fundamental”. If the human ear hears a proper series of harmonics for a particular bass note, the listener will hear the fundamental of that series, even if it is not present.

A processing algorithm using this principle allows for improving the apparent low frequency response of an audio system below what it is actually capable of. Below is a diagram of the implementation of this algorithm.

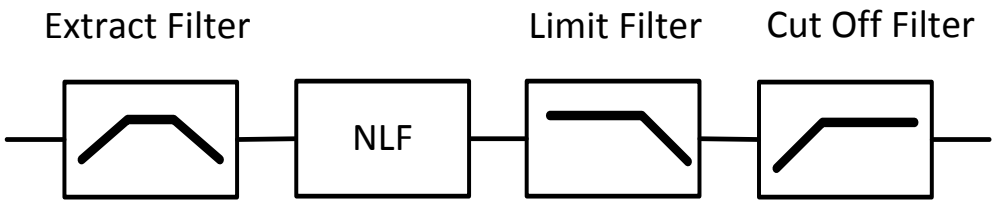

Figure 12. Bass Enhancement

3.7.4. Treble Enhancement

One of the mechanisms used to limit the bit rate for compressed audio is to first remove high frequency information before compression. When these files are decompressed, this can lead to dull sounding audio. The TSI treble enhancement replaces these lost high frequencies.

The enhanced treble function works much like the enhanced bass, however it's intended use is different. The Enhanced treble uses a non linear function to add treble harmonics to a signal that has limited high-frequency bandwidth (such as a low bit rate MP3). In this case, the algorithm makes use of the audio fact that presence of audio between 4-8K is a good predictor of audio between 10K-20K.

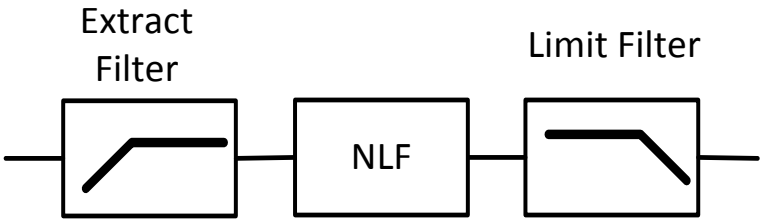

Figure 13. Treble Enhancement

The enhanced treble NLF has a different set of requirements than the psychoacoustic bass. In particular, the presence of odd high frequency harmonics is objectionable. Thus the most promising NLF for enhanced treble is a half wave rectifier.

3.8. Mute and De-Emphasis

The TSCS42xx has a Soft Mute function, which is used to gradually attenuate the digital signal volume to zero. The gain returns to its previous setting if the soft mute is removed. At startup, the codec is muted by default; to enable audio play, the mute bit must be cleared to 0.

After the equalization filters, de-emphasis may be performed on the audio data to compensate for pre-emphasis that may be included in the audio stream. De-emphasis filtering is only available for 48kHz, 44.1kHz, and 32kHz sample rates.

3.9. Mono Operation and Phase Inversion

Normal stereo operation converts left and right channel digital audio data to analog in separate DACs. However, it is also possible to have the same signal (left or right) appear on both analog output channels by disabling one channel; alternately, there is a mono-mix mode that mixes the two channels digitally before converting to analog using only one DAC. In this mode, the other DAC is switched off, and the resulting mixed stream signal can appear on both analog output channels.

The DAC output defaults to non-inverted. Setting DACPOLL and DACPOLR bits will invert the DAC output phase on the left and right channels.

3.9.1. DAC Control Register

Register Address	Bit	Label	Type	Default	Description
R24 (18h) CNVRTR1	7	DACPOLR	RW	0	Invert DAC Right signal
	6	DACPOLL	RW	0	Invert DAC Left signal
	5:4	DMONOMIX [1:0]	RW	00	DAC mono mix 00: stereo 01: mono ((L/2)+(R/2)) into DACL, ‘0’ into DACR 10: mono ((L/2)+(R/2)) into DACR, ‘0’ into DACL 11: mono ((L/2)+(R/2)) into DACL and DACR
	3	DACMU	RW	1	Digital Soft Mute 1 = mute 0 = no mute (signal active)
	2	DEEMP	RW	0	De-emphasis Enable 1 = Enabled 0 = Disable
	1:0	DACDITH	RW	00	DAC Dither Mode: 0 = Dynamic, half amplitude 1 = Dynamic, full amplitude 2 = DAC dither disabled 3 = Static

Table 64. CNVRTR1 Register

3.10. Analog Outputs

3.10.1. Headphone Output

The HPOut pins can drive a 16Ohm or 32Ohm headphone or alternately drive a line output. The signal volume of the headphone amplifier can be independently adjusted under software control by writing to HPVOL_L and HPVOL_R. Setting the volume to 0000000 will mute the output driver; the output remains at ground, so that no click noise is produced when muting or un-muting.
Gains above 0dB run the risk of clipping large signals.
To minimize artifacts such as clicks and zipper noise, the headphone and BTL outputs feature a volume fade function that smoothly changes volume from the current value to the target value.

Headphone Volume Control Registers

Register Address	Bit	Label	Type	Default	Description
R0 (00h) HPVOLL	7	RSVD	R	0	Reserved
	6:0	HPVOL_L [6:0]	RW	1110111 (0dB)	Left Headphone Volume 1111111 = +6dB 1111110 = +5.25dB … 1110111 = 0dB ... 0000001 = -88.5dB 0000000 = Analog mute Note: If HPVOLU is set, this setting will take effect after the next write to the Right Input Volume register.
R1 (01h) HPVOLR	7	RSVD	R	0	Reserved
	6:0	HPVOL_R [6:0]	RW	1110111	Right Headphone Volume 1111111 = +6dB 1111110 = +5.25dB … 1110111 = 0dB ... 0000001 = -88.5dB 0000000 = Analog mute

Table 65. HPVOL L/R Registers

3.10.2. Speaker Output

The RSPKOut (R+, R-) and LSPKOut (L+, L-) pins are controlled similarly, but independently of, the headphone output pins. They are intended to drive an 8 ohm or 4 ohm speaker pair.

3.10.2.1. Speaker Volume Control Registers

Register Address	Bit	Label	Type	Default	Description
R2 (2h) SPKVOLL	7	RSVD	R	0	Reserved
	6:0	SPKVOL_L [6:0]	RW	1101111 (0dB)	Left Speaker Volume 1111111 = +12dB 1111110 = +11.25dB … 1101111 = 0dB ... 0001000 to 0000001 = -77.25dB 0000000= Mute Note: If SPKVOLU is set, this setting will take effect after the next write to the Right Input Volume register.
R3 (3h) SPKVOLR	7	RSVD	R	0	Reserved
	6:0	SPKVOL_R [6:0]	RW	1101111 (0dB)	Right Speaker Volume 1111111 = +12dB 1111110 = +11.25dB … 1101111 = 0dB ... 0001000 to 0000001 = -77.25dB 0000000 = Mute

Table 66. SPKVOL L/R Registers

3.10.3. DDX™ Class D Audio Processing

For additional information on the DDX™ Class D solution, please see the application note on www.Temposemi.com.

The DDX™ Class D PWM Controller performs the following signal processing:

- Feedback filters are applied to shape any noise. The filters move noise from audible frequencies to frequencies above the audio range.
- The PWM block converts the data streams to tri-state PWM signals and sends them to the power stages.
- Finally, the Class-D controller block adjusts the output volume to provide constant output power across supply voltage.

The power stages boost the signals to higher levels, sufficient to drive speakers at a comfortable listening level.

3.10.3.1. Constant Output Power Mode

In normal operation the BTL amplifier is rated at 0.5W (full scale digital with 6dB BTL gain) into an 8 ohm load at 3.6V but will vary from about 0.38W to about 1.2W across a 3.1V to 5.5V supply range. However, when constant output power mode is enabled, the full scale output is held constant from 3.1V to 5.5V.

The BTL amplifier in TSCS42xx will continuously adjust to power supply changes to ensure that the full scale output power remains constant. This is not an automatic level control. Rather, this function prevents sudden volume changes when switching between battery and line power. Please note, when in this mode the amplifier efficiency may be reduced and decreases with higher supply voltages and lower target values.

A simple 5-bit ADC is used to monitor PVDD. As PVDD raises or lowers, the analog circuit will send a 5-bit code to the digital section that will average and then calculate a gain adjustment. The BTL audio signal will be multiplied by this gain value (in addition to the user volume controls).

The user will select a target value for the circuit. The constant output function will calculate a gain adjustment that will provide approximately the same full scale output voltage as provided when PVDD causes the same code value. So, if the target is 9 then a PVDD voltage of about 3.7V would generate a code value of 9 and a full scale output power of about 630mW into 8 ohms. If PVDD should rise to 4V, generating a code of 13, then the constant output power circuit would reduce the gain by 0.75dB (4 codes * 0.1875dB) to keep the full scale output at the target level.

The circuit may be configured to add gain, attenuation, or both to maintain the full-scale output level. If the needed adjustment falls outside of the range of the circuit (only attenuation is enabled and gain is needed, for example) then the circuit will apply as much correction as it is able. Through the use of

gain, attenuation, and target values, different behaviors may be implemented:

- Attenuation only, target set to mimic a low supply voltage - Constant output level across battery state with constant quality (THD/SNR)
- Attenuation only, target set to mimic a moderate supply voltage - Output limiting to an approximate power level. Level will decrease at lower supply voltages but won't increase beyond a specific point.
- Gain only, target at or near max - Output will remain relatively constant but distortion will increase as PVDD is lowered. This mimics the behavior of common class-AB amplifiers.
- Gain and attenuation - Output remains at a level below the maximum possible at the highest supply voltage and above the theoretical full scale at minimum supply. Full scale PCM input clips when the supply voltage is low but won't become too loud when the supply voltage is high.

In addition to maintaining a constant output level, PVDD may be monitored for a large, sudden, change. If the High Delta function is enabled and PVDD changes more than 4 code steps since the last cycle, the output will be rapidly reduced then gradually increased to the target level.

When using this circuit, please take note of the following:

- The full scale output power may be limited by the supply voltage.
- Full scale output power is affected by other gain controls in the output path including the EQ and compressor/limiter.
- The Constant Output Power function is intended to help maintain a constant output level, not an exact output level. The output level for a specific target may vary part to part. If limiting is required for safety or other reasons, be conservative and set the target well below the maximum allowable level.
- Noise on the PVDD supply may cause erratic behavior. Use the recommended supply decoupling caps and verify that the power supply can support the peak currents demanded by a class-D amplifier.

Constant Output Power error (dB) relative to a target of 8 for an ideal part and the output error if left uncorrected across a 3.1 to 5.5V supply range.

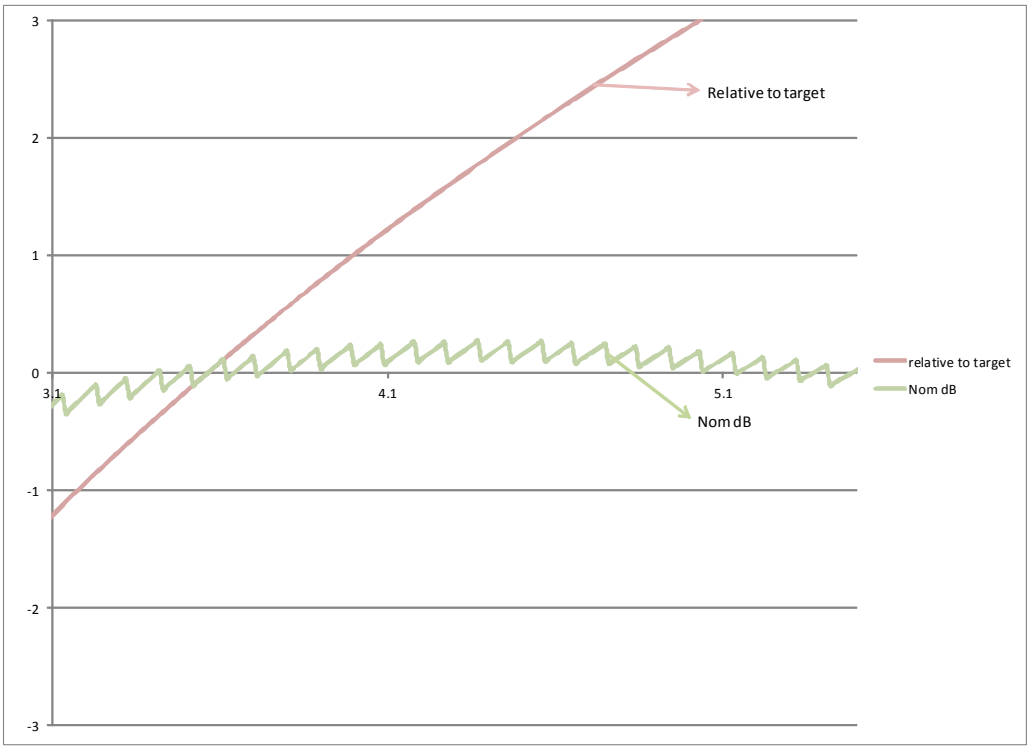

Figure 14. Constant Output Power Error

Constant Output Power for nominal and high/low reference across a 3.1 to 5.5V supply range.(Uncorrected power shown for reference) A target of 8 roughly corresponds to 0.5W at 3.6V into 8 ohms.

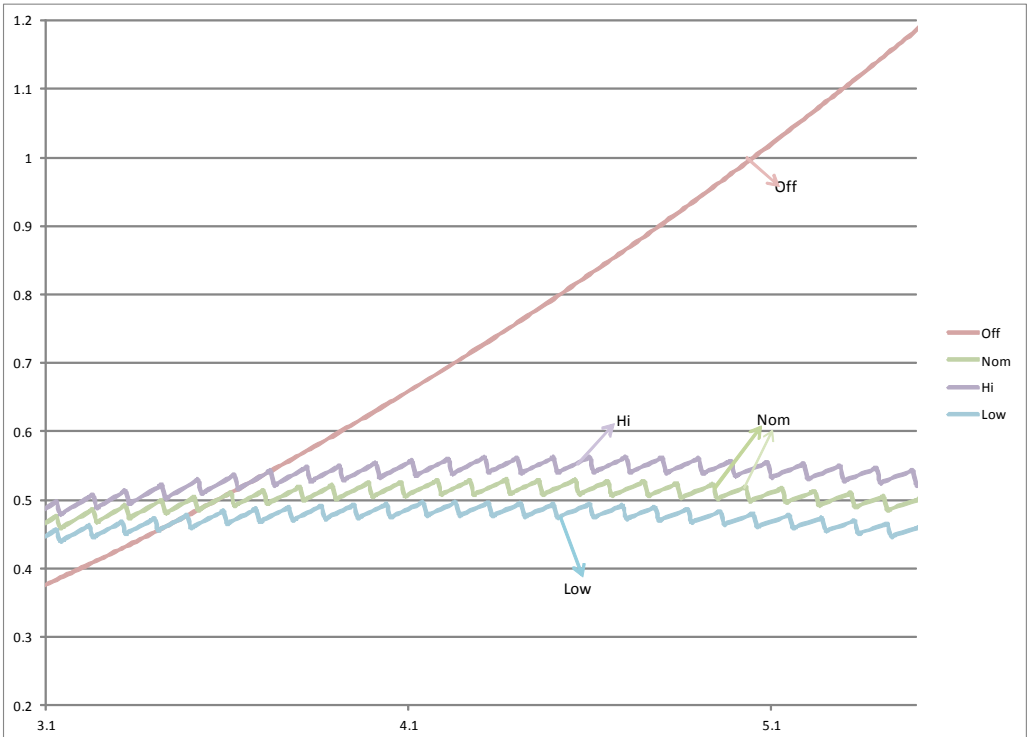

Figure 15. Constant Output Power nominal and high/low

3.10.3.2. Under Voltage Lock Out

When the PVDD supply becomes low, the BTL amplifier may be disabled to help prevent undesirable amplifier operation (overheat) or system level problems (battery under-voltage.)

The same circuit that monitors the PVDD supply to help maintain a constant output power is used to monitor the PVDD supply for a critical under-voltage situation. If the sense circuit consistently returns a 0 code then the PVDD supply is less than the minimum required for proper operation. To prevent accidental shutdown due to a noisy supply at the minimum operating range, the output of the PVDD sense circuit will be averaged for at least 200ms.

3.10.3.3. Register

Registers Constant Output Power 1

Register Address	Bit	Label	Type	Default	Description
R34 (22h) COP1	7	COPAtten	RW	0	1 = Constant Output Power function will use attenuate the BTL output if the PVDD sense circuit returns a code higher than the target value.
	6	COPGain	RW	0	1 = Constant Output Power function will use attenuate the BTL output if the PVDD sense circuit returns a code higher than the target value.
	5	HDeltaEn	RW	0	1 = If the PVDD code value has changed more than 4 counts since the last gain adjustment, the output will be reduced rapidly then slowly returned to the target level.
	4:0	COPTarget[4:0]	RW	8h	5-bit target for the Constant Output Power function.

Table 67. COP1 Register

Registers Constant Output Power 2

Register Address	Bit	Label	Type	Default	Description
R35 (23h) COP2	7	RSVD	R	0	Reserved
	6	RSVD	R	0	Reserved
	5:3	AvgLength[2:0]	RW	000	Constant Output Power Average Length (number of supply detect cycles to average): 0h = 1 1h = 2 2h = 4 3h = 8 4h = 16 5h = 32 6h = 64 7h = 128 8h = 256 9h = 512 Ah-Fh = Reserved
	2:0	MonRate[2:0]	RW	100	Supply Detect Monitor Rate: 0h = 0.25ms 1h = 0.5ms 2h = 1ms 3h = 2ms

Table 68. COP2 Register

Registers Constant Output Power 3

Register Address	Bit	Label	Type	Default	Description
R137 (89h) COP3	7	HIGHDELTA	R	0	Constant Output Power High Delta Status: 0 = No high delta event is currently being detected or recovered from 1 = A high delta event has been detected and the COP function is adjusting.
	6	UNDERVOLTAGE	R	0	Under Voltage Lockout Status: 0 = Supply is not below the UVLO threshold 1 = Supply is below the UVLO threshold.
	5:0	COPADJ	R	0h	Constant Output Power Adjustment Status (0.1875dB Steps, Twos Complement Value): 20h = -6dB 21h = -5.8125dB ... FFh = -0.1875dB 00h = 0dB 01h = +0.1875dB ... 1Fh = +5.8125dB

Table 69. COP3 Register

Configuration Register

Register Address	Bit	Label	Type	Default	Description
R31 (1Fh) CONFIG0	7:6	ASDM[1:0]	RW	10h	ADC Modulater Rate: 00b = Reserved 01b = Half 10b = Full 11b = Auto
	5:4	DSDM[1:0]	RW	10h	DAC Modulater Rate: 00b = Reserved 01b = Half 10b = Full 11b = Auto
	3:2	RSVD	R	0h	Reserved for future use.
	1	DC_BYPASS	RW	0	DAC DC Filter Bypass: 0 = Filter enabled 1 = Filter bypassed
	0	SD_FORCE_ON	RW	0	Supply Detect Force On: 0 = Supply detect not forced on 1 = Supply detect forced on. Note: If not forced on, the supply detect logic will automatically be enabled when features that use it are enabled (COP, UVLO

Table 70. CONFIG0 Register

PWM Control 0 Register

Register Address	Bit	Label	Type	Default	Description
R66 (42h) PWM0	7:6	SCTO	RW	11	Class-D Short Circuit Detect Time-out 00 = 10uS 01 = 100uS 10 = 500uS 11 = 100mS
	5	UVLO	RW	1	Under Voltage Lock Out 1 = BTL output disabled if PVDD sense circuit returns code 0
	4	RESERVED	RW	1	Reserved
	3	BFCLR	RW	0	PWM Noise Shaper Clear: 0 = Filter enabled 1 = Filter disabled.
	2	PWMMODEr	RW	1	PWM Modulation Type: 0 = Binary 1 = Ternary
	1	RESERVED	RW	0	Reserved
	0	NOOFFSET	RW	0	No Offset between left/right PWM frames: 0 = Frames offset 1 = Frames aligned

Table 71. PWM0 Register

PWM Control 1 Register

Register Address	Bit	Label	Type	Default	Description
R67 (43h) PWM1	7	RSVD	R	0	Reserved
	6:2	dithpos[4:0]	RW	0	Dither position, where dither inserted after NS. 0,1,2 = dither bits 2:0 4 = dither bits 3:1 5 = dither bits 4:1 .... 19 = dither bits 19:17
	1	dith_range	RW	0	1 = dither -1 to +1, 0 = -3 to +3
	0	dithclr	RW	0	1 = disable dither

Table 72. PWM1 Register

PWM Control 2 Register

Register Address	Bit	Label	Type	Default	Description
R68 (44h) PWM2	7:2		R	0h	Reserved
	1		R	0	Reserved
	0		R	0	Reserved

Table 73. PWM2 Register

PWM Control 3 Register

Register Address	Bit	Label	Type	Default	Description
R69 (45h) PWM3	7:6	outctrl[1:0]	RW	0h	pwm output muxing 0 = normal 1 = swap 0/1 2 = ch0 on both 3 = ch1 on both
	5:3		R	0h	Reserved
	2:0	cvalue[2:0]	RW	3h	PWM C Value

Table 74. PWM3 Register

3.10.4. Other Output Capabilities

Each audio analog output can be separately enabled. Disabling outputs serves to reduce power consumption, and is the default state of the device.

3.10.4.1. Audio Output Control

See Power management section. The output enable bits are also power management bits and the outputs will be turned off when disabled.

Register Address	Bit	Label	Type	Default	Description
R27 (1Bh) PWRM2	7	D2S	RW	0	Analog Input D2S: 0 = Power down 1 = Power up
	6	HPL	RW	0	Headphone Left Output Buffer + DAC: 0 = Power down 1 = Power up
	5	HPR	RW	0	Headphone Right Output Buffer + DAC: 0 = Power down 1 = Power up
	4	SPKL	RW	0	Speaker Left Output Buffer: 0 = Power down 1 = Power up
	3	SPKR	RW	0	Speaker Right Output Buffer: 0 = Power down 1 = Power up
	2	RSVD	RW	0	Reserved(bit implemented but unused)
	1	RSVD	RW	0	Reserved (bit implemented bur unused)
	0	VREF	RW	1	Vref (necessary for all other functions): 0 = Power down 1 = Power up
Note: A value of “1” indicates the output is enabled; a value of ‘0’ disables the output.					

Table 75. PWRM2 Register

3.10.5. Headphone Switch

The HP_DET pin is used to detect connection of a headphone. When headphone insertion is detected, the codec can automatically disable the speaker outputs and enable the headphone outputs. Control bits determine the meaning and polarity of the input.

In addition to enabling and disabling outputs, the EQ may also be controlled using the HP_DET pin. The 2 EQ filters may be configured so that one EQ is active when the Headphone output is active and the other EQ is active when the Speaker output is active (independent HP and Speaker EQ). One EQ may be enabled only when the Speaker is active and the other EQ may be on when either of the outputs are active (Speaker compensation and USER EQ) or other combinations are possible. Note that the EQ coefficients must be programmed and the EQs must be enabled using their control registers. The HP_DET logic can only disable the EQ filters.

3.10.5.1. Headphone Switch Register

Register Address	Bit	Label	Type	Default	Description
R28 (1Ch) CTL	7	HPSWEN	RW	0h	Headphone Switch Enable: 0 = Headphone switch disabled 1 = Headphone switch enabled
	6	HPSWPOL	RW	0h	Headphone Switch Polarity: 0 = HPDETECT high indicates headphone 1 = HPDETECT high indicates speaker
	5:4	EQ2SW	RW	0h	EQ2 behavior due to speaker/headphone output state: 00b = EQ is not disabled due to headphone/speaker logic 01b = EQ is disabled when headphone output is active 10b = EQ is disabled when speaker output is active 11b = EQ is disabled when headphone AND speaker output are active
	3:2	EQ1SW	RW	0h	EQ1 behavior due to speaker/headphone output state: 00b = EQ is not disabled due to headphone/speaker logic 01b = EQ is disabled when headphone output is active 10b = EQ is disabled when speaker output is active 11b = EQ is disabled when headphone AND speaker output are active
	1	TSDEN	RW	0h	Thermal Shutdown Enable (See section 7.9) 0: thermal shutdown disabled 1: thermal shutdown enabled
	0	TOEN	RW	0h	Zero Cross Time-out Enable 0: Time-out Disabled 1: Time-out Enabled - volumes updated if no zero cross event has occurred before time-out

Table 76. CTL Register

3.10.5.2. Speaker Operation

HPSWEN	HPSWPOL	HP_DET Pin state	SPKOut[1]	Speaker Enabled
0	X	X	0	no
0	X	X	1	yes
1	0	0	0	no
1	0	0	1	yes
1	0	1	X	no
1	1	0	X	no
1	1	1	0	no
1	1	1	1	yes

Table 77. Speaker Operation

1.SPKOut = Logical OR of the SPKL and SPKR enable (power state) bits

3.10.5.3. EQ Operation

EQnSW1	EQnSW0	EQ Behavior[1]
0	0	EQ is not disabled due to Headphone/Speaker logic
0	1	EQ is disabled when Headphone output is active
1	0	EQ is disabled when Speaker output is active
1	1	EQ is disabled when Headphone AND Speaker output are active

Table 78. EQ Operation

1.EQ must be enabled. EQ behavior is dependent on HP_DET and Output power state programming.

3.11. Thermal Shutdown

To avoid overpowering and overheating the codec when the amplifier outputs are driving large currents, the TSCS42xx incorporates a thermal protection circuit. If enabled, and the device temperature reaches approximately 150°C, the speaker and headphone amplifier outputs will be disabled. Once the device cools, the outputs will be automatically re-enabled.

3.11.1. Algorithm description:

There are 2 trip points, “high” and “low”. High indicates a critical overheat requiring a reduction in volume to avoid damage to the part. Low is set for a slightly lower temperature point, indicating that the current level is safe but that increased volume would result in a critical overheat condition.

Normally, the overheat bits are polled every 8ms but may be polled at 4ms, 8ms, 16ms, or 32ms by adjusting the Poll value. Reductions in volume will be allowed to happen at the Poll rate. Increases in volume are programmable to happen every 1, 2, 4, or 8 Poll cycles and in steps of 0.75dB to 6dB. This allows a full scale volume increase in a range of 10s of milliseconds to 10s of seconds.

When both overheat bits are 0, the volume is allowed to increment by the IncStep size, unless the volume has already reached the maximum value allowed. Any subsequent increment will be held off until the programmed number of polling cycles have occurred.

When the low overheat bit is 1 and the high overheat bit is 0, this indicates that the volume is currently at a safe point but the temperature is higher than desired and incrementing the volume may cause severe overheating. The volume is held at the current value.

When the high overheat bit is 1, damage could occur, so the volume setting will be immediately reduced by the Decrement Step value. As the overheat bits are re-polled, this volume reduction will continue until the high overheat bit drops to 0 or the volume value reaches the minimum setting. If the high overheat bit remains 1 even at the minimum setting, then the mute control bit will be asserted. If the high overheat bit persists even after mute, then the BTL amp will be powered down.

3.11.2. Thermal Trip Points.

The high and low trip points can be adjusted to suit the needs of a particular system implementation. There is a “shift” value (TripShift) which sets the low trip point, and there is a “split” value (TripSplit) that sets how many degrees above the low trip point the high trip point is.

By default:

TripShift = 2 (140 degrees C)
TripSplit = 0 (15 degrees C)

Therefore:

High Trip Point = 155°C.
Low Trip Point = 140°C.

3.11.3. Instant Cut Mode

This mode can be used to make our algorithm react faster to reduce thermal output but will cause more pronounced volume changes. If enabled:

- Only the high overheat is used, the low overheat is ignored.
- Whenever polled, if the high overheat is 1, then the volume setting will immediately be set to 0h.
- Conversely, if the high overheat is 0, the volume setting will immediately be set to the MaxVol value.
- Both volume clear and volume set events occur at the polling rate.

During this mode, the algorithm still possesses the ability to mute and then power down the BTL amp if the high overheat continues to be 1. This mode is disabled by default.

3.11.4. Short Circuit Protection

To avoid damage to the outputs if a short circuit condition should occur, both the headphone and BTL amplifiers implement short circuit protection circuits. The headphone output amplifier will detect the load current and limit its output if in an over current state. The BTL amplifier will sense a short to PVDD, ground, or between its +/- outputs and disable its output if a short is detected. After a brief time, the amplifier will turn on again. If a short circuit condition is still present, the amplifier will disable itself again.

3.11.5. Thermal Shutdown Registers

3.11.5.1. Temp Sensor Control/Status Register

Register Address	Bit	Label	Type	Default	Description
R29 (1Dh) THERMTS	7	TripHighStat	R	0	Temp sensor high trip point status 0 = Normal Operation 1 = Over Temp Condition
	6	TripLowStat	R	0	Temp sensor low trip point status 0 = Normal Operation 1 = Over Temp Condition
	5:4	TripSplit[1:0]	RW	0h	Temp sensor “split” setting. Determines how many degrees above the low trip point the high trip is set: 0h = 15 Degrees C 1h = 30 Degrees C 2h = 45 Degrees C 3h = 60 Degrees C.
	3:2	TripShift[1:0]	RW	2h	Temp sensor “shift” setting. Determines the low trip temperature: 0h = 110 Degrees C 1h = 125 Degrees C 2h = 140 Degrees C 3h = 155 Degrees C.
	1:0	Poll[1:0]	RW	1h	Temp sensor polling interval 0h = 4ms 1h = 8ms 2h = 16ms 3h = 32ms

Table 79. THERMTS Register

3.11.5.2. Temp Sensor Status Register

Register Address	Bit	Label	Type	Default	Description
R30 (1Eh) THERMSPKR1	7	ForcePwd	RW	1	Force powerdown enable for the speaker thermal algorithm: 0 = Speaker will remain powered up even if the temp sensor continues to report an overheat condition at minimum volume (mute) 1 = Speaker will be powered down if the temp sensor reports an overheat at the minimum volume (mute)
	6	InstCutMode	RW	0	Instant Cut Mode 0 = Both temp sensor status bits used to smoothly adjust the volume. 1 = Only the high temp sensor status bit will be used to set the volume. volume will be set to the full volume or mute (IncStep and DecStep are ignored.)
	5:4	IncRatio[1:0]	RW	0h	Increment interval ratio. Determines the ratio between the speaker volume increment interval and the speaker volume decrement interval (increment rate is equal to or slower than decrement rate): 0h = 1:1 1h = 2:1 2h = 4:1 3h = 8:1
	3:2	IncStep[1:0]	RW	0h	Increment step size for the speaker thermal control algorithm (occurs at the temp sensor polling rate X the increment interval ratio.) 0h = 0.75dB 1h = 1.5dB 2h = 3.0dB 3h = 6.0dB
	1:0	DecStep[1:0]	RW	1h	Decrement step size for the speaker thermal control algorithm (occurs at the temp sensor polling rate.) 0h = 3dB 1h = 6dB 2h = 12dB 3h = 24dB

Table 80. THERMTSPKR1 Register

Register Address	Bit	Label	Type	Default	Description
R136 (88h) THERMSPKR2	7	ForcePwdStatus	R	0	0: Speaker not powered down due to thermal algorithm 1: Speaker has been powered down because overtemp condition was present even though the speaker was muted.
	6:0	VolStatus[6:0]	R	08	Current speaker volume value. If no overheat is being reported by the temperature sensor, this value should be equal to the greater of the left or right speaker volume setting.

Table 81. THERMTSPKR2 Register

4. INPUT AUDIO PROCESSING

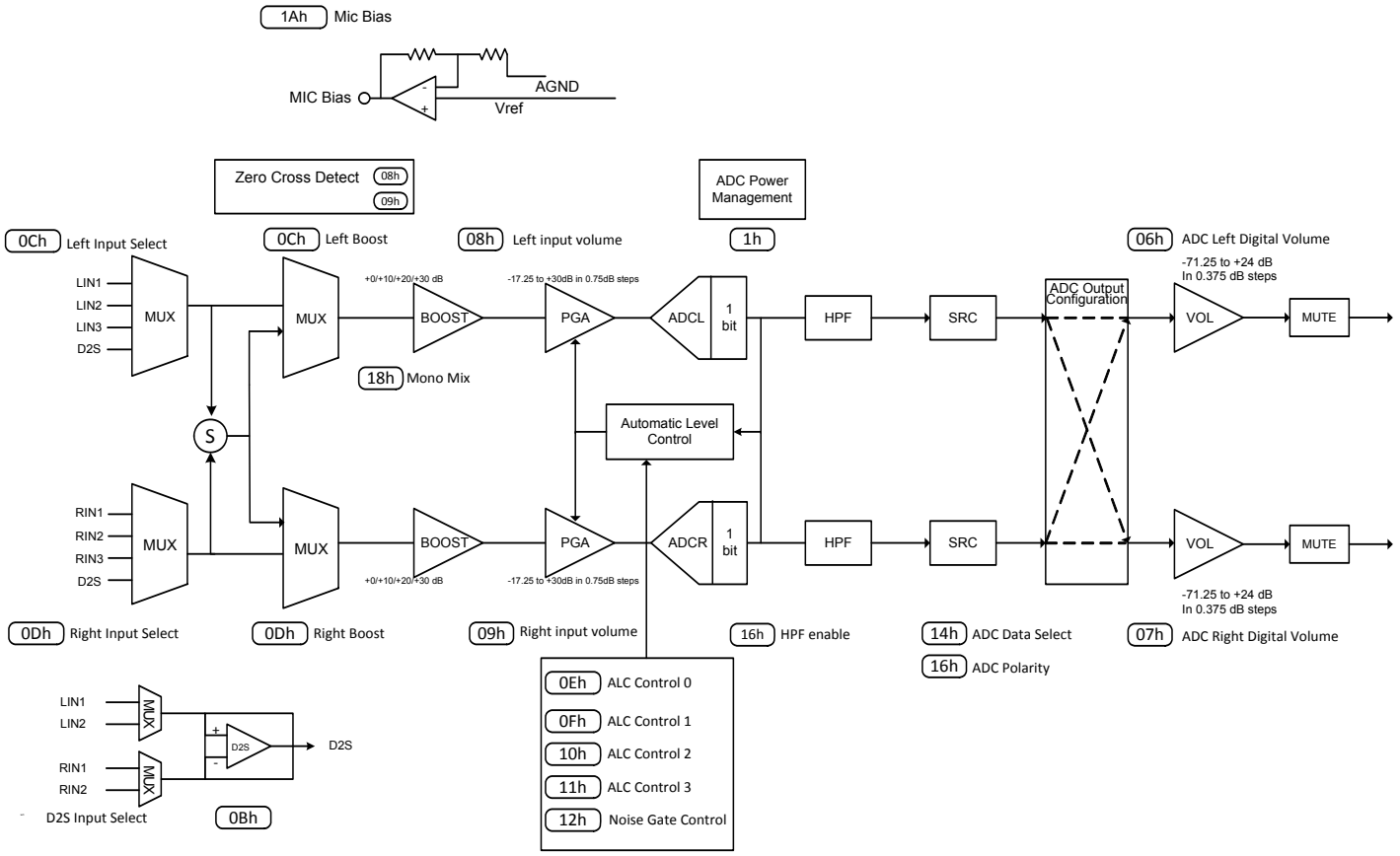

Figure 16. Input Audio Processing

4.1. Analog Inputs

The TSCS42xx provides multiple high impedance, low capacitance AC-coupled analog inputs with an input signal path to the stereo ADCs. Prior to the ADC, there is a multiplexor that allows the system to select which input is in use. Following the mux, there is a programmable gain amplifier and also an optional microphone gain boost. The gain of the PGA can be controlled either by the system, or by the on-chip level control function. The stereo record path can also operate with the two channels mixed to mono either in the analog or digital domains.

Signal inputs are biased internally to AVSS but AC coupling capacitors are required when connecting microphones (due to the 2.5V microphone bias) or when offsets would cause unacceptable “zipper noise” or pops when changing PGA or boost gain settings. To avoid audio artifacts, the line inputs are kept biased to analog ground when they are muted or the device is placed into standby mode.

4.1.1. Input Software Control Register

Register Address	Bit	Label	Type	Default	Description
R12 (0Ch) INSELL	7:6	INSEL_L	RW	00	Left Channel Input Select 00 = LINPUT1 01 = LINPUT2 10 = LINPUT3 11 = D2S
	5:4	MICBST_L	RW	00	Left Channel Microphone Gain Boost 00 = Boost off (bypassed) 01 = 10dB boost 10 = 20dB boost 11 = 30dB boost
	3:0	RSVD	R	0000	Reserved
R13 (0Dh) INSELR	7:6	INSEL_R	RW	00	Right Channel Input Select 00 = RINPUT1 01 = RINPUT2 10 = RINPUT3 11 = D2S
	5:4	MICBST_R	RW	00	Right Channel Microphone Gain Boost 00 = Boost off (bypassed) 01 = 10dB boost 10 = 20dB boost 11 = 30dB boost
	3:0	RSVD	R	0000	Reserved

Table 82. INSELL and INSLR Register

4.2. Mono Mixing and Output Configuration

The stereo ADC can operate as a stereo or mono device, or the two channels can be mixed to mono. Mixing can occur either in the input path (analog, before ADC) or after the ADC. MONOMIX determines whether to mix to mono, and where.

For analog mono mix, either the left or right channel ADC can be used for the audio stream. The other ADC may be powered off to conserve power. A differential input amplifier may be selected as a mono source to either ADC input. This D2S amplifier can select either Input 1 or Input 2 using the DS bit.

The system also has the flexibility to select the data output. ADCDSEL configures the interface, assigning the source of the left and right ADC independently.

4.2.1. ADC D2S Input Mode Register

Register Address	Bit	Label	Type	Default	Description
R11 (0Bh) INMODE	7:1	RSVD	R	0h	Reserved
	0	DS	RW	0	Differential Input Select 0: LIN1 - RIN1 1: LIN2 - RIN2

Table 83. INMODE Register

4.2.2. ADC Mono, Filter, and Inversion

Register Address	Bit	Label	Type	Default	Description
R22 (16h) CNVRTR0	7	ADCPOLR	RW	0	ADC Right Channel Polarity 0 = normal 1 = inverted
	6	ADCPOLL	RW	0	ADC Left Channel Polarity 0 = normal 1 = inverted
	5:4	AMONOMIX [1:0]	RW	00	ADC mono mix 00: Stereo 01: Analog Mono Mix (using left ADC) 10: Analog Mono Mix (using right ADC) 11: Digital Mono Mix (ADCL/2 + ADCR/2 on both Left and Right ADC outputs)
	3	ADCMU	RW	1	1 = Mute ADC
	2	HPOR	RW	0	High Pass Offset Result 0 = discard offset when HPF disabled 1 = store and use last calculated offset when HPF disabled
	1	ADCHPDR	RW	0	ADC High Pass Filter Disable (Right)
	0	ADCHPDL	RW	0	ADC High Pass Filter Disable (Right)

Table 84. CNVRTR0 Register

4.2.3. ADC Data Output Configuration

Register Address	Bit	Label	Type	Default	Description
R20 (14h) AIC2	7:6	DACDSEL[1:0]	RW	00	00: left DAC = left I2S data; right DAC = right I2S data 01: left DAC = left I2S data; right DAC = left I2S data 10: left DAC = right I2S data; right DAC = right I2S data 11: left DAC = right I2S data; right DAC = left I2S data
	5:4	ADCDSEL[1:0]	RW	00	00: left I2S data = left ADC; right I2S data = right ADC 01: left I2S data = left ADC; right I2S data = left ADC 10: left I2S data = right ADC; right I2S data = right ADC 11: left I2S data = right ADC; right I2S data = left ADC
	3	TRI	RW	0	Interface Tri-state (See Section 5.4.3)
	2:0	BLRCM	RW	0	Bitclock and LRClock mode (See Section 5.4.3)

Table 85. AIC2 Register

4.3. Microphone Bias

The MICBIAS output is used to bias electric type microphones. It provides a low noise reference voltage used for an external resistor biasing network. The MICB control bit is used to enable the output.

The MICBIAS can source up to 3mA of current; therefore, the external resistors must be large enough to conform to this limit.

4.3.1. Microphone Bias Control Bit

Register Address	Bit	Label	Type	Default	Description
R26 (1Ah) PWRM1	1	MICB	RW	0	Microphone Bias Enable 0 = OFF (high impedance output) 1 = ON

Table 86. Mic Bias Enable

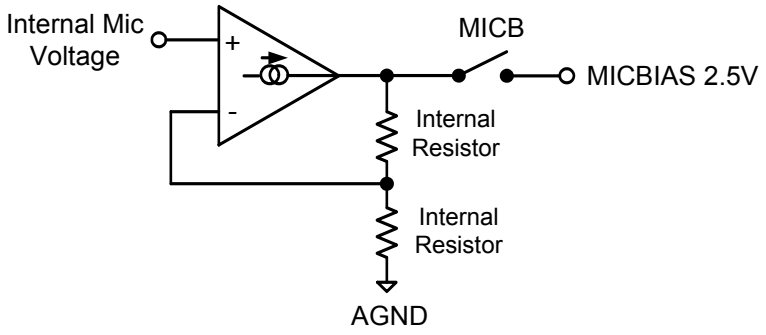

Figure 17. Mic Bias

4.4. Programmable Gain Control

The Programmable Gain Amplifier (PGA) enables the input signal level to be matched to the ADC input range. Amplifier gain is adjustable across the range +30dB to –17.25dB (using 0.75dB steps). The PGA can be controlled directly by the system software using the Input Volume Control registers (INVOLL and INVOLR), or alternately the Automatic Level Control (ALC) function can automatically control the gain. If the ALC function is used, writing to the Input Volume Control registers has no effect.

Left and right input gains are independently adjustable. By controlling the update bit INVOLU in R10, the left and right gain settings can be simultaneously updated. To eliminate zipper noise, LZCEN and RZCEN bits enable a zero-cross detector to insure changes only occur when the signal is at zero. A time-out for zero-cross is also provided, using TOEN in register R28 (1Dh).

4.4.1. Input PGA Software Control Register

Register Address	Bit	Label	Type	Default	Description
R8 (08h) INVOLL	7	INMUTEL	RW	0	Left Input Mute: 1 = Enable mute 0 = Disable mute Note: If INVOLU is set, this setting will take effect after the next write to the right Input Volume Register
	6	IZCL	RW	0	Left Channel Zero Cross Detector 1 = Change gain on zero cross only 0 = Change gain immediately Note: If INVOLU is set, this setting will take effect after the next write to the Right Input Volume register.
	5:0	INVOL_L [5:0]	RW	010111 (0dB)	Left Channel Input Volume Control 111111 = +30dB 111110 = +29.25dB ... 0.75dB steps down to 000000 = -17.25dB Note: If INVOLU is set, this setting will take effect after the next write to the Right Input Volume register.
R9 (09h) INVOLR	7	RSVD	R	0	Reserved
	6	IZCR	RW	0	Right Channel Zero Cross Detector 1 = Change gain on zero cross only 0 = Change gain immediately
	5:0	INVOL_R [5:0]	RW	010111 (0dB)	Right Channel Input Volume Control 111111 = +30dB 111110 = +29.25dB ... 0.75dB steps down to 000000 = -17.25dB
R28 (1Ch) CTL	0	TOEN	RW	0	Zero Cross Time-out Enable 0: Time-out Disabled 1: Time-out Enabled - volumes updated if no zero cross event has occurred before time-out

Table 87. INVOLL/ INVOLR Register

4.5. ADC Digital Filter

To provide the correct sampling frequency on the digital audio outputs, ADC filters perform true 24-bit signal processing and convert the raw multi-bit oversampled data from the ADC using the digital filter path illustrated below.

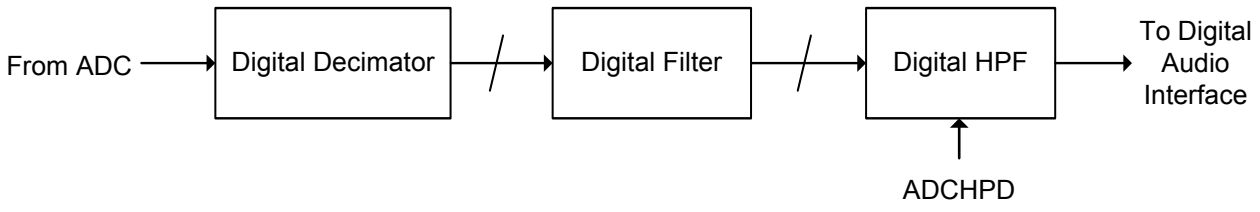

Figure 18. ADC Filter Data Path

The ADC digital filters contain a software-selectable digital high pass filter. When the high-pass filter is enabled, the dc offset is continuously calculated and subtracted from the input signal. The HPOR bit enables the last calculated DC offset value to be stored when the high-pass filter is disabled; this value will then continue to be subtracted from the input signal. To provide support for calibration, the stored and subtracted value will not change unless the high-pass filter is enabled even if the DC value is changed. The high pass filter may be enabled separately for each of the left and right channels.

The output data format can be programmed by the system. This allows stereo or mono recording streams at both inputs. Software can change the polarity of the output signal.

4.5.1. ADC Signal Path Control Register

Register Address	Bit	Label	Type	Default	Description
R22 (16h) CNVRTR0	7	ADCPOLR	RW	0	0 = Right polarity not inverted 1 = Right polarity inverted
	6	ADCPOLL	RW	0	0 = Left polarity not inverted 1 = Left polarity inverted
	5:4	AMONOMIX [1:0]	RW	00	ADC mono mix 00: Stereo 01: Analog Mono Mix (using left ADC) 10: Analog Mono Mix (using right ADC) 11: Digital Mono Mix
	3	ADCMU	RW	1	1 = Mute ADC
	2	HPOR	RW	0	High Pass Offset Result 0 = discard offset when HPF disabled 1 = store and use last calculated offset when HPF disabled
	1	ADCHPDR	RW	0	ADC High Pass Filter Disable (Right)
	0	ADCHPDL	RW	0	ADC High Pass Filter Disable (Right)

Table 88. CNVRTR0 Register

4.5.2. ADC High Pass Filter Enable Modes

ADCHPDR	ADCHPDL	High Pass Mode
0	0	High-pass filter enabled on left and right channels
0	1	High-pass filter disabled on left channel, enabled on right channel
1	0	High-pass filter enabled on left channel, disabled on right channel
1	1	High-pass filter disabled on left and right channels

Table 89. ADC HPF Enable

4.6. Digital ADC Volume Control

The ADC volume can be controlled digitally, across a gain and attenuation range of -71.25dB to +24dB (0.375dB steps). The level of attenuation is specified by an eight-bit code ‘ADCVOL_x’, where ‘x’ is L, or R. The value “00000000” indicates mute; other values describe the number of 0.375dB steps above -71.25dB.

The ADCVOLU bit controls the updating of digital volume control data. When ADCVOLU is written as ‘0’, the ADC digital volume is immediately updated with the ADCVOL_L data when the Left ADC Digital Volume register is written. When ADCVOLU is set to ‘1’, the ADCVOL_L data is held in an internal holding register until the Right ADC Digital Volume Register is written.

4.6.1. ADC Digital Volume Control Register

Register Address	Bit	Label	Type	Default	Description
R6 (06h) ADCVOLL	7:0	ADCVOL_L [7:0]	RW	10111111 (0dB)	Left ADC Digital Volume Control 0000 0000 = Digital Mute 0000 0001 = -71.25dB 0000 0010 = -70.875dB ... 0.375dB steps up to 1111 1111 = +24dB Note: If ADCVOLU is set, this setting will take effect after the next write to the Right Input Volume register.
R7 (07h) ADCVOLR	7:0	ADCVOL_R [7:0]	RW	10111111 (0dB)	Right ADC Digital Volume Control 0000 0000 = Digital Mute 0000 0001 = -71.25dB 0000 0010 = -70.875dB ... 0.375dB steps up to 1111 1111 = +24dB

Table 90. ADCVOLL/ADCVOLR Register

4.7. Automatic Level Control (ALC)

The TSCS42xx has an automatic level control to achieve constant recording volume across a range of input signal levels. The device uses a digital peak detector to monitor and adjusts the PGA gain to provide a constant signal level at the ADC input. A range of adjustment between –6dB and –28.5dB (relative to ADC full scale) can be selected. The device provides programmable attack, hold, and decay times to smooth adjustments. The level control also features a peak limiter to prevent clipping when the ADC input exceeds a threshold. Note that if the ALC is enabled, the input volume controls are ignored.

4.7.1. ALC Operation

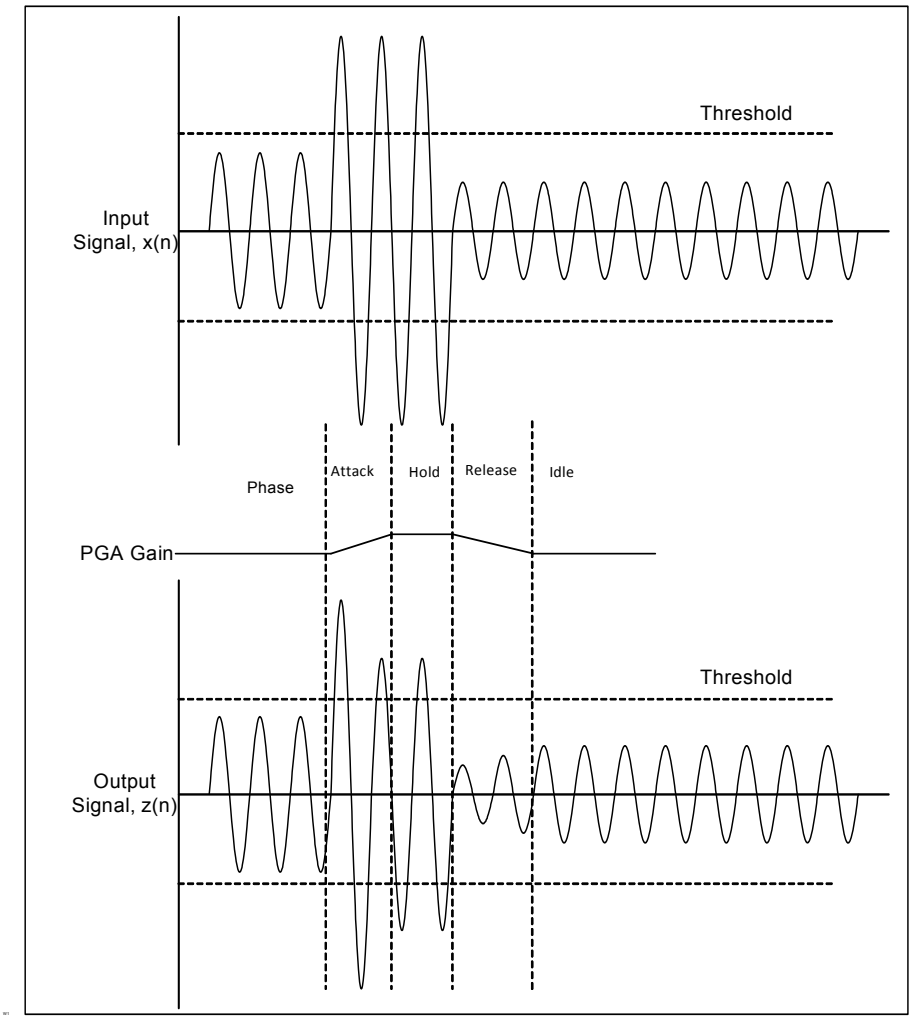

Figure 19. ALC Operation

When ALC is enabled, the recording volume target can be programmed between –6dB and –28.5dB (relative to ADC full scale). The ALC will attempt to keep the ADC input level to within +/-0.5dB of the target level. An upper limit for the PGA gain can also be imposed, using the MAXGAIN control bits.

Hold time specifies the delay between detecting a peak level being below target, and the PGA gain beginning to ramp up. It is specified as 2^n*2.67mS, enabling a range between 0mS and over 40s.; ramp-down begins immediately if the signal level is above the target.

Decay (Gain Ramp-Up) Time is the time that it takes for the PGA to ramp up across 90% of its range. The time is 2^n*24mS. The time required for the recording level to return to its target value therefore depends on the decay time and on the gain adjustment required.

Attack (Gain Ramp-Down) Time is the time that it takes for the PGA to ramp down across 90% of its range. Time is specified as 2^n*24mS. The time required for the recording level to return to its target value depends on both the attack time and on the gain adjustment required.

When operating in stereo, the peak detector takes the maximum of left and right channel peak values, and both PGAs use the same gain setting, to preserve the stereo image. If the ALC function is only enabled on one channel, only one PGA is controlled by the ALC mechanism, and the other channel runs independently using the PGA gain set through the control registers.

If one ADC channel is unused, the peak detector will ignore that channel.

The ALC function can operate when the two ADC outputs are mixed to mono in the digital domain or in the analog domain.

4.7.2. ALC Control Register

Register Address	Bit	Label	Type	Default	Description
R14 (0Eh) ALC0	7:3	RSVD	R	00000	Reserved
	2	ALC MODE	RW	0	0: ALC Mode 1: Limiter mode
	1:0	ALCSEL [1:0]	RW	00 (OFF)	ALC function select 00 = ALC off (PGA gain set by register) 01 = Right channel only 10 = Left channel only 11 = Stereo (PGA registers unused) Note: ensure that LINVOL and RINVOL settings (reg. 0 and 1) are the same before entering this mode.
R15 (0Fh) ALC1	7	RSVD	R	0	Reserved
	6:4	MAXGAIN [2:0]	RW	111 (+30dB)	Set Maximum Gain of PGA 111: +30dB 110: +24dB ….(-6dB steps) 001: -6dB 000: -12dB
	3:0	ALCL [3:0]	RW	1011 (-12dB)	ALC target – sets signal level at ADC input 0000 = -28.5dB fs 0001 = -27.0dB fs … (1.5dB steps) 1110 = -7.5dB fs 1111 = -6dB fs
R16 (10h) ALC2	7	RSVD	R	0	Reserved
	6:4	MINGAIN	RW	000	Sets the minimum gain of the PGA 000 = -17.25db 001 = -11.25 ... 110 = +18.75dB 111 = +24.75db where each value represents a 6dB step.
	3:0	HLD [3:0]	RW	0000 (0ms)	ALC hold time before gain is increased. 0000 = 0ms 0001 = 2.67ms 0010 = 5.33ms … (time doubles with every step) 1111 = 43.691s
R17 (11h) ALC3	7:4	DCY [3:0]	RW	0011 (192ms)	ALC decay (gain ramp-up) time 0000 = 24ms 0001 = 48ms 0010 = 96ms … (time doubles with every step) 1010 or higher = 24.58s
	3:0	ATK [3:0]	RW	0010 (24ms)	ALC attack (gain ramp-down) time 0000 = 6ms 0001 = 12ms 0010 = 24ms … (time doubles with every step) 1010 or higher = 6.14s

Table 91. ALC0/1/2/3 Registers

4.7.3. Peak Limiter

To prevent clipping, the ALC circuit also includes a limiter function. If the ADC input signal exceeds 87.5% of full scale (–1.16dB), the PGA gain is ramped down at the maximum attack rate, until the signal level falls below 87.5% of full scale. This function is automatically enabled whenever the ALC is enabled.

4.7.4. Input Threshold

To avoid hissing during quiet periods, the TSCS42xx has an input threshold noise gate function that compares the signal level at the inputs to a noise gate threshold. Below the threshold, the programmable gain can be held , or the ADC output can be muted. The threshold can be adjusted in increments of 1.5dB.

The noise gate activates when the signal-level at the input pin is less than the Noise Gate Threshold (NGTH) setting.

The ADC output can be muted. Alternatively, the PGA gain can be held .

The threshold is adjusted in 1.5dB steps. The noise gate only works in conjunction with the ALC, and always operates on the same channel(s) as the ALC.

4.7.5. Noise Gate Control Register

Register Address	Bit	Label	Type	Default	Description
R12 (12h) NGATE	7:3	NGTH [4:0]	RW	00000	Noise gate threshold (compared to ADC full-scale range) 00000 -76.5dBfs 00001 -75dBfs … 1.5 dB steps 11110 -31.5dBfs 11111 -30dBfs
	2:1	NGG [1:0]	RW	00	Noise gate type X0 = PGA gain held constant 01 = mute ADC output 11 = reserved (do not use this setting)
	0	NGAT	RW	0	Noise gate function enable 1 = enable 0 = disable

Table 92. NGATE Register

4.8. Digital Microphone Support

Line Input 3 may be an analog line (mic) or digital microphone input depending on the part option.

The digital microphone interface permits connection of a digital microphone(s) to the CODEC via the DMIC_DAT, and DMIC_CLK 2-pin interface. DMIC_DAT is an input that carries individual channels of digital microphone data to the ADC. In the event that a single microphone is used, the data is ported to both ADC channels. This mode is selected using a control bit and the left time slot is copied to the ADC left and right inputs.

The DMIC_CLK output is synchronous to the internal master (DSP) clock and is adjustable in 4 steps. Each step provides a clock that is a multiple of the chosen ADC base rate and modulator rate.The default frequency is 320/3 times the ADC base rate for 32KHz, and 80 times the base rate for 44.1KHz and 48KHz base rates.

4.8.1. DMIC Clock

SDM Rate	DMRate [1:0]	Base Rate	DSPCLK	DMIC_CLK divisor	DMIC_CLK
Full	00	32 KHz	40.960 MHz	12	3.413333 MHz
		44.1 KHz	56.448 MHz	16	3.528 MHz
		48 KHz	61.440 MHz	16	3.84 MHz
	01	32 KHz	40.960 MHz	16	2.56 Mhz
		44.1 KHz	56.448 MHz	20	2.8224 MHz
		48 KHz	61.440 MHz	20	3.072 MHz
	10	32 KHz	40.960 MHz	20	2.048 Mhz
		44.1 KHz	56.448 MHz	24	2.352 MHz
		48 KHz	61.440 MHz	24	2.56 MHz
	11	32 KHz	40.960 MHz	24	1.706667 Mhz
		44.1 KHz	56.448 MHz	32	1.764 MHz
		48 KHz	61.440 MHz	32	1.92 MHz
Half	00	32 KHz	40.960 MHz	16	2.56 MHz
		44.1 KHz	56.448 MHz	16	3.528 MHz
		48 KHz	61.440 MHz	16	3.84 MHz
	01	32 KHz	40.960 MHz	24	1.706667 MHz
		44.1 KHz	56.448 MHz	24	2.352 MHz
		48 KHz	61.440 MHz	24	2.56 MHz
	10	32 KHz	40.960 MHz	32	1.28 MHz
		44.1 KHz	56.448 MHz	32	1.764 MHz
		48 KHz	61.440 MHz	32	1.92 MHz
	11	32 KHz	40.960 MHz	40	1.024 MHz
		44.1 KHz	56.448 MHz	40	1.4112 MHz
		48 KHz	61.440 MHz	40	1.536 MHz

Table 93. DMIC Clock

The two DMIC data inputs are shown connected to the ADCs through the same multiplexors as the analog ports. Although the internal implementation is different between the analog ports and the digital microphones, the functionality is the same. In most cases, the default values for the DMIC clock rate and data sample phase will be appropriate and an audio driver will be able to configure and use the digital microphones exactly like an analog microphone.

To conserve power, the analog portion of the ADC will be turned off if the D-mic input is selected. When switching from the digital microphone to an analog input to the ADC, the analog portion of the ADC will be brought back to a full power state and allowed to stabilize before switching from the digital microphone to the analog input. This should take less than 10mS.

If the ADC path is powered down, the DMIC_CLK output will be driven low to place the DMIC element into a low power state. (Many digital microphones will enter a low power state if the clock input is held at a DC level or toggled at a slow rate.)

4.8.2. Digital Mic Configuration

The TSCS42xx codec supports the following digital microphone configurations:

Digital Mics	Data Sample	Notes
0	N/A	No Digital Microphones
1	Single Edge	When using a microphone that supports multiplexed operation (2-mics can share a common data line), configure the microphone for "Left" and select mono operation. "Left" D-mic data is used for ADC left and right channels.
2	Double Edge	External logic required to support sampling on a single Digital Mic pin channel on rising edge and second Digital Mic right channel on falling edge of DMIC_CLK for those digital microphones that don't support alternative clock edge (multiplexed output) capability.

Table 94. Valid Digital Mic Configuration

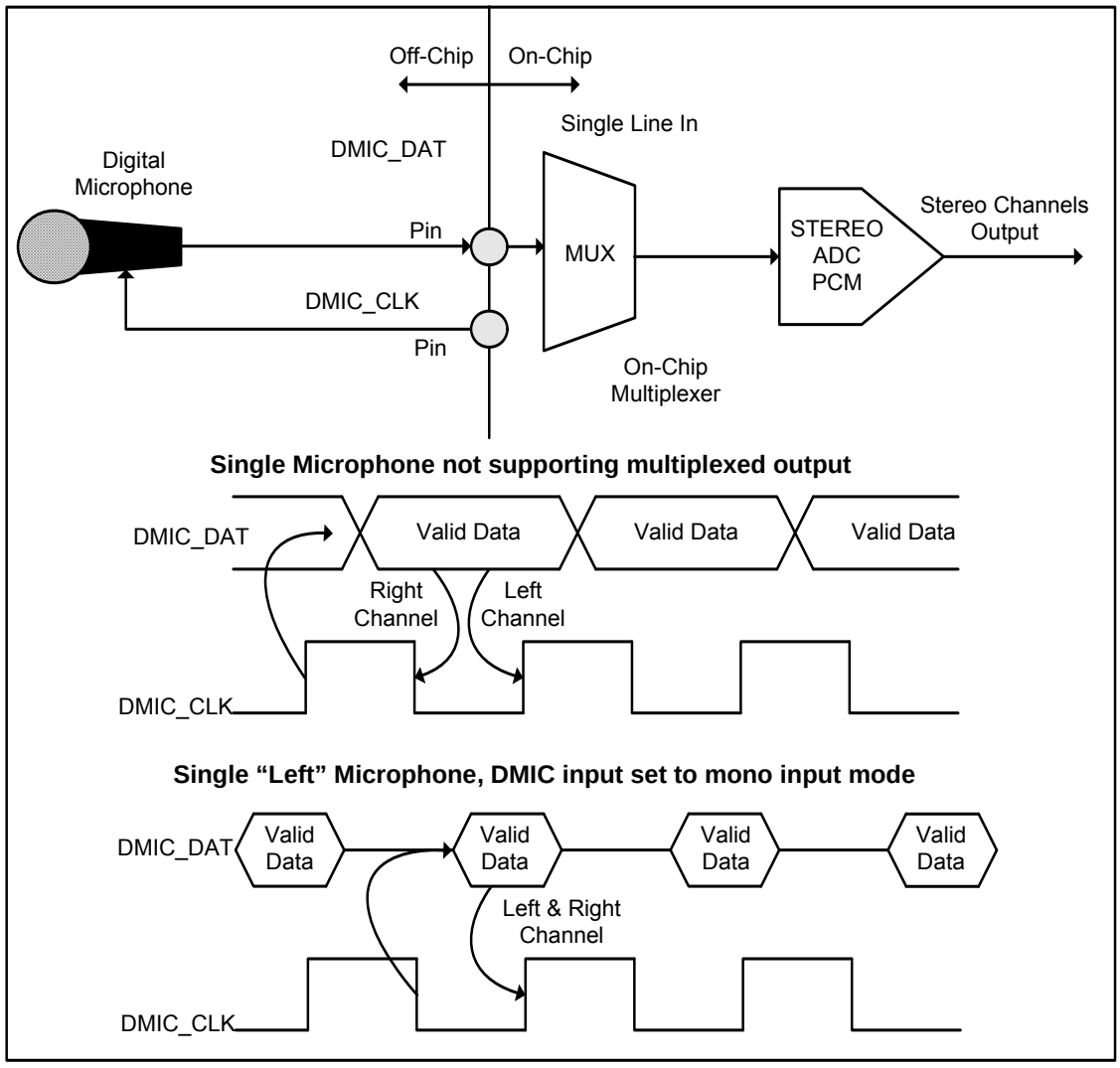

Figure 20. Single Digital Microphone (data is ported to both left and right channels

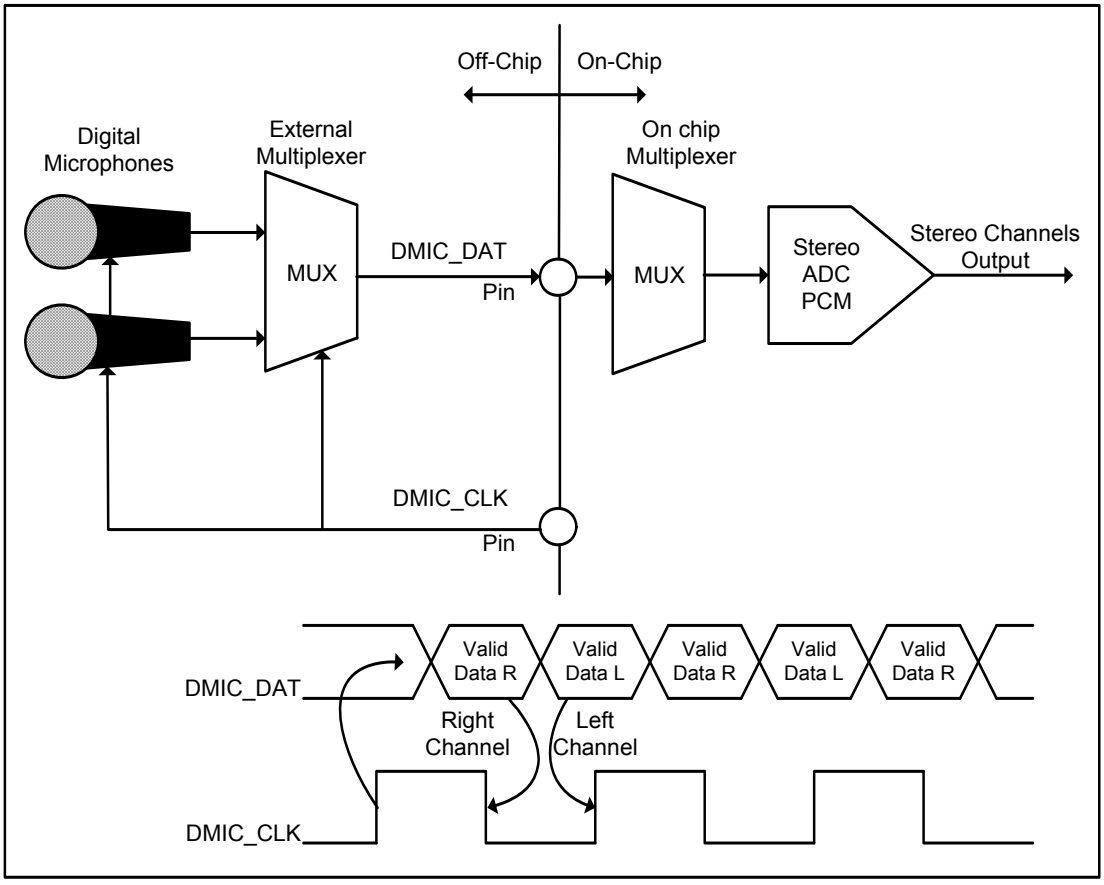

Figure 21. Stereo Digital Microphone Configuration

Note: Some digital microphones support data on either DMIC_CLK edge, in which case, an external mux, as shown in Figure 21 above may not be required.

5. DIGITAL AUDIO AND CONTROL INTERFACES

5.1. Data Interface

For digital audio data, the TSCS42xx uses six pins to input and output digital audio data.

- ADCDOUT: ADC data output
- ADCLRCLK: ADC data alignment clock
- ADCBCLK: Bit clock, for synchronization
- DACDIN: DAC data input
- DACLRCLK: DAC data alignment clock
- DACBCLK: Bit clock, for synchronization

The clock signals ADCBCLK, ADCLRCLK, DACBCLK, and DACLRCK are outputs when the TSCS42xx operates as a master; they are inputs when it is a slave. Four different data formats are supported:

- Left justified
- Right justified
- I^2S
- PCM Bluetooth

All of these modes are MSB first.

5.2. Master and Slave Mode Operation

The TSCS42xx can be used as either a master or slave device, selected by the MS Bit. When operating as a master, the TSCS42xx generates ADCBCLK, ADCLRCLK, DACBCLK and DACLRCLK and controls sequencing of the data transfer the data pins. In slave mode, the TSCS42xx provides data aligned to clocks it receives.

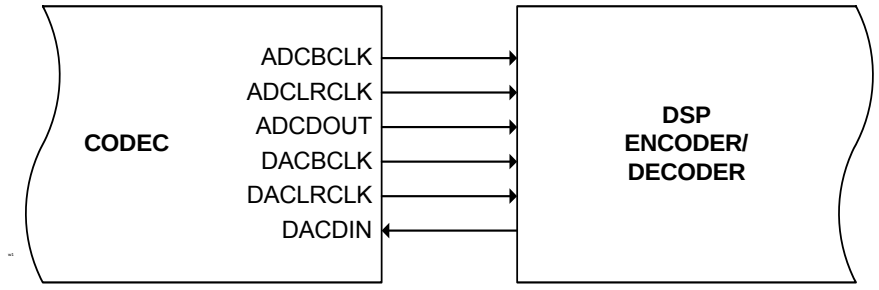

Figure 22. Master mode

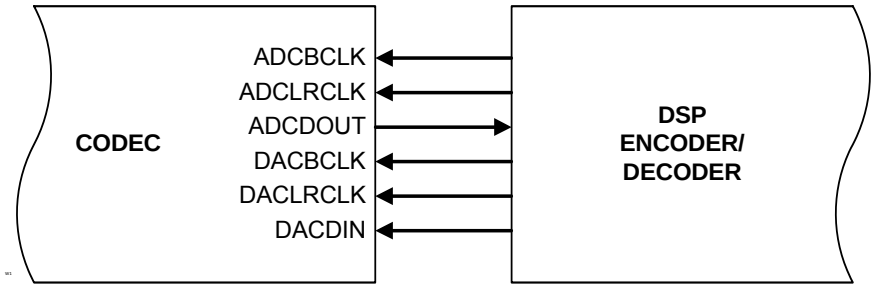

Figure 23. Slave mode

5.3. Audio Data Formats

The TSCS42xx supports 4 common audio interface formats and programmable clocking that provides broad compatibility with DSPs, Consumer Audio and Video SOCs, FPGAs, handset chipsets, and many other products.

In all modes, depending on word length, BCLK frequency and sample rate, there may be unused BCLK cycles before each LRCLK transition. If the converter word length is smaller than the number of clocks per sample in the frame then the DAC will ignore (truncate) the extra bits while the ADC will zero pad the output data. If the converter word length chosen is larger than the number of clocks available per sample in the frame, the ADC data will be truncated to fit the frame and the DAC data will be zero padded.

5.3.1. PCM Interface

PCM Mode is a time-division multiplexed format. The PCM interface operates in either slave or master mode. Data is sampled on the falling edge of the bit clock and transmitted on the rising edge. A control bit selects between a delayed and non-delayed data timing relative to the start of the frame sync. The LRCLK is one bit clock long for a Short Frame Sync and one slot wide for a Long Frame Sync. PCM mode supports mono and stereo formats

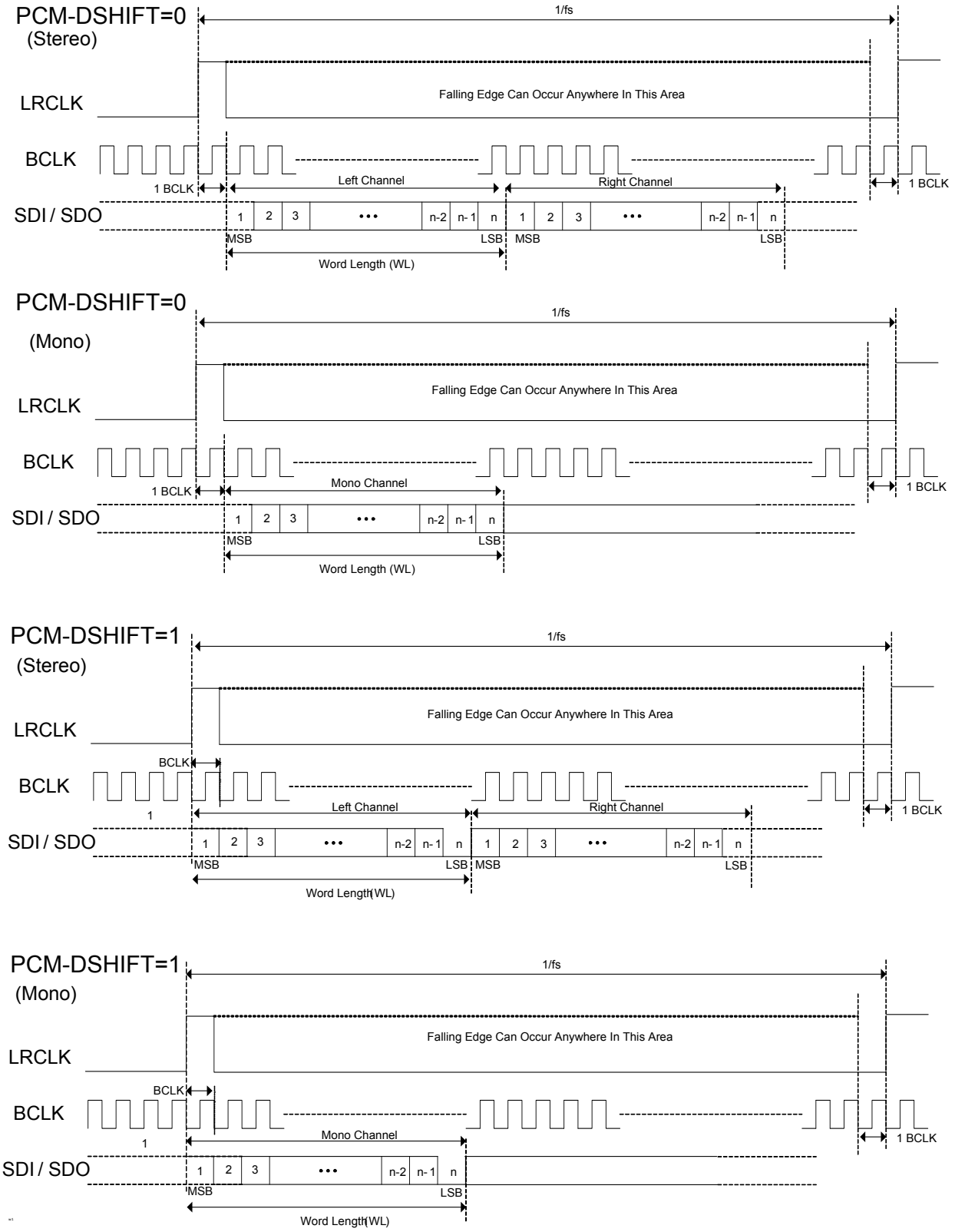

Figure 24. PCM Audio Interface

For digital audio data, the TSCS42xx uses below six pins for PCM audio interface.

- ADCDOUT: PCM data out
- ADCCLRCLK: ADC PCM data alignment
- ADCBCLK: ADC PCM Bit clock, for synchronization
- DACDIN: PCM data in
- DACLRCLK:DAC PCM data alignment
- DACBCLK: DAC PCM Bit clock, for synchronization

5.3.1.1. PCM control Registers

	Bit	Label	Read/ Write	Reset Value	Description
R195(C3h) ADCPCMCTL1	7:5	GAINCODE	RW	0	PCM gain code to be sent
	4	GAINENABLE	RW	0	PCM gain code enable-if 1, replace lsb bits of data if 0, normal mode
	3	BDELAYO	RW	0	output Bit clock delay, 0 = data not delayed, 1 = data delayed.
	2	PCMFL	RW	0	PCM Frame Length in master mode, 0 = 128 bits peer frame, 1 = 256 bits per frame
	1	SLSYNC	RW	0	short-Long Frame Sync, 0 = one clock wide, 1 = one slot wide
	0		R	0	Reserved

Table 95. ADCPCMCTL1 Register

	Bit	Label	Read/ Write	Reset Value	Description
R196(C4h) ADCPCMCTL2	7	RSVD	R	0	Reserved
	6	PCMMOMP	RW	0	PCM mono output mode, 0- When number of slots = 1, select left data for slot0, 1-select left data for slot0 = 1, select right data for slot0.
	5	PCMSOP	RW	0	Number of Active Slots per PCM Output Frame, 0 = one, 1 = two
	4:3	PCMDSSP	RW	0	PCM Data Slots Size, 00 = 16 bit, 01 =24 bit, 10 = 32 bit, 11=Reserved
	2		R	0	Reserved
	1		R	0	Reserved
	0		R	0	Reserved

Table 96. ADCPCMCTL2 Register

	Bit	Label	Read/ Write	Reset Value	Description
R197(C5h) DACPCMCTL1	7:5		R	0	Reserved
	4		R	0	Reserved
	3	BDELAYI	RW	0	Input Bit clock delay, 0 = data not delayed, 1 = data delayed.
	2	PCMFL	RW	0	PCM Frame Length in master mode, 0 = 128 bits peer frame, 1 = 256 bits per frame
	1	SLSYNC	RW	0	short-Long Frame Sync, 0 = one clock wide, 1 = one slot wide
	0		R	0	Reserved

Table 97. DACPCMCTL1Register

	Bit	Label	Read/ Write	Reset Value	Description
R198(C6h) DACPCMCTL2	7	PCMFORMAT	RW	0	DAC input path set to PCM format if 1
	6	PCMMIM	RW	0	PCM mono input mode, 0- When number of slots = 1, select left data for slot0, 1-select left data for slot0 = 1, select right data for slot0.
	5	PCMSI	RW	0	Number of Active Slots per PCM Output Frame, 0 = one, 1 = two
	4:3	PCMDSS	RW	0	PCM Data Slots Size, 00 = 16 bit, 01 =24 bit, 10 = 32 bit, 11=Reserved
	2	PCMSIGNEXT	RW	0	Data is received in 13bit sign extended mode, left shift by 3 and pad with 0s
	1	PCM13MODE	RW	0	Data is received with un-used gain bits, set these to 0
	0		R	0	Reserved

Table 98. DACPCMCTL2 Register

5.3.2. Left Justified Audio Interface

In Left Justified mode, the MSB is available on the first rising edge of BCLK following a LRCLK transition. The other bits are then transmitted in order. The LRCLK signal is high when left channel data is present and low when right channel data is present.

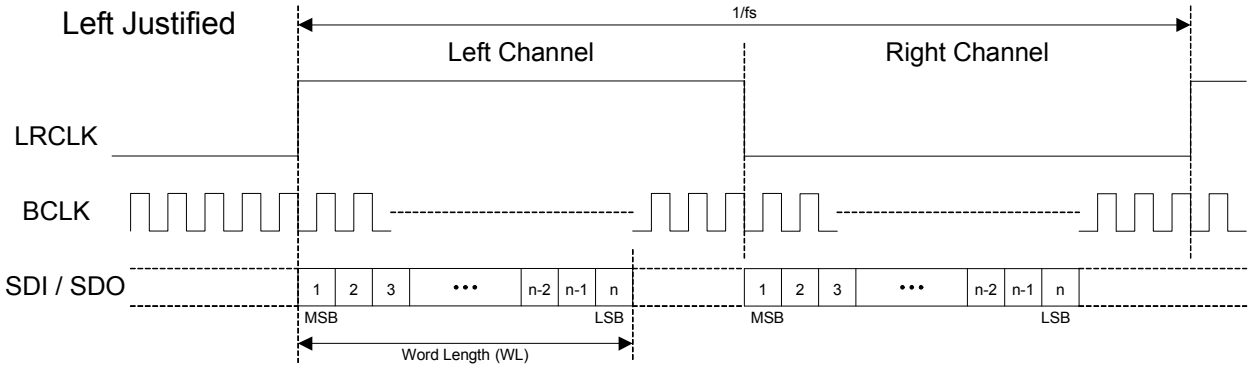

Figure 25. Left Justified Audio Interface (assuming n-bit word length)

5.3.3. Right Justified Audio Interface (assuming n-bit word length)

In Right Justified mode, the LSB is available on the last rising edge of BCLK before a LRCLK transition. All other bits are transmitted in order. The LRCLK signal is high when left channel data is present and low when right channel data is present.

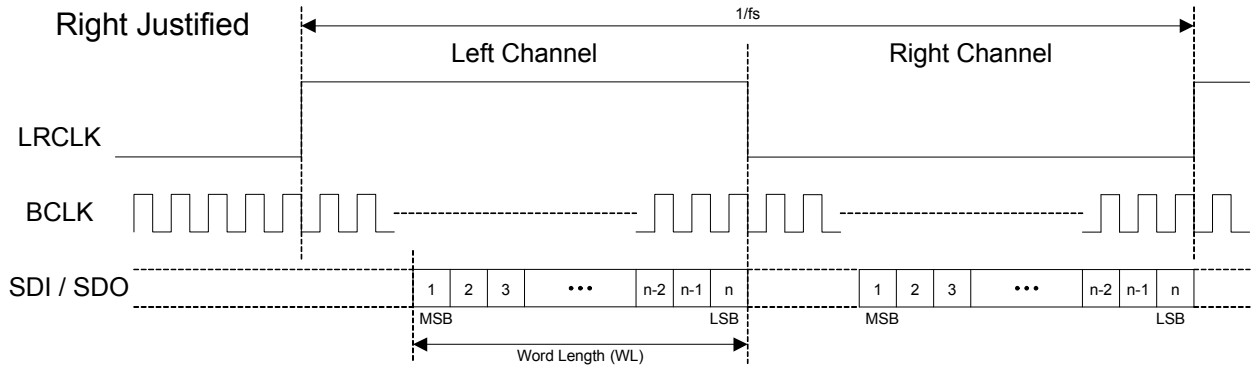

Figure 26. Right Justified Audio Interface (assuming n-bit word length)

5.3.4. I²S Format Audio Interface

In I²S mode, the MSB is available on the second rising edge of BCLK following a LRCLK transition. The other bits up to the LSB are then transmitted in order.

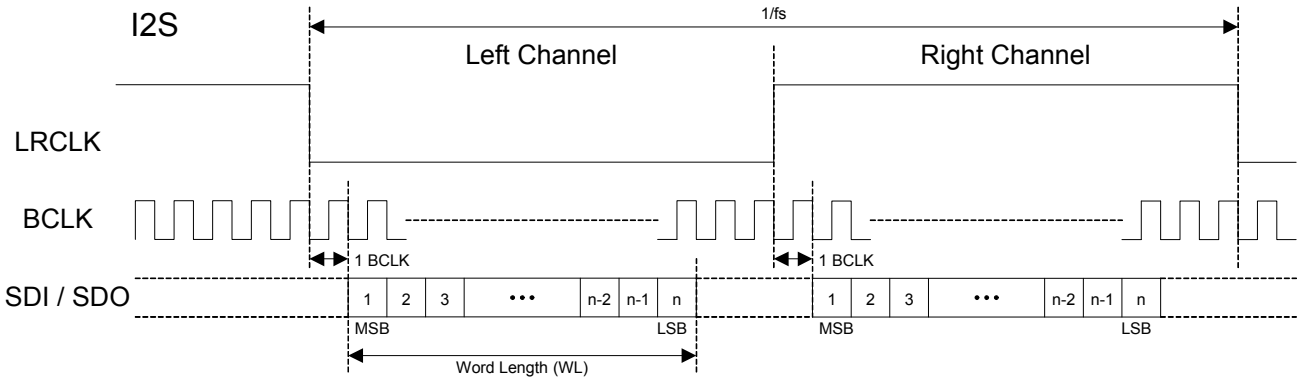

Figure 27. I²S Justified Audio Interface (assuming n-bit word length)

5.4. Audio Data Interface Registers

5.4.1. I2S Interface Control Registers

Register Address	Bit	Label	Default	Description
R19(13h)AIC1	7	RESERVED	0h	Reserved
	6	BCLKINV	0h	BCLK Invert (master and slave modes): 1 = BCLK inverted 0 = BCLK not inverted
	5	MS	0h	Master/Slave Control: 0 = Slave; 1 = Master
	4	LRP	0h	LRClk Polarity: 0 = Not inverted; 1 = Inverted
	3:2	WL	2h	Audio Data Word Length: 0h = 16 bits; 1h = 20 bits; 2h = 24 bits; 3h = 32 bits
	1:0	FORMAT	2h	Audio Data Format: 0h = Right justified; 1h = Left justified; 2h = I2S 3h = Reserved

Table 99. AIC1 Register

5.4.2. Digital Mic Interface Control

Register Address	Bit	Label	Type	Default	Description
R36 (24h) DMICCTL	7	DMicEn	RW	0	Digital Microphone Enable 0 = DMIC interface is disabled (DMIC_CLK low, DMIC muted) 1 = DMIC interface is enabled
	6:5	RSVD	R	00	Reserved
	4	DMono	RW	0	0 = stereo operation, 1 = mono operation (left channel duplicated on right)
	3:2	DMPhAdj[1:0]	RW	00	Selects when the D-Mic data is latched relative to the DMIC_CLK. 00 = Left data rising edge / right data falling edge 01 = Left data center of high / right data center of low 10 = Left data falling edge / right data rising edge 11 = Left data center of low / right data center of high
	1:0	DMRate[1:0]	RW	00	Selects the DMIC clock rate: See table 93

Table 100. DMICCTL Register

5.4.3. Audio Interface Output Tri-state

TRI is used to tri-state the ADCDOUT, ADCLRCLK, DACLRCLK, ADCBCLK, and DACBCLK pins. In Slave mode (MS bit=0) only ADCDOUT will be tri-stated since the other pins are configured as inputs. The Tri-stated pins are pulled low with an internal pull-down resistor unless that resistor is disabled.

Register Address	Bit	Label	Type	Default	Description
R20 (14h) AIC2	7:6	DACDSEL[1:0]	RW	00	00: left DAC = left I2S data; right DAC = right I2S data 01: left DAC = left I2S data; right DAC = left I2S data 10: left DAC = right I2S data; right DAC = right I2S data 11: left DAC = right I2S data; right DAC = left I2S data
	5:4	ADCDSEL[1:0]	RW	00	00: left I2S data = left ADC; right I2S data = right ADC 01: left I2S data = left ADC; right I2S data = left ADC 10: left I2S data = right ADC; right I2S data = right ADC 11: left I2S data = right ADC; right I2S data = left ADC
	3	TRI	RW	0	Tri-states ADCDOUT, ADCLRCLK, DACLRCLK, ADCBCLK, and DACBCLK pins. 0 = ADCDOUT is an output, ADCLRCLK, DACLRCLK, ADCBCLK, and DACBCLK are inputs (slave mode) or outputs (master mode) 1 = ADCDOUT, ADCLRCLK, DACLRCLK, ADCBCLK, and DACBCLK are high impedance
	2:0	BLRCM[2:0]	RW	000	Bitclock and LRClock mode. See Table Below

Table 101. AIC2 Register

5.4.4. Bit Clock and LR Clock Mode Controls

Although the DAC and ADC interfaces implement separate Bit Clock and LR Clock pins, it is also possible to share one or both of the clocks.

the following restrictions must be observed when the BCLK from one path (DAC or ADC) is combined with the LRCLK from the other path (ADC or DAC) as described by the Bit Clock and LR Clock Mode Selection table below:

1. Both the DAC and ADC must be programmed for the same sample rate
2. Both the DAC and ADC must be programmed for the same number of clocks per frame
3. When in slave mode, the DAC and ADC data must be aligned relative to the provided BCLK and LRCLK (this is guaranteed in master mode)
4. The DAC and ADC must be powered down when changing the BLRCM mode
5. If sharing the BCLK from one path (DAC or ADC) and the LRCLK from the other path (ADC or DAC), shut down both the DAC and ADC before programming the sample rate and clocks per frame for either. (Again, both must match.)

MS	BLRCM [2:0]	MODE[1]	DAC BCLK	ADC BCLK	DAC LRCLK	ADC LRCLK
0	000	Independent	Input for playback path	input for record path	Input for playback path	input for record path
0	001	Independent	Input for playback path	input for record path	Input for playback path	input for record path
0	010	Shared BCLK (DAC)	Input for playback and record	unused	Input for playback path	input for record path
0	011	Shared BCLK & LRCLK (DAC)	Input for playback and record	unused	Input for playback and record	unused
0	100	Shared BCLK (DAC) & LRCLK (ADC)	Input for playback and record	unused	unused	Input for playback and record
0	101	Shared BCLK (ADC)	unused	Input for playback and record	Input for playback path	input for record path
0	110	Shared BCLK (ADC) & LRCLK (DAC)	unused	Input for playback and record	Input for playback and record	unused
0	111	Shared BCLK & LRCLK (ADC)	unused	Input for playback and record	unused	Input for playback and record
1	000	Independent (off if converter off)	Output for playback path (off when DACs off)[2]	Output for record path (Off when ADC off)[3]	Output for playback path (off when DACs off)	Output for record path (off when ADCs off)
1	001	Independent (off if all converters off)	Output for playback path (off when DACs and ADCs off)	Output for record path (off when DACs and ADCs off)	Output for playback path (off when DACs and ADCs off)	Output for record path (off when DACs and ADCs off)
1	010	Shared BCLK (DAC)	Output for playback and record (stays on if either DAC or ADC on)	unused (off)	Output for playback path (Off if DAC is off)	Output for record path (off when ADCs off)
1	011	Shared BCLK & LRCLK (DAC)	Output for playback and record (stays on if either DAC or ADC on)	unused (off)	Output for playback and record (stays on if either DAC or ADC on)	unused (off)
1	100	Shared BCLK(DAC)& LRCLK(ADC)	Output for playback and record (stays on if either DAC or ADC on)	unused (off)	unused (off)	Output for playback and record (stays on if either DAC or ADC on)
1	101	Shared BCLK (ADC)	unused (off)	Output for playback and record (stays on if either DAC or ADC on)	Output for playback path (Off if DAC is off)	Output for record path (off when ADCs off)
1	110	Shared BCLK(ADC)& LRCLK(DAC)	unused (off)	Output for playback and record (stays on if either DAC or ADC on)	Output for playback and record (stays on if either DAC or ADC on)	unused (off)
1	111	Shared BCLK & LRCLK(ADC)	unused (off)	Output for playback and record (stays on if either DAC or ADC on)	unused (off)	Output for playback and record (stays on if either DAC or ADC on)

Table 102. Bit Clock and LR Clock Mode Selection

1.When sharing both the BCLK and LRCLK between the DAC and ADC interfaces, both the DAC and ADC must be programmed for the same rate, the same number of clocks per frame, and data must be aligned the same with respect to LRCLK. Disable all converters before changing modes.

2.DAC (playback path) is off when HPL, HPR, SPKL, and SPKR power states are off.

3.ADC is off when ADCL, and ADCR power states are off (PGA, D2S, Boost power states are not considered.)

5.4.5. ADC Output Pin State

Tri-state (TRI)	Record Path Power State	ADC Data Out Pull-down (ADOPDD)	ADC Data Out State
0	Off	0	Off, pulled-low
	Off	1	Off, floating
	On	NA	Active
1	NA	0	Off, pulled-low
	NA	1	Off, floating

Table 103. ADC Data Output pin state

5.4.6. Audio Interface Control 3 Register

Register Address	Bit	Label	Type	Default	Description
R21 (15h) AIC3	7:6	RSVD	R	0	Reserved
	5	ADOPDD	RW	0	ADCDOUT Pull-Down Disable 0 = Pull-Down active when tri-stated or the ADC path is powered down. 1 = Pull-Down always disabled
	4	ALRPDD	RW	0	ADCLRCLK Pull-Down Disable 0 = Pull-Down active when configured as input 1 = Pull-Down always disabled
	3	ABCPDD	RW	0	ADCBCLK Pull-Down Disable 0 = Pull-Down active when configured as input 1 = Pull-Down always disabled
	2	DDIPDD	RW	0	DACDIN Pull-Down Disable 0 = Pull-Down active 1 = Pull-Down always disabled
	1	DLRPDD	RW	0	DACLRCLK Pull-Down Disable 0 = Pull-Down active when configured as input 1 = Pull-Down always disabled
	0	DBCPDD	RW	0	DACBCLK Pull-Down Disable 0 = Pull-Down active when configured as input 1 = Pull-Down always disabled

Table 104. AIC3 Register

5.4.7. Bit Clock Mode

The default master mode bit clock generator automatically produces a bit clock frequency based on the sample rate and word length. When enabled by setting the appropriate BCM bits, the bit clock mode (BCM) function overrides the default master mode bit clock generator to produce the bit clock frequency shown below: Note that selecting a word length of 24-bits in Auto mode generates 64 clocks per frame (64fs)

.

Register Address	Bit	Label	Type	Default	Description
R23/R25 (17h/19h) ADCSR/DACSR	7:6	ABCM[1:0] DBCM[1:0]	RW	00	BCLK Frequency 00 = Auto 01 = 32 x fs 10 = 40 x fs 11 = 64 x fs

Table 105. ADCSR/ DACSR Register

The BCM mode bit clock generator produces 16, 20, or 32 bit cycles per sample.

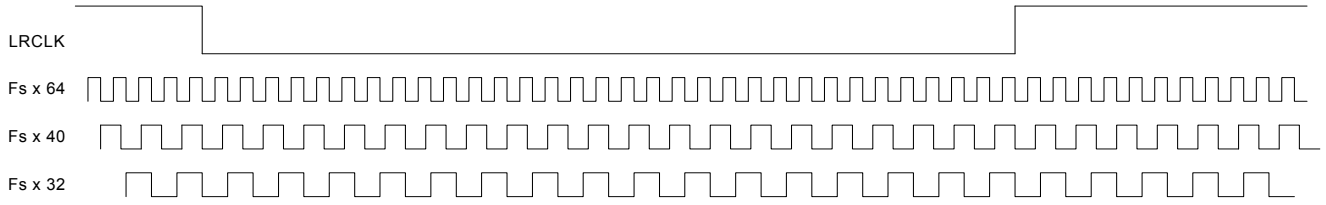

Figure 28. Bit Clock mode

Note: The clock cycles are evenly distributed throughout the frame (true multiple of LRCLK not a gated clock.)

5.5. I2C /Control Interface

The registers are accessed through a serial control interface using a multi-word protocol comprised of 8-bit words. The first 8 bits provide the device address and Read/Write flag. In a write cycle, the next 8 bits provide the register address; all subsequent words contain the data, corresponding to the 8 bits in each control register.The control interface operates using a standard 2-wire interface, as a slave device only. The TSCS42xx has 8 bit device address E2 for Analog mic version of the part and D2 for Digital mic version of the part

5.5.1. Register Write Cycle

The controller indicates the start of data transfer with a high to low transition on SDA while SCL remains high, signalling that a device address and data will follow. All devices on the 2-wire bus respond to the start condition and shift in the next eight bits on SDIN (7-bit address + Read/Write bit, MSB first). If the device address received matches the address of the TSCS42xx and the R/W bit is '0', indicating a write, then the TSCS42xx responds by pulling SDA low on the next clock pulse (ACK); otherwise, the TSCS42xx returns to the idle condition to wait for a new start condition and valid address.

Once the TSCS42xx has acknowledged a correct device address, the controller sends the TSCS42xx register address. The TSCS42xx acknowledges the register address by pulling SDA low for one clock pulse (ACK). The controller then sends a byte of data (B7 to B0), and the TSCS42xx acknowledges again by pulling SDA low.

When there is a low to high transition on SDA while SCL is high, the transfer is complete. After receiving a complete address and data sequence the TSCS42xx returns to the idle state. If a start or stop condition is detected out of sequence, the device returns to the idle condition.

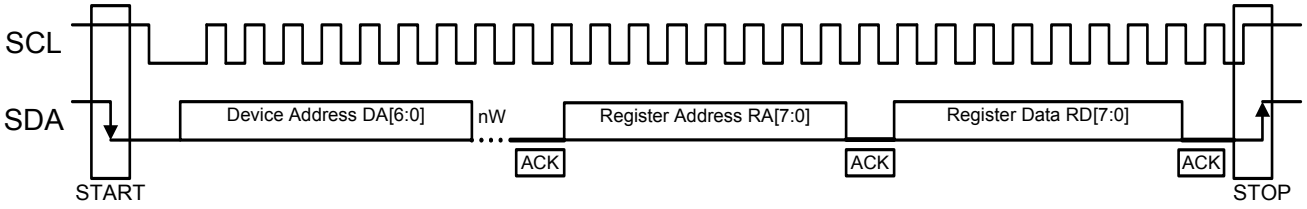

Figure 29. 2-Wire Serial Control Interface

5.5.2. Multiple Write Cycle

The controller may write more than one register within a single write cycle. To write additional registers, the controller will not generate a stop or start (repeated start) command after receiving the acknowledge for the second byte of information (register address and data). Instead the controller will continue to send bytes of data. After each byte of data is received, the register address is incremented.

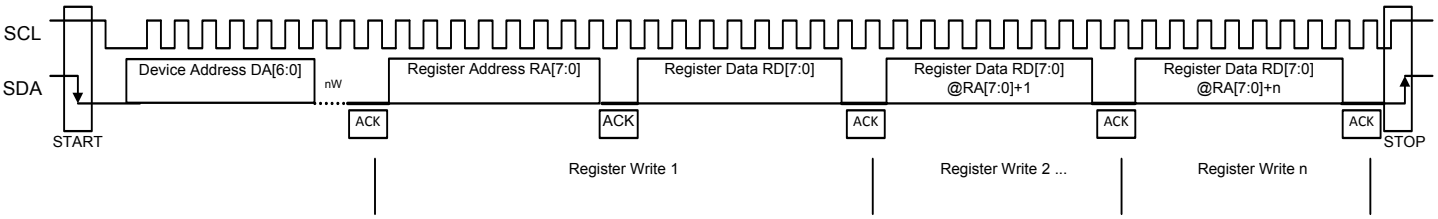

Figure 30. Multiple Write Cycle

5.5.3. Register Read Cycle

The controller indicates the start of data transfer with a high to low transition on SDA while SCL remains high, signalling that a device address and data will follow. If the device address received matches the address of the TSCS42xx and the R/W bit is ‘0’, indicating a write, then the TSCS42xx responds by pulling SDA low on the next clock pulse (ACK); otherwise, the TSCS42xx returns to the idle condition to wait for a new start condition and valid address.

Once the TSCS42xx has acknowledged a correct address, the controller sends a restart command (high to low transition on SDA while SCL remains high). The controller then re-sends the devices address with the R/W bit set to ‘1’ to indicate a read cycle.The TSCS42xx acknowledges by pulling SDA low for one clock pulse. The controller then receives a byte of register data (B7 to B0).

For a single byte transfer, the host controller will not acknowledge (high on data line) the data byte and generate a low to high transition on SDA while SCL is high, completing the transfer. If a start or stop condition is detected out of sequence, the device returns to the idle condition.

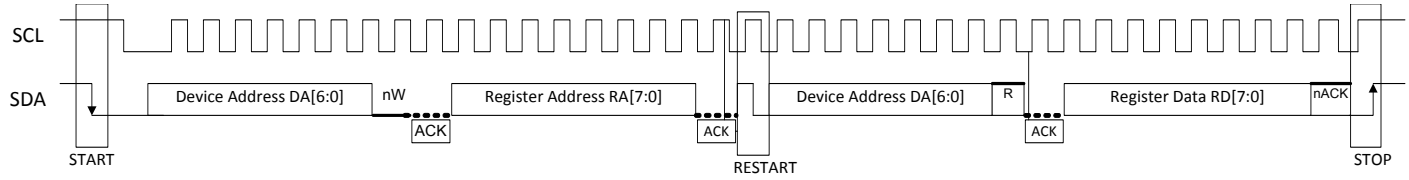

Figure 31. Read Cycle

5.5.4. Multiple Read Cycle

The controller may read more than one register within a single read cycle. To read additional registers, the controller will not generate a stop or start (repeated start) command after sending the acknowledge for the byte of data. Instead the controller will continue to provide clocks and acknowledge after each byte of received data. The codec will automatically increment the internal register address after each register has had its data successfully read (ACK from host) but will not increment the register address if the data is not received correctly by the host (nACK from host) or if the bus cycle is terminated unexpectedly (however the EQ/Filter address will be incremented even if the register address is not incremented when performing EQ/Filter RAM reads). By automatically incrementing the internal register address after each byte is read, all the internal registers of the codec may be read in a single read cycle.

Figure 32. Multiple Read Cycle

5.5.5. Device Addressing and Identification

The TSCS42xx has a default slave address of D2. However, it is sometimes necessary to use a different address. The TSCS42xx has a device address register for this purpose. The part itself has an 8-bit Identification register and an 8-bit revision register that provide device specific information for software. In addition, an 8-bit programmable subsystem ID register can allow firmware to provide a descriptive code to higher level software such as an operating system driver or application software.

5.5.6. Device Address Register

Register Address	Bit	Label	Type	Default	Description
R124 (7Ch) DEVADR	7:1	ADDR[7:1]	RW	1101001	7-bit slave address
	0	RSVD	R	0	

Table 106. DEVADRl Register

5.5.7. Device Identification Registers

Register Address	Bit	Label	Type	Default	Description
R126 (7Eh) DEVIDH	7:0	DID[15:8]	R	xxh	16-bit device identification number. Contact TSI.
R125 (7Dh) DEVIDL	7:0	DID[7:0]	R	xxh	

Table 107. DEVID H&L Registers

5.5.8. Device Revision Register

Register Address	Bit	Label	Type	Default	Description
R127 (7Fh) REVID	7:4	MAJ[3:0]	R	xh	4-bit major revision number. Contact TSI.
	3:0	MNR[3:0]	R	xh	4-bit minor revision number. Contact TSI.

Table 108. REVID Register

5.5.9. Register Reset

The TSCS42xx registers may be reset to their default values using the reset register. Writing a special, non-zero value to this register causes all other registers to assume their default states. Device status bits will not necessarily change their values depending on the state of the device.

Register Address	Bit	Label	Type	Default	Description
R128 (80h) RESET	7:0	Reset[7:0]	RW	00h	Reset register Writing a value of 85h will cause registers to assume their default values. Reading this register returns 00h

Table 109. RESET Register

6. GPIO'S

Two GPIO's are available on the GPIO1-GPIO0 pins. These GPIO pins are accessed via register bits. The general-purpose input/output (GPIO) pins can be used as either inputs or outputs. These pins are readable and can be set or read through the control interface. These pins are useful for interfacing to external hardware.

6.1. GPIO Usage Summary

GPIO Pin	Function 1	Function 2	Pull-Up Pull-Down
GPIO0	GPIO0 Register Bit	RSVD	Pull-Up
GPIO1	GPIO1 Register Bit	RSVD	Pull-Up

Table 110. GPIO Pin Usage Summary

6.2. GPIO Control Registers

6.2.1. GPIO Control 1 Register

Register Address	Bit	Label	Type	Default	Description
Reg192 (C0h) GPIOCTL1	7	RESERVED	R	0	Reserved
	6	RESERVED	R	0	Reserved
	5	GPIO1CFG	RW	0	GPIO1 Configuration 0 = GPIO1 Configured as Input/Output 1 = GPIO1 Configured as Interrupt
	4	GPIO0CFG	RW	0	GPIO0 Configuration 0 = GPIO0 Configured as Input/Output 1 = GPIO0 Configured as Interrupt
	3	RESERVED	R	0	Reserved
	2	RESERVED	R	0	Reserved
	1	GPIO1DIR	RW	0	GPIO1 Input/Output 0 = GPIO1 configured as input 1 = GPIO1 configured as output
	0	GPIO0DIR	RW	0	GPIO0 Input/Output 0 = GPIO0 configured as input 1 = GPIO0 configured as output

Table 111. GPIOCTL1 Register

6.2.2. GPIO Control 2 Register

Register Address	Bit	Label	Type	Default	Description
Reg 193 (C1h) GPIOCTL2	7:2	RESERVED	R	0	Reserved
	1	GPIO1PU	R	0	GPIO1 Pull up 0 = GPIO1 pull up enabled 1 = GPIO1 pull up disenabled
	0	GPIO0PU	R	0	GPIO0 Pull up 0 = GPIO0 pull up enabled 1 = GPIO0 pull up disenabled

Table 112. GPIOCTL2 Register

7.CLOCK GENERATION

The TSCS42xx uses two PLL to generate two high frequency reference clocks. The clock frequencies of each reference clock are based on multiples of 44.1KHz and 48KHz sample rates.The clock source for the PLL's can be the XTAL input, MCLK1 input via the XTAL_IN pin, the MCLK2 pin, or one of the I2S interface BCLK inputs. Each PLL can be independently powered down if the audio sample rates generated by that particular PLL are not required.

7.1. On-Chip PLLs

The TSCS42xx generates two high-quality, high-frequency clocks 122.880MHz and 112.896MHz. The PLL's support a wide range of input clock frequencies. Some typical frequencies are: 19.2MHz, 22MHz, 22.5792MHz, 24MHz, 24.576 MHz, 27MHz, and 36MHz. It should be noted that some input clock frequencies may not result in being able to generate the 122.880MHz and 112.896Mhz clocks exactly resulting in an error in the audio sample rate.

Audio Clocks - Each PLL generates one of two clock frequencies based on two audio sample rates.

122.880 MHz (2560 x 48 KHz)
112.896 MHz (2560 x 44.1 KHz)

It is important that the crystal oscillator and needed PLLs remain on until all audio functions, including jack detection, are disabled.

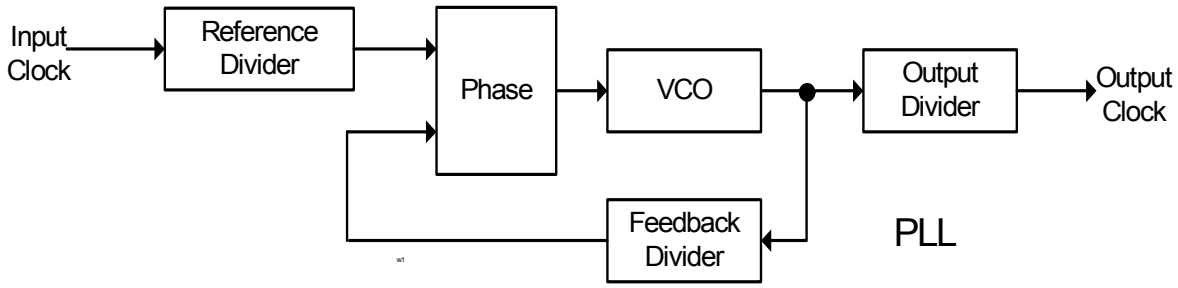

Figure 33. PLL Block Diagram

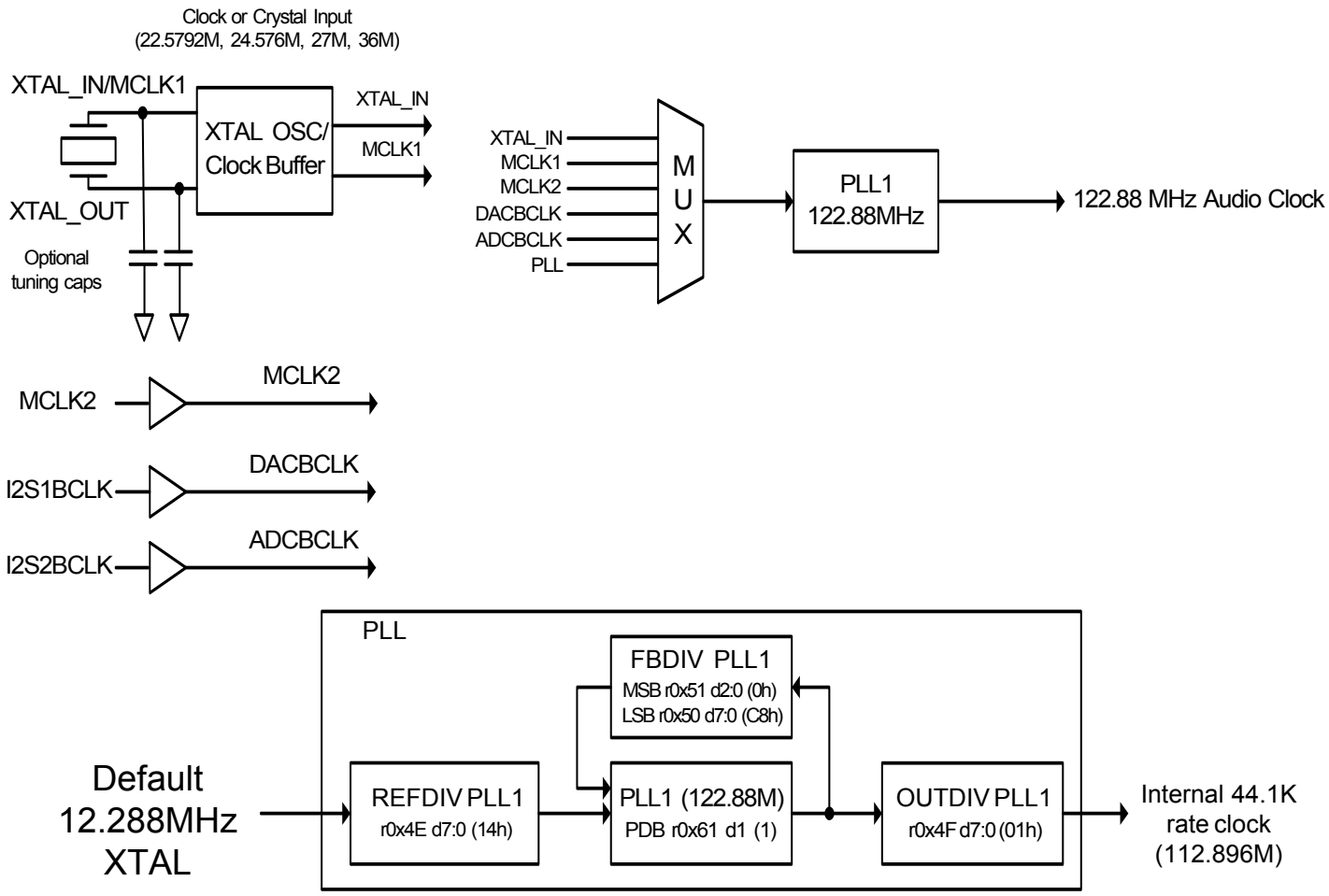

Figure 34. System Clock Diagram

7.2. System Clock Generation

The TSCS42xx supports an internal clock and audio sample rate that is selectable between 11.025KHz, 12KHz, 22.050KHz, 24KHz, 44.1KHz, 48KHz, 88.2KHz, and 96KHz. One bi-directional stereo I2S interface is available. In Master mode an internal timing generator is used to specify the audio sample rate. The sample rate specified in Master mode is independent from the internal clock rate and the specified range is 8KHz to 96KHz. A variety of sample rates based on 44.1K, 48K and 32K are supported. A highly programmable PLL enables just about any input frequency to be used.

7.2.1 PLL Dividers

The chosen input frequency is multiplied up by the PLL's to generate the required output frequencies; 122.88MHz and 112.896MHz. It should be noted that it may not always be possible to generate the required output frequencies with zero error. Some values for the PLL dividers relative a specific input frequency are shown in the table below.

OUTPUT FREQUENCY

Xtal Input	TimeBase	PLL1 Default Power (122.88MHz)							PLL2 Default Power (112.896MHz)						
MHz	77h	52h	60h	4Eh	4Fh	50h	51h	Fvco	57h	60h	53h	54h	55h	56h	Fvco
0.51200	0x01	0x22	0x04	0x01	0x03	0xD0	0x02	368.64	0x1B	0x10	0x01	0x04	0x72	0x03	451.58
0.70560	0x02	0x22	0x04	0x02	0x03	0x15	0x04	368.68	0x22	0x10	0x01	0x04	0x80	0x02	451.58
1.02400	0x03	0x22	0x04	0x02	0x03	0xD0	0x02	368.64	0x1B	0x10	0x02	0x04	0x72	0x03	451.58
1.41120	0x05	0x39	0x04	0x07	0x02	0xC3	0x04	245.75	0x1B	0x10	0x03	0x03	0xD0	0x02	338.69
1.53600	0x05	0x1A	0x04	0x02	0x03	0xE0	0x01	368.64	0x1A	0x10	0x02	0x03	0xB9	0x01	338.69
2.04800	0x07	0x22	0x04	0x04	0x03	0xD0	0x02	368.64	0x1B	0x10	0x04	0x04	0x72	0x03	451.58
2.40000	0x08	0x22	0x04	0x05	0x03	0x00	0x03	368.64	0x23	0x10	0x05	0x05	0x98	0x04	564.48
2.82240	0x0A	0x23	0x04	0x07	0x04	0xC3	0x04	491.5	0x22	0x10	0x05	0x03	0x58	0x02	338.69
3.07200	0x0B	0x22	0x04	0x07	0x03	0x48	0x03	368.64	0x1A	0x10	0x04	0x03	0xB9	0x01	338.69
5.64480	0x15	0x23	0x04	0x0E	0x04	0xC3	0x04	491.5	0x1A	0x10	0x08	0x03	0xE0	0x01	338.69
6.14400	0x17	0x1A	0x04	0x08	0x03	0xE0	0x01	368.64	0x1A	0x10	0x08	0x03	0xB9	0x01	338.69
12.00000	0x2E	0x1B	0x04	0x19	0x03	0x00	0x03	368.64	0x2A	0x10	0x19	0x05	0x98	0x04	564.48
12.28800	0x2F	0x1A	0x04	0x12	0x03	0x1C	0x02	368.64	0x22	0x10	0x20	0x03	0x72	0x03	338.69
19.20000	0x4A	0x13	0x04	0x14	0x03	0x80	0x01	368.64	0x1A	0x10	0x19	0x03	0xB9	0x01	338.69
22.00000	0x55	0x2A	0x04	0x37	0x05	0x00	0x06	614.4	0x22	0x10	0x26	0x03	0x49	0x02	338.68
22.57920	0x57	0x22	0x04	0x31	0x03	0x20	0x03	368.64	0x1A	0x10	0x1D	0x03	0xB3	0x01	338.69
24.00000	0x5D	0x13	0x04	0x19	0x03	0x80	0x01	368.64	0x1B	0x10	0x19	0x05	0x4C	0x02	564.48
24.57600	0x5F	0x13	0x04	0x1D	0x03	0xB3	0x01	368.64	0x22	0x10	0x40	0x03	0x72	0x03	338.69
25.00000	0x61	0x1B	0x04	0x37	0x03	0x2B	0x03	368.64	0x1A	0x10	0x2A	0x03	0x39	0x02	338.69
26.00000	0x65	0x23	0x04	0x41	0x05	0x00	0x06	614.4	0x1A	0x10	0x26	0x03	0xEF	0x01	338.68
27.00000	0x68	0x22	0x04	0x4B	0x03	0x00	0x04	368.64	0x2A	0x10	0x7D	0x03	0x20	0x06	338.69
36.00000	0x8C	0x1B	0x04	0x4B	0x03	0x00	0x03	368.64	0x2A	0x10	0x7D	0x03	0x98	0x04	338.69
40.00000	0x9B	0x22	0x08	0x7D	0x03	0x80	0x04	368.64	0x23	0x10	0x7D	0x05	0xE4	0x06	564.48

Table 113. Typical PLL Divider Value

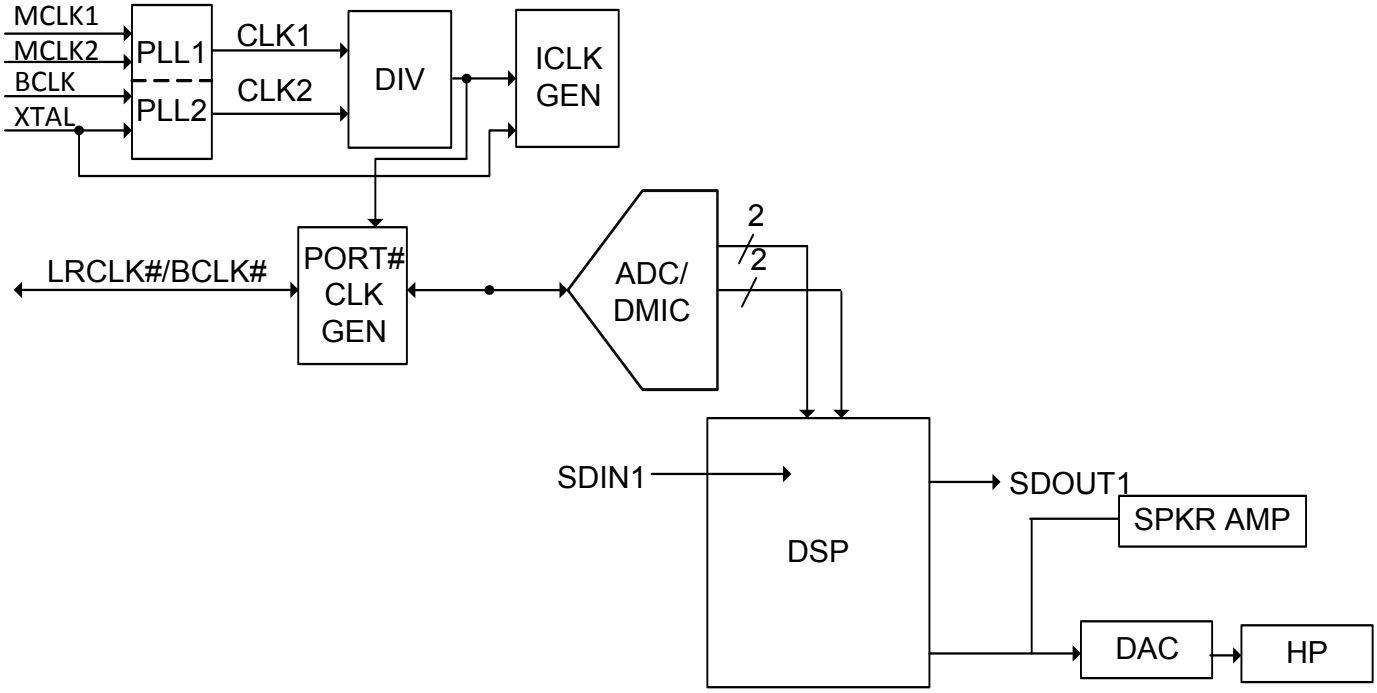

Figure 35. Simplified System Clock Block Diagram

7.2.1.1. PLL Control Register

Register Address	Bit	Label	Type	Default	Description
R96(60h) PLLCTL1B	7:6	RSVD	R	0h	Reserved
	5:4	VCOI_PLL2	RW	1h	PLL2 VCO/ICO current setting
	3:2	VCOI_PLL1	RW	1h	PLL1 VCO/ICO current setting
	1:0	RSVD	R	0h	Reserved

Table 114. PLLCTL1B Register

7.2.1.2. PLL Status Register

Register Address	Bit	Label	Type	Default	Description
R142(8Eh) PLLCTL0	7:2	RSVD	R	00h	Reserved
	1	PLL2LK	R	0h	1 = PLL2 has obtained lock
	0	PLL1LK	R	0h	1 = PLL1 has obtained lock

Table 115. PLLCTL0 Register

7.2.1.3. PLL Reference Register

Register Address	Bit	Label	Type	Default	Description
R143(8Fh) PLLREFSEL	7	RSVD	R	0h	Reserved
	6:4	PLL2_REF_SEL	RW	0h	PLL2 Reference Mux, 000 = xtal_in/mclk1; 001 = mclk2; 010 = dac_bclk; 011 = adc_bclk; 100 = pll1 output; 101 - 111 = reserved
	3	RSVD	R	0h	Reserved
	2:0	PLL1_REF_SEL	RW	0h	PLL1 Reference Mux, 000 = xtal_in/mclk1; 001 = mclk2; 010 = dac_bclk; 011 = adc_bclk; 100 = pll2 output; 101 - 111 = reserved

Table 116. PLLREFSEL Register

7.2.1.4. PLL1 Control Register

Register Address	Bit	Label	Type	Default	Description
R82(52h) PLLCTLD	7:5	RSVD	R	0	Reserved
	4:3	RZ_PLL1	RW	3h	PLL1 Zero R setting
	2:0	CP_PLL1	RW	2h	PLL1 main charge pump current setting

Table 117. PLLCTLD Register

7.2.1.5.PLL1 Reference Clock Divider Register

Register Address	Bit	Label	Type	Default	Description
R78(4Eh) PLLCTL9	7:0	REFDIV_PLL1	RW	19h	PLL1 refclk divider

Table 118. PLLCTL9 Register

7.2.1.6.PLL1 Output Divider Register

Register Address	Bit	Label	Type	Default	Description
R79(4Fh) PLLCTLA	7:0	OUTDIV_PLL1	RW	03h	PLL1 output divider

Table 119. PLLCTLA Register

7.2.1.7.PLL1 Feedback Divider Low Register

Register Address	Bit	Label	Type	Default	Description
R80(50h) PLLCTLB	7:0	FBDIVL_PLL1	RW	80h	PLL1 feedback divider

Table 120. PLLCTLB Register

7.2.1.8. PLL1 Feedback Divider High Register

Register Address	Bit	Label	Type	Default	Description
R81(51h) PLLCTLC	7:3	RSVD	R	0	Reserved
	2:0	FBDIVH_PLL1	RW	1h	PLL1 feedback divider

Table 121. PLLCTLC Register

7.2.1.9.PLL2 Control Register

Register Address	Bit	Label	Type	Default	Description
R87(57h) PLLCTL12					
	7:6		R	0	Reserved
	5:3	RZ_PLL2	RW	3h	PLL2 Zero R setting
	2:0	CP_PLL2	RW	2h	PLL2 main charge pump current setting

Table 122. PLLCTL12 Register

7.2.1.10.PLL2 Reference Clock Divider Register

Register Address	Bit	Label	Type	Default	Description
R83(53h) PLLCTLE	7:0	REFDIV_PLL2	RW	12h	PLL2 reference clock divider

Table 123. PLLCTLE Register

7.2.1.11.PLL2 Output Divider Register

Register Address	Bit	Label	Type	Default	Description
R84(54h) PLLCTLEF	7:0	OUTDIV_PLL2	RW	03h	PLL2 output divider

Table 124. PLLCTLF Register

7.2.1.12.PLL2 Feedback Divider Low Register

Register Address	Bit	Label	Type	Default	Description
R85(55h) PLLCTL10	7:0	FBDIVL_PLL2	RW	1ch	PLL2 feedback low divider

Table 125. PLLCTL10 Register

7.2.1.13.PLL2 Feedback Divider High Register

Register Address	Bit	Label	Type	Default	Description
R86(56h) PLLCTL11	7:3	RSVD	R	0	Reserved
	2:0	FBDIVH_PLL2	RW	2h	PLL2 feedback high divider

Table 126. PLLCTL11 Register

7.2.1.14.PLL Pwer DownControl Register

Register Address	Bit	Label	Type	Default	Description
R97(61h) PLLCTL1C	7:3	RESERVED	R	0h	Reserved
	2	PDB_PLL2	RW	0h	PLL2 Power Down: 1 = Power Up 0 = Power Down
	1	PDB_PLL1	RW	0h	PLL1Power down 1 = Power Up 0 = Power Down
	0	RESERVED	R	0h	Reserved

Table 127. PLLCTL1C Register

7.2.2 PLL Power Down Control

Each PLL can be powered down to save power if only one set of base audio rates is required. The base audio rates are defined as 44.1KHz based rates or 48KHz based rates. If support for either 44.1KHz or 48KHz based rates is not needed then the PLL associated with the unused rate can be powered down.

7.2.3 Audio Clock Generation

Figure 33 shows the simplified block diagram. The TSCS42xx utilizes internal PLLs to generate the PLL clocks at 112.896 MHz (22.5792MHz *5) and122.880 MHz (24.576 *5). Intermediate clocks (61.44MHz, 40.96MHz, 56.448MHz) are then generated which are then used to generate the audio sample rates. There is one internal clock rate that can be specified to operate at 11.025KHz, 12 KHz, 22.050KHz, 24KHz, 44.1KHz, 48KHz, 88.2KHz, and 96KHz. When changing sample rates, a delay of up to 5mS may be needed for the part to properly lock PLLs, flush filters, etc.

7.2.3.1.PLL Clock Source

The clock source for the PLL can be selected from the XTAL input, MCLK1 input via the XTAL_IN pin, the MCLK2 pin or one of the I2S BCLK inputs via a selectable mux.

7.2.3.2. Internal Sample Rate Control Register

These register define the internal sample rate.

Register Address	Bit	Label	Default	Description
R23(17h) ADCSR	7:6	ABCM	0h	ADC bit Clock Mode (for ADCBCLK generation in master mode): 0h=Auto 1h = 32x Fs 2h = 40x Fs 3h = 64x Fs
	5	RSVD	0h	Reserved
	4:3	ABR	2h	ADC Base Rate 0h = 32kHz 1h = 44.1kHz 2h = 48KHz 3h = Reserved
	2:0	ABM	2h	ADC Base Rate Multiplier 0h = 0.25x 1h = 0.5x 2h = 1x 3h = 2x 4h-7h = Reserved

Table 128. ADCSR Register

Register Address	Bit	Label	Default	Description
R25 (19h) DACSR	7:6	DBCM	0h	DAC bit Clock Mode (for DACBCLK generation in master mode): 0h=Auto 1h = 32x Fs 2h = 40x Fs 3h = 64x Fs
	5	RSVD	0h	Reserved
	4:3	DBR	2h	DAC Base Rate 0h = 32kHz 1h = 44.1kHz 2h = 48KHz 3h = Reserved
	2:0	DBM	2h	DAC Base Rate Multiplier 0h = 0.25x 1h = 0.5x 2h = 1x 3h = 2x 4h-7h = Reserved

Table 129. DACSR Register

Internal Sample Rates

xBR [4:3]	xBM [2:0]	BASE RATE	SAMPLE RATE
00	000	40.96MHz	8kHz(MCLK/5120)
	001		16kHz(MCLK/2560)
	010		32 kHz (MCLK/1280)
	011		64kHz (MCLK/640)
01	000	56.448MHz	11.025kHz(MCLK/5120)
	001		22.050kHz(MCLK/2560)
	010		44.1 kHz (MCLK/1280)
	011		88.2 kHz (MCLK/640)
10	000	61.44 MHz	12kHz(MCLK/5120)
	001		24kHz(MCLK/2560)
	010		48 kHz (MCLK/1280)
	011		96 kHz (MCLK/640)

Table 130. DAC/ADC Sample rates

7.2.3.3. MCLK2 Pin

The MCLK2 pin is configured to be an input and can provide a clock to drive the input to the PLLs or the I2S Master Mode clock generators.

7.2.3.4. I2S Master Mode Clock Generation

I2S input audio source can operate as a timing Slave or Master. When operated in Master Mode an internal clock generator is used to produce the required bit and frame clocks to be driven out of the LRCLK and BCLK pins of each input I2S interface. The clock source for the I2S master clock generation can be selected between the PLL generated internal timing or an externally supplied clock via the MCLK2 input.

7.2.3.5. I2S Master Mode Sample Rate Control

TI2S slave or master mode is set in register 13 MS bit. The I2S BR bits set the base audio sample to be either 44.1Khz or 48KHz. The I2S BM bits are then used to set the base rate multiplier ratio. The I2S BCM bits set the BCLK ratio vs sample rate. The I2S BR, BM and BCM bits are located in register 17h for the ADC while register 19h for the DAC.

7.2.3.6. DAC/ADC Clock Control

The power consumption and audio quality may be adjusted by changing the converter modulator rate. By default the DAC and ADC Sigma-Delta modulators run at a high rate for the best audio quality. The modulator rates for the converters may be forced to run at half their nominal rate to conserve power. A third option allows the modulator rate to automatically drop to half rate when low sampling rates are chosen (1/2 or 1/4 the base rate.) The DACs and ADCs are independently controlled

Register Address	Bit	Label	Type	Default	Description
R31(1Fh) CONFIG0	7:6	ASDM[1:0]	RW	2h	ADC Modulator Rate 00b = Reserved 01b = Half 10b = Full 11b = Auto
	5:4	DSDM[1:0]	RW	2h	DAC Modulator Rate 00b = Reserved 01b = Half 10b = Full 11b = Auto
	3:2	RSVD	R	0	Reserved
	1	DC_BYPASS	RW	0h	DAC DC Filter Bypass: 0 = Filter enable 1 = Filter bypassed
	0	SD_FORCE_ON	RW	0h	Supply Detect Force On: 0 = Supply detect not forced on 1 = Supply detect forced on Note If not forced on, the supply detect logic will automatically be enabled when features that use it are enabled (COP,UVLO)

Table 131. CONFIG0 Register

DSDM[1:0] ASDM[1:0]	BM [2:0]	Modulator Rate
00	NA	Reserved
01	000 (1/4x)	Half
	001 (1/2x)	
	010 (1x)	
	011 (2x)	
10	000 (1/4x)	Full
	001 (1/2x)	
	010 (1x)	
	011 (2x)	
11	000 (1/4x)	Auto (Half)
	001 (1/2x)	Auto (Half)
	010 (1x)	Auto (Full)
	011 (2x)	Auto (Full)

Table 132. ADC and DAC Modulator Rates

7.2.3.7. Timebase Register

Register Address	Bit	Label	Type	Default	Description
R119(77h) TMBASE	7-0	TIMEBASE[7:0]	RW	2F	Internal Time Base Divider. This value should be programmed as [round(ref clock/256000)]-1

Table 133. TIMEBASE Register

8. CHARACTERISTICS

8.1. Electrical Specifications

8.1.1. Absolute Maximum Ratings

Stresses above the ratings listed below can cause permanent damage to the TSCS42xx. These ratings, which are standard values for TSI commercially rated parts, are stress ratings only. Functional operation of the device at these or any other conditions above those indicated in the operational sec-tions of the specifications is not implied. Exposure to absolute maximum rating conditions for extended periods can affect product reliability. Electrical parameters are guaranteed only over the recommended operating temperature range.

Item	Maximum Rating
Voltage on any pin relative to Ground	Vss - 0.3V TO Vdd + 0.3V
Operating Temperature	0 ^{O}C TO 70 ^{O}C
Storage Temperature	-55 ^{O}C TO +125 ^{O}C
Soldering Temperature	260 ^{O}C
MICBias Output Current	3mA
Amplifier Maximum Supply Voltage	6 Volts = PVDD
Audio Maximum Supply Voltage	3 Volts = AVDD/CPVDD
Digital I/O Maximum Supply Voltage	3.6 Volts = DVDD_IO
Digital Core Maximum Supply Voltage	2.0 Volts = DVDD

Table 134. Electrical Specification: Maximum Ratings

8.1.2. Recommended Operating Conditions

Parameter		Min.	Typ.	Max.	Units
Power Supplies	DVDD_Core	1.4		2.0	V
	DVDD_IO	1.4		3.5	
	AVDD/CPVDD	1.7		2.0	
	PVDD	3.0		5.5	V
Ambient Operating Temperature	Analog - 5 V	0	25	70	^{O}C
Case Temperature	T_{case}			90	^{O}C

Table 135. Recommended Operating Conditions

ESD: The TSCS42xx is an ESD (electrostatic discharge) sensitive device. The human body and test equipment can accumulate and discharge electrostatic charges up to 4000 Volts without detection. Even though the TSCS42xx implements internal ESD protection circuitry, proper ESD precautions should be followed to avoid damaging the functionality or performance.

8.2. Device Characteristics

($T_{ambient}$ = 25 ºC, DVDD_CORE=DVDD_IO=AVDD=1.9V, PVDD=3.6V, 997Hz signal, fs=48KHz, Input Gain=0dB, 24-bit audio)

Parameter	Symbol	Test Conditions	Min	Typ	Max	Unit
Analog Inputs (L_{IN1}, L_{IN2}, L_{IN3}, R_{IN1}, R_{IN2}, R_{IN3})						
Full Scale Input Voltage	V_{FSIV}	$L/R_{IN1,2,3}$ Single Ended		0.5 -6		Vrms dBV
		$L/R_{IN1,2,3}$ Differential Mic		0.5 -6		Vrms dBV
Input Impedance				50		Kohm
Input Capacitance				10		pF
Analog Input Boost Amplifier						
Programmable Gain Min				0.0		dB
Programmable Gain Max				30.0		dB
Programmable Gain Step Size				10.0		dB
Analog Input PGA						
Programmable Gain Min				-17.25		dB
Programmable Gain Max				30.0		dB
Programmable Gain Step Size		Guaranteed Monotonic		0.75		dB
Digital Volume Control Amplifier						
Programmable Gain Min				-97		dB
Programmable Gain Max				30.0		dB
Programmable Gain Step Size		Guaranteed Monotonic		0.5		dB
Mute Attenuation				-999		dB
Analog Inputs (L_{IN1}/R_{IN1}, L_{IN2}/R_{IN2} Differential) to ADC						
Signal To Noise Ratio	SNR	A-weighted 20-20KHz		90		dB
Total Harmonic Distortion + Noise	THD+N	-1dBFS input		-80 0.01		dB %
Analog Inputs (L_{IN1}, L_{IN2}, L_{IN3}, R_{IN1}, R_{IN2}, R_{IN3} Single Ended) to ADC						
Signal To Noise Ratio	SNR	A-weighted 20-20KHz		90		dB
Total Harmonic Distortion + Noise	THD+N	-1dBFS input		-80 0.01		dB %
ADC channel Separation		997Hz full scale signal		70		dB
Channel Matching		997Hz signal			2	%
DAC to Line-Out (HPL, HPR with 10K / 50pF load)						
Signal to Noise Ratio[1]	SNR	A-weighted		102		dB
Total Harmonic Distortion +Noise[2]	THD+N	997Hz full scale signal		-84		dB
Channel Separation		997Hz full scale signal		70		dB
Mute attenuation				-999		dB
Headphone Outputs (HPL, HPR)						
Full Scale Output Level	V_{FSOV}	RL = 10Kohm		1.0		Vrms
		R_L = 16ohm		0.75		Vrms
Output Power	P_O	997Hz full scale signal, R_L = 16ohm		35		mW (ave)
Signal to Noise Ratio	SNR	A-weighted, R_L = 16ohm		102		dB

Table 136. Device Characteristics

Parameter	Symbol	Test Conditions	Min	Typ	Max	Unit
Total Harmonic Distortion +Noise	THD+N	R_L = 16ohms, -3dBFS		-76		dB
		R_L = 32ohms, -3dBFS		-78		dB
Speaker Outputs (L+, L-, R+, R- with 8ohms bridge-tied load)						
Full Scale Output Level	V_{FSOV}	PVDD=5V PVDD=3.6V		3.0 2.1		Vrms
Output Power	P_O	997Hz full scale signal, output power mode disabled PVDD=5V, 8ohm PVDD=3.6V, 8ohm		1.5 7		W(ave)
		PVDD = 5V, 4 ohm DIDD = 3.6V, 4 ohm		3 1.4		W(ave)
Signal to Noise Ratio	SNR	A-weighted		90		dB
Total Harmonic Distortion + Noise	THD+N	5V/8ohms/0.5W		0.05		%
Speaker Supply Leakage Current	I_{PVDD}			1		uA
Efficiency	h	PVDD=3.6V RL=8,P_O = 0.5W PVDD=5V RL=8,P_O = 1W		92		%
Analog Voltage Reference Levels						
Charge Pump Output	V-		-5%	-AVDD +100mV	+5%	V
Microphone Bias						
Bias Voltage	$V_{MICBIAS}$		-	2.5	-	V
BIAS current Source					3	mA
Power Supply Rejection Ratio	$PSRR_{MICBIAS}$	3.3V<PVDD<5.25V		80		dB
		3.0V<PVDD<3.3V		40		dB
Digital Input/Output						
ADC/DAC BCLK input rate	Fmax			30		MHz
I2S BCLK/LRCLK ratio			32		1022	clocks/ frame
Input High Level	V_{IH}		0.7x DVDD_ IO			V
Input LOW Level	V_{IL}				0.3x DVDD_IO	V
Output High Level	V_{OH}	I_{OH}=-1mA	0.9x DVDD_IO			V
Output LOW Level	V_{OL}	I_{OL}=1mA		0.1xDVDD_IO		V
Input Capacitance				5		pF
Input Leakage			-0.9		0.9	uA
Internal Pull-Up Resistor	R_{PU} / R_{PU}	All Digital I/O pins with pull-up or pull-down		50		kΩ
ESD / Latchup						
IEC1000-4-2			1			Level
JESD22-A114-B			2			Class
JESD22-C101			4			Class

Table 136. Device Characteristics

1.Ratio of Full Scale signal to idle channel noise output is measured “A weighted” over a 20 Hz to a 20 kHz bandwidth. (AES17-1991 Idle Channel Noise or EIAJ CP-307 Signal-to-noise Ratio).

2.THD+N ratio as defined in AES17 and outlined in AES6id, non-weighted, swept over 20 Hz to 20 kHz bandwidth.

8.3. Electrical Characteristics

Unless stated otherwise, DVDD_Core=1.8V -0.1V/+0.2V, Ambient Temp -10C to +80C

Parameter	Symbol	Test Conditions	Min	Typ	Max	Unit
Operating Voltage	DVDD_CORE		1.7	1.8	2.0	V
Supply Current	$I_{DVDD\_CORE(PLL)}$	No Load, VDD=1.9V		11	15	mA
Input High Level	V_{IH}		0.7x DVDD_CORE			V
Input LOW Level	V_{IL}				0.3xDVDD_CORE	V
Input Capacitance	C_{IN}			5		pF
Load Capacitance, X1 and X2	C_L			5		pF
Output High Level	V_{OH}	I_{OH}=-2mA	0.8x DVDD_CORE			V
Output LOW Level	V_{OL}	I_{OL}=2mA			0.2xDVDD_CORE	V
Power Up Time	t_{PU}	From minimum VDD to outputs stable		1.5	4	ms
Output Enable Time					20	ns
Output Disable Time					20	ns

Table 137. PLL Section DC Characteristics

8.3.1. Low Power Mode Consumption

Mode	AVDD (V)	PVDD (V)	DVDD_CORE (V) DVDD_IO	I AVDD (mA)	IPVDD (mA)	IDVDD_IO IDVDD_CORE (mA)	PTOTAL (mW)
Out of Reset	1.7	5	1.7	0.0064	0	0.18	0.31688
HP Full Power 10k Ω Note 1	1.7	5	1.7	8.7	0	9.9	31.62
HP Low Power 10k Ω Note 2	1.7	5	1.7	3.47	0	4.64	13.787
HP Low Power silence Note 1	1.7	5	1.7	1.87	0	3.45	9.044
Line In Full Power Note 2	1.7	5	1.7	7.27	0	7.45	25.024
Line In Low Power Note 2	1.7	5	1.7	3	0	2.72	9.724
Line In Low Power Silence Note 2	1.7	5	1.7	2.9	0	0.49	5.763
Note 1 - DAC is 48kHz with BCLK at 1.536MHz with -3dB signal input							
Note 2 - ADC is 48kHz with BCLK at 1.536MHz with -9dB signal input							

Table 138. Low Power Mode Consumption

9. REGISTER MAP

Table 139. Register Map

Register (D15:9)	Name	Remarks	Bit[7]	Bit[6]	Bit[5]	Bit[4]	Bit[3]	Bit[2]	Bit[1]	Bit[0]	Default
R0 (00h)	HPVOLL	Left HP volume	HPVOL_L[6:0]								79h
R1 (01h)	HPVOLR	Right HP volume	HPVOL_R[6:0]								79h
R2 (02h)	SPKVOLL	SPKR Left volume	SPKVOL_L[6:0]								6Fh
R3 (03h)	SPKVOLR	SPKR Right volume	SPKVOL_R[6:0]								6Fh
R4 (04h)	DACVOLL	Left DAC volume	DACVOL_L[7:0]								FFh
R5 (05h)	DACVOLR	Right DAC volume	DACVOL_R[7:0]								FFh
R6 (06h)	ADCVOLL	Left ADC volume	ADCVOL_L[7:0]								BFh
R7 (07h)	ADCVOLR	Right ADC volume	ADCVOL_R[7:0]								BFh
R8 (08h)	INVOLL	Left Input volume	INMUTEL	IZCL	INVOL_L						17h
R9 (09h)	INVOLR	Right Input volume		IZCR	INVOL_R						17h
R10 (0Ah)	VUCTL	Volume Update Control	ADCFade	DACFade		INVOLU	ADCVOLU	DACVOLU	SPKVOLU	HPVOLU	C0h
R11 (0Bh)	INMODE	ADC input mode								DS	00h
R12 (0Ch)	INSELL	ADCL signal path	INSEL_L[1:0]		MICBST_L[1:0]						00h
R13 (0Dh)	INSELR	ADCR signal path	INSEL_R[1:0]		MICBST_R[1:0]						00h
R14 (0Eh)	ALC0	ALC0						ALC MODE	ALCSEL[1:0]		00h
R15 (0Fh)	ALC1	ALC1		MAXGAIN[2:0]			ALCL[3:0]				7Bh
R16 (10h)	ALC2	ALC2		MINGAIN[2:0]			HLD[3:0]				00h
R17 (11h)	ALC3	ALC3	DCY[3:0]				ATK[3:0]				32h
R18 (12h)	NGATE	Noise Gate	NGTH[4:0]					NGG[1:0]		NGAT	00h
R19 (13h)	AIC1	Audio Interface 1		BCLKINV	MS	LRP	WL[1:0]		FORMAT[1:0]		0Ah
R20 (14h)	AIC2	Audio Interface 2	DACDSEL[1:0]		ADCDSEL[1:0]		TRI	BLRCM[2:0]			00h
R21 (15h)	AIC3	Audio Interface 3			ADOPDD	ALRPDD	ABCPDD	DDIPDD	DLRPDD	DBCPDD	00h
R22 (16h)	CNVRTR0	ADC Control	ADCPOLR	ADCPOLL	AMONOMIX[1:0]		ADCMU	HPOR	ADCHPDR	ADCHPDL	08h
R23 (17h)	ADCSR	ADC Sample rate	ABCM[1:0]			ABR[1:0]		ABM[2:0]			12h
R24 (18h)	CNVRTR1	DAC Control	DACPOLR	DACPOLL	DMONOMIX[1:0]		DACMU	DEEMP	DACDITH1	DACDITH0	08h
R25 (19h)	DACSR	DAC Sample rate	DBCM[1:0]			DBR[1:0]		DBM[2:0]			12h
R26 (1Ah)	PWRM1	Pwr Mgmt (1)	BSTL	BSTR	PGAL	PGAR	ADCL	ADCR	MICB	DIGENB	00h
R27 (1Bh)	PWRM2	Pwr Mgmt (2)	D2S	HPL	HPR	SPKL	SPKR			VREF	00h
R28 (1Ch)	CTL	Additional control	HPSWEN	HPSWPOL	EQ2SW1	EQ2SW0	EQ1SW1	EQ1SW0	TSDEN	TOEN	00h
R29 (1Dh)	THERMTS	Temp Sensor Control	TripHighStat	TripLowStat	TripSplit[1:0]		TripShift[1:0]		Poll[1:0]		09h
R30 (1Eh)	THERMSPKR1	Speaker Thermal Algorithm Control	ForcePwd	InstCutMode	IncRatio[1:0]		IncStep[1:0]		DecStep[1:0]		81h
R31 (1Fh)	CONFIG0	CONFIG0	ASDM1	ASDM0	DSDM1	DSDM0			dc_bypass	sd_force_on	A0h
R32 (20h)	CONFIG1	CONFIG1	EQ2_EN	EQ2_BE2	EQ2_BE1	EQ2_BE0	EQ1_EN	EQ1_BE2	EQ1_BE1	EQ1_BE0	00h
R33 (21h)	GAINCTL	Gain Control	zerodet_flag		zerodetlen 1	zerodetlen 0	auto_pwr	auto_mute			24h
R34 (22h)	COP1	Constant Output Power1	COPAtten	COPGain	HDeltaEn	COPTarget[4:0]					08h
R35 (23h)	COP2	Constant Output Power2			AvgLength[2:0]				MonRate[2:0]		04h
R36 (24h)	DMICCTL	D-Mic Control	DMicEn			DMono	DMPhAdj1	DMPhAdj0	DMRate1	DMRate0	00h
R37 (25h)	CLECTL	CMPLMTCTL				Lvl_Mode	WindowSel	Exp_En	Limit_En	Comp_En	00h
R38 (26h)	MUGAIN	CLEMakeUpGain				CLEMUG4	CLEMUG3	CLEMUG2	CLEMUG1	CLEMUG0	00h
R39 (27h)	COMPTH	Compressor Threshold	COMPTH7	COMPTH6	COMPTH5	COMPTH4	COMPTH3	COMPTH2	COMPTH1	COMPTH0	00h
R40 (28h)	CMPRAT	Compressor Ratio				CMPRAT4	CMPRAT3	CMPRAT2	CMPRAT1	CMPRAT0	00h
R41 (29h)	CATKTCL	Comp Attack time const Low	CATKTC7	CATKTC6	CATKTC5	CATKTC4	CATKTC3	CATKTC2	CATKTC1	CATKTC0	00h
R42(2Ah)	CATKTCH	Comp Attack time const High	CATKTC15	CATKTC14	CATKTC13	CATKTC12	CATKTC11	CATKTC10	CATKTC9	CATKTC8	00h

Register (D15:9)	Name	Remarks	Bit[7]	Bit[6]	Bit[5]	Bit[4]	Bit[3]	Bit[2]	Bit[1]	Bit[0]	Default
R43 (2Bh)	CRELTCL	Comp release time const Low	CRELTC7	CRELTC6	CRELTC5	CRELTC4	CRELTC3	CRELTC2	CRELTC1	CRELTC0	00h
R44 (2Ch)	CRELTCH	Comp release time const High	CRELTC15	CRELTC14	CRELTC13	CRELTC12	CRELTC11	CRELTC10	CRELTC9	CRELTC8	00h
R45 (2Dh)	LIMTH	Limiter Threshold	LIMTH7	LIMTH6	LIMTH5	LIMTH4	LIMTH3	LIMTH2	LIMTH1	LIMTH0	00h
R46 (2Eh)	LIMTGT	Limiter Target	LIMTGT7	LIMTG6	LIMTGT5	LIMTGT4	LIMTGT3	LIMTGT2	LIMTGT1	LIMTGT0	00h
R47 (2Fh)	LATKTCL	Limiter Attack time constant Low	LATKTC7	LATKTC6	LATKTC5	LATKTC4	LATKTC3	LATKTC2	LATKTC1	LATKTC0	00h
R48 (30h)	LATKTCH	Limiter Attack time constant High	LATKTC15	LATKTC14	LATKTC13	LATKTC12	LATKTC11	LATKTC10	LATKTC9	LATKTC8	00h
R49 (31h)	LRELTCL	Limiter Release time constant Low	LRELTC7	LRELTC6	LRELTC5	LRELTC4	LRELTC3	LRELTC2	LRELTC1	LRELTC0	00h
R50 (32h)	LRELTCH	Limiter Release time constant High	LRELTC15	LRELTC14	LRELTC13	LRELTC12	LRELTC11	LRELTC10	LRELTC9	LRELTC8	00h
R51 (33h)	EXPTH	Expander Threshold	EXPTH7	EXPTH6	EXPTH5	EXPTH4	EXPTH3	EXPTH2	EXPTH1	EXPTH0	00h
R52 (34h)	EXPRAT	Expander Ratio						EXPRAT2	EXPRAT1	EXPRAT0	00h
R53 (35h)	XATKTCL	Expander Attack time constant Low	XATKTC7	XATKTC6	XATKTC5	XATKTC4	XATKTC3	XATKTC2	XATKTC1	XATKTC0	00h
R54 (36h)	XATKTCH	Expander Attack time constant High	XATKTC15	XATKTC14	XATKTC13	XATKTC12	XATKTC11	XATKTC10	XATKTC9	XATKTC8	00h
R55 (37h)	XRELTCL	Expander Release time constant Low	XRELTC7	XRELTC6	XRELTC5	XRELTC4	XRELTC3	XRELTC2	XRELTC1	XRELTC0	00h
R56 (38h)	XRELTCH	Expander Release time constant High	XRELTC15	XRELTC14	XRELTC13	XRELTC12	XRELTC11	XRELTC10	XRELTC9	XRELTC8	00h
R57 (39h)	FXCTL	Effects Control				3DEN	TEEN	TNLFBYP	BEEN	BNLFBYP	00h
R58 (3Ah)	DACCRWRL	DACCRAM_WRITE_LO	DACCRWD[7:0]								00h
R59 (3Bh)	DACCRWRM	DACCRAM_WRITE_MID	DACCRWD[15:8]								00h
R60 (3Ch)	DACCRWRH	DACCRAM_WRITE_HI	DACCRWD[23:16]								00h
R61 (3Dh)	DACCRRDL	DACCRAM_READ_LO	DACCRRD[7:0]								00h
R62 (3Eh)	DACCRRDM	DACCRRAM_READ_MID	DACCRRD[15:8]								00h
R63 (3Fh)	DACCRRDH	DACCRRAM_READ_HI	DACCRRD[23:16]								00h
R64 (40h)	DACCRADDR	DACCRAM_ADDR	DACCRADD[7:0]								01h
R65 (41h)	DCOFSEL	DC_COEF_SEL						dc_coef_sel[2:0]			05h
R66 (42h)	PWM0	PWM Control 0	SCTO1	SCTO0	UVLO		bfclr	PWMMODE	l	NOOFFSET	C4h
R67 (43h)	PWM1	PWM Control 1	-	dithpos4	dithpos3	dithpos2	dithpos1	dithpos0	dith_range	dithclr	12h
R68 (44h)	PWM2	PWM Control 2									00h
R69 (45h)	PWM3	PWM Control 3	outctrl1	outctrl0				cvalue2	cvalue1	cvalue0	03h
R78 (4Eh)	PLLCTL9	PLL Control 9	refdiv_pll1[7:0]								05h
R79 (4Fh)	PLLCTLA	PLL Control 10	outdiv_pll1[7:0]								03h
R80 (50h)	PLLCTLB	PLL Control 11	fbdivl_pll1[7:0]								72h
R81 (51h)	PLLCTLC	PLL Control 12						fbdivH_pll1[10:8]			03h
R82 (52h)	PLLCTLD	PLL Control 13				rz_pll1[1:0]		cp_pll1[2:0]			22h
R83 (53h)	PLLCTLE	PLL Control 14	refdiv_pll2[7:0]								12h
R84 (54h)	PLLCTLF	PLL Control 15	outdiv_pll2[7:0]								03h
R85 (55h)	PLLCTL10	PLL Control 16	fbdivl_pll2[7:0]								1Ch
R86 (56h)	PLLCTL11	PLL Control 17						fbdivh_pll2[10:8]			02h
R87 (57h)	PLLCTL12	PLL Control 18				rz_pll2[1:0]		cp_pll2[2:0]			1Ah
R96 (60h)	PLLCTL1B	PLL Control 27			vcoi_pll2		vcoi_pll1				14h
R97 (61h)	PLLCTL1C	PLL Control 28						pdb_pll2	pdb_pll1		0Fh
R119(77h)	TIMEBASE	Divider	TIMEBASE[7:0]								61h
R124(7Ch)	DEVADR	I2C Device Address	ADDR[7:1]								D2h
R125(7Dh)	DEVIDL	Device IDLow	DID7	DID6	DID5	DID4	DID3	DID2	DID1	DID0	01h
R126(7Eh)	DEVIDH	Device ID High	DID15	DID14	DID13	DID12	DID11	DID10	DID9	DID8	00h
R127(7Fh)	REVID	Device Revision	MAJ3	MAJ2	MAJ1	MAJ0	MNR3	MNR2	MNR1	MNR0	11h
R128(80h)	RESET	Reset	Writing 0x85 to this register resets all registers to their default state								00h
R136(88h)	THERMSPKR2	Speaker Thermal Algorithm Status	ForcePwd Status	VolStatus[6:0]							08h

Register (D15:9)	Name	Remarks	Bit[7]	Bit[6]	Bit[5]	Bit[4]	Bit[3]	Bit[2]	Bit[1]	Bit[0]	Default
R137(89h)	COP3	Constant Output Power Status	HighDelta	UNDER VOLTAGE	COPAdj[5:0]						00h
R138(8Ah)	DACCRSTAT	DACCRAM_STATUS	DACCR_Busy								00h
R139(8Bh)	HPDETSTAT	HP detect pin status	HP_Detect								00h
R142(8Eh)	PLLCTL0	PLL Control 0							PLL2LK	PLL1LKk	00h
R143(8Fh)	PLLREFSEL	PLL Control		PLL2_REF				PLL1_REF			00h
R192(C0h)	GPIOCTL1	GPIO CONTROL			GPIO1CFG	GPIO0CFG			GPIO1DIR	GPIO0DIR	00h
R193(C1h)	GPIOCTL2	GPIO CONTROL							GPIO1PU	GPIO0PU	00h
R195(C3h)	ADCPCMCTL1	ADC PCM CONTROL1	GAINCODE			GAINENAB	BDELAYO	PCMFL	SLSYNC		00h
R196(C4h)	ADCPCMCTL2	ADC PCM CONTROL2		PCMMOMP	PCMSOP	PCMDSSP					00h
R197(C5h)	DACPCMCTL1	DAC PCM CONTROL1					BDELAYI	PCMFL	SLSYNC		00h
R198(C6h)	DACPCMCTL2	DAC PCM CONTROL2	PCMFORMAT	PCMMIM	PCMSI	PCMDSS		PCMSIGNEXT	PCM13MODE		00h
R199(C7h)	DACMBCEN	Multi-Band SELECTOR						MBCEN3	MBCEN2	MBCEN1	00h
R200(C8h)	DACMBCCTL	Multi-Band SELECTOR			LVLMODE3	WINSEL3	LVLMODE2	WINSEL2	LVLMODE1	WINSEL1	00h
R201(C9h)	DACMBCMUG1	Multi-Band SELECTOR			PHASE	MUGAIN					00h
R202(CAh)	DACMBCTHR1	COMPRESSOR	THRESH								00h
R203(CBh)	DACMBCRAT1	COMPRESSOR				RATIO					00h
R204(CCh)	DACMBCATK1L	COMPRESSOR	TCATKL								00h
R205(CDh)	DACMBCATK1H	COMPRESSOR	TCATKH								00h
R206(CEh)	DACMBCREL1L	COMPRESSOR	TCRELL								00h
R207(CFh)	DACMBCREL1H	COMPRESSOR	TCRELH								00h
R208(D0h)	DACMBCMUG2	Multi-Band 2SELECTOR			PHASE	MUGAIN					00h
R209(D1h)	DACMBCTHR2	COMPRESSOR 2	THRESH								00h
R210(D2h)	DACMBCRAT2	COMPRESSOR 2				RATIO					00h
R211(D3h)	DACMBCATK2L	COMPRESSOR 2	TCATKL								00h
R212(D4h)	DACMBCATK2H	COMPRESSOR 2	TCATKH								00h
R213(D5h)	DACMBCREL2L	COMPRESSOR 2	TCRELL								00h
R214(D6h)	DACMBCREL2H	COMPRESSOR 2	TCRELH								00h
R215(D7h)	DACMBCMUG3	Multi-Band 3 SELECTOR			PHASE	MUGAIN					00h
R216(D8h)	DACMBCTHR3	COMPRESSOR 3	THRESH								00h
R217(D9h)	DACMBCRAT3	COMPRESSOR 3				RATIO					00h
R218(DAh)	DACMBCATK3L	COMPRESSOR 3	TCATKL								00h
R219(DBh)	DACMBCATK3H	COMPRESSOR 3	TCATKH								00h
R220(DCh)	DACMBCREL3L	COMPRESSOR 3	TCRELL								00h
R221(DDh)	DACMBCREL3H	COMPRESSOR 3	TCRELH								00h

Note:

- Registers not described in this map should be considered “reserved”.
- Numerous portions of the register map are compatible with popular codecs from other vendors.

10. PIN INFORMATION

10.1. TSCS42A1 Pin Diagram

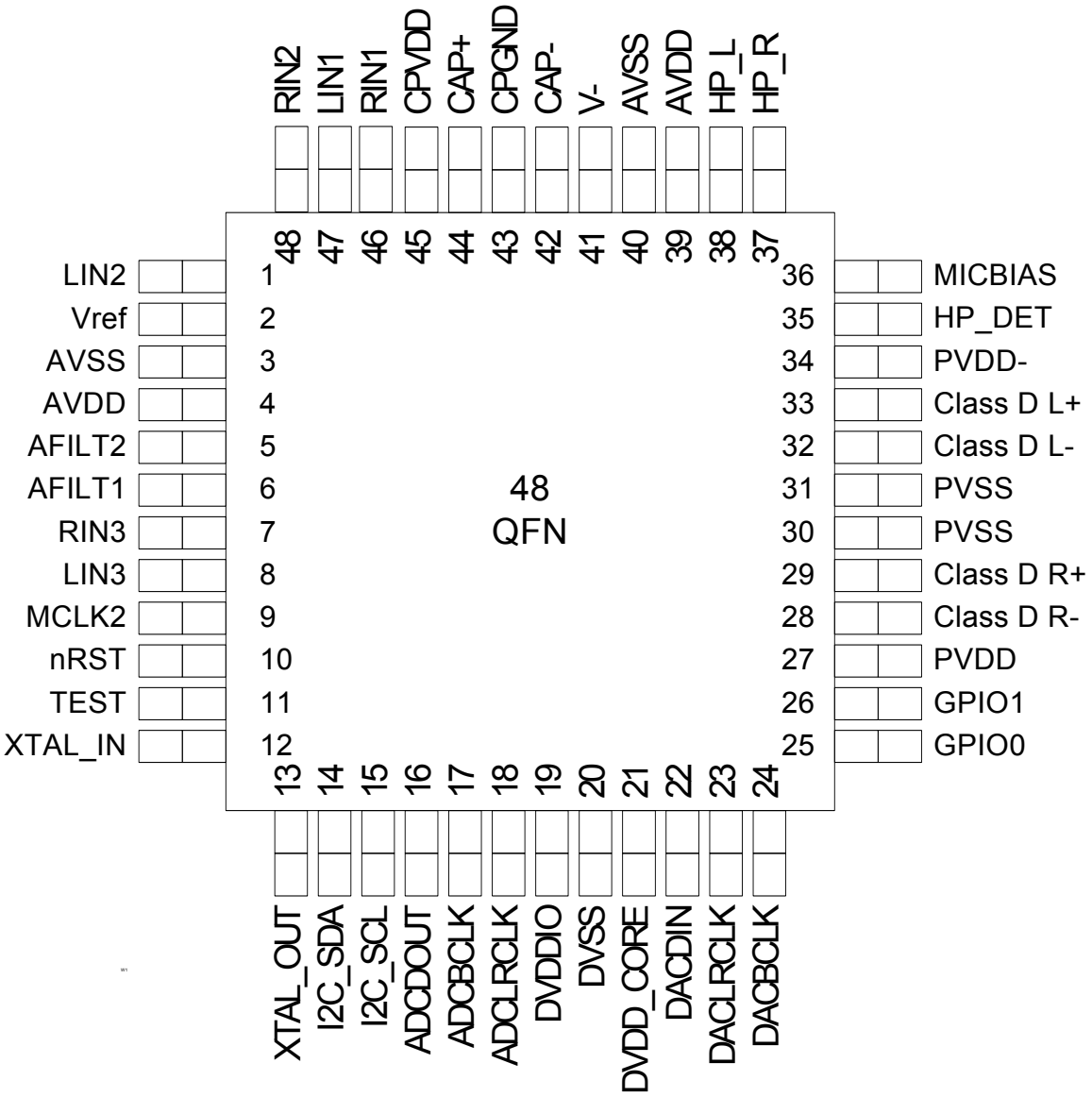

Figure 36. TSCS42A1 48QFN Pin Assignment

10.2. TSCS42A2 Pin Diagram

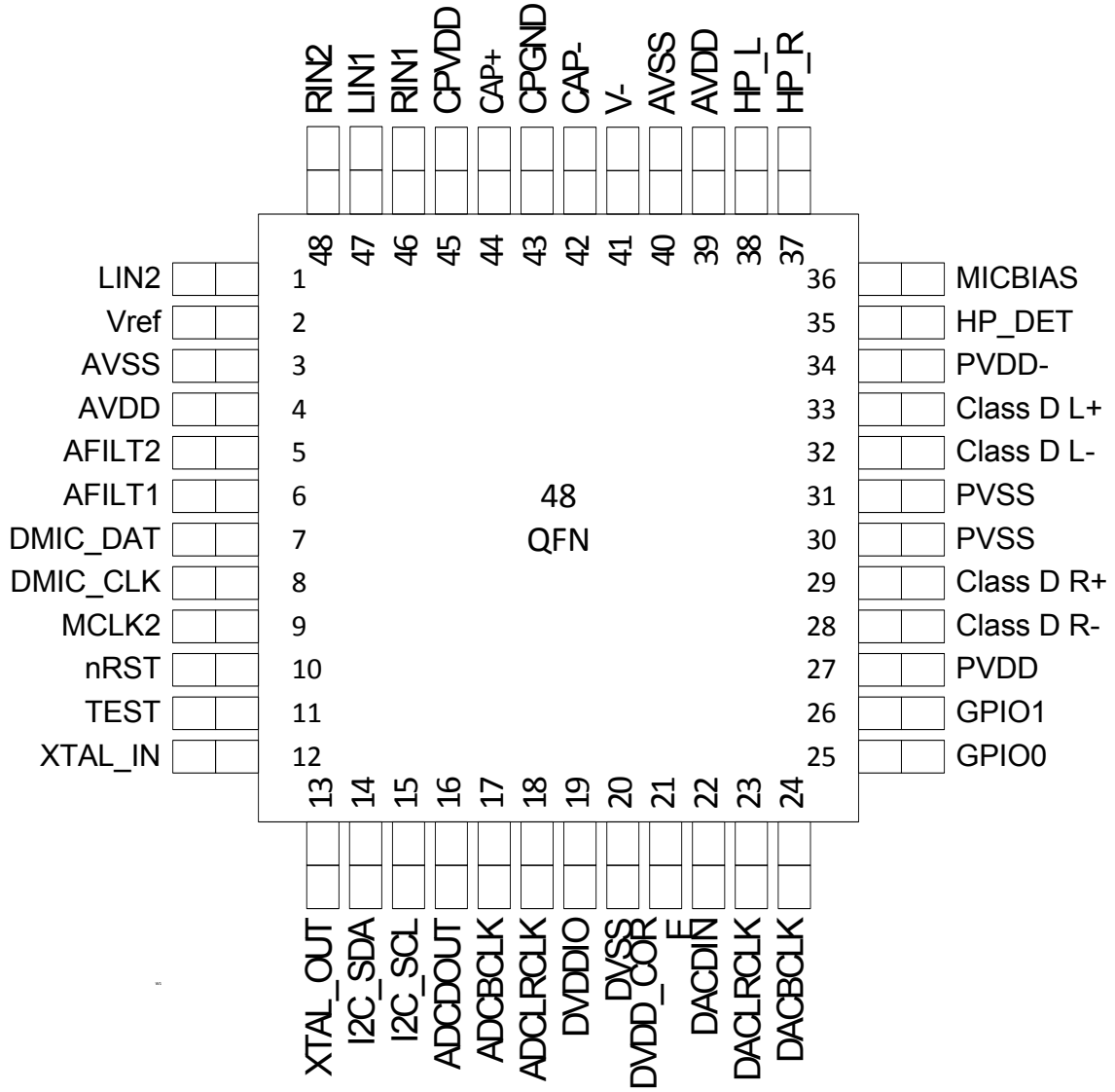

Figure 37. TSCS42A2 48QFN Pin Assignment

10.3. Pin Tables

10.3.1. Power Pins

Pin Name	Pin Function	I/O	Internal Pull-up Pull-down	Pin Location
PVDD	BTL supply	I(Power)	None	27,34
PVSS	BTL supply	I(Power)	None	30, 31
DVDD_Core	DSP and other core logic+clocks	I(Power)	None	21
DVDDIO	Interface (I^2S, I^2C, GPIO)	I(Power)	None	19
DVSS	Digital return	I(Power)	None	20
AVDD	Analog core supply	I(Power)	None	4,39
AVSS	Analog return	I(Power)	None	3, 40
CPVDD	Charge pump supply	I(Power)	None	45
CAP+	Flying cap	I/O(Power)	None	44
CAP-	Flying cap	I/O(Power)	None	42
V-	Negative Analog supply (Bypass cap)	O(Power)	None	41
CPGND	Charge pump group	I(Power)	None	43

Table 140. Power Pins

Total Pins: 16

10.3.2. Reference Pins

Pin Name	Pin Function	I/O	Internal Pull-up Pull-down	Pin Location
MICBIAS	2.5V 1.5 mA microphone bias	O(Analog)	None	36
AFILT1	ADC input filter cap	I(Analog)	None	6
AFILT2	ADC input filter cap	I(Analog)	None	5
Vref	VREF reference pin (bypass)	I(Analog)	None	2

Table 141. Reference Pins

Total Pins: 4

10.3.3. Analog Input Pins

Pin Name	Pin Function	I/O	Internal Pull-up Pull-down	Pin Locations
LIN1	Left Input #1	I(Analog)	None	47
RIN1	Right Input #1	I(Analog)	None	46
LIN2	Left Input #2	I(Analog)	None	1
RIN2	Right Input #2	I(Analog)	None	48
LIN3 DMIC_CLK	Left Input #3 for TSCS42A1 Digital Mic Clock forTSCS42A2	I(Analog)	None	8
RIN3 DMIC_DAT	Right Input #3 for TSCS42A1 Digital Mic Data for TSCS42A2	I(Analog)	None	7

Table 142. Analog Input Pins

Total Pins: 6

10.3.4. Analog Output Pins

Pin Name	Pin Function	I/O	Internal Pull-up Pull-down	Pin Locations
HP_L	Headphone output	O(Analog)	None	38
HP_R	Headphone output	O(Analog)	None	37
Class D L+	BTL Left positive output	O(Analog)	None	33
Class D L-	BTL Left negative output	O(Analog)	None	32
Class D R+	BTL Right positive output	O(Analog)	None	29
Class D R-	BTL Right negative output	O(Analog)	None	28

Table 143. Analog Output Pins

Total Pins: 6

10.3.5. Data and Control Pins

Pin Name	Pin Function	I/O	Internal Pull-up Pull-down	Pin Locations
GPIO1	General purpose I/O	I/O	None	26
GPIO0	General purpose I/O	I/O	None	25
nRST	Reset	I(Digital)	None	10
ADCBCLK	ADC I^2S shift clock	I/O(Digital)	Pull-Down	17
ADCLRCLK	ADC I^2S framing clock	I/O(Digital)	Pull-Down	18
ADCDOUT	ADC I^2S output data	O(Digital)	Pull-Down	16
DACBCLK	DAC I^2S shift clock	I/O(Digital)	Pull-Down	24
DACLRCLK	DAC I^2S framing clock	I/O(Digital)	Pull-Down	23
DACDIN	DAC I^2S input data	I(Digital)	Pull-Down	22
I2C_SCL	SCL I^2C shift clock	I(Digital)	Pull-Up	15
I2C_SDA	SDA I^2C shift data	I(Digital)	Pull-Up	14
HP_DET	Headphone jack detection	I(Digital)	Pull-Up	35
TEST	Reserved test pin	I(Analog)	None	11

Table 144. Data and Control Pins

Total Pins: 13

10.3.6. PLL Pins

Pin Name	Pin Function	I/O	Internal Pull-up Pull-down	48 Pin loc
XTAL_IN/MCLK1	Crystal input/MASTER CLK	I(XTAL)	None	12
XTAL_OUT	Crystal output	O(XTAL)	None	13
MCLK2	MASTER CLK	I(CLK)	None	9

Table 145. PLL Pins

Total Pins: 3

11. PACKAGE DRAWINGS

11.1. 48QFN Package Outline and Package Dimensions

Package dimensions are kept current with JEDEC Publication No. 95

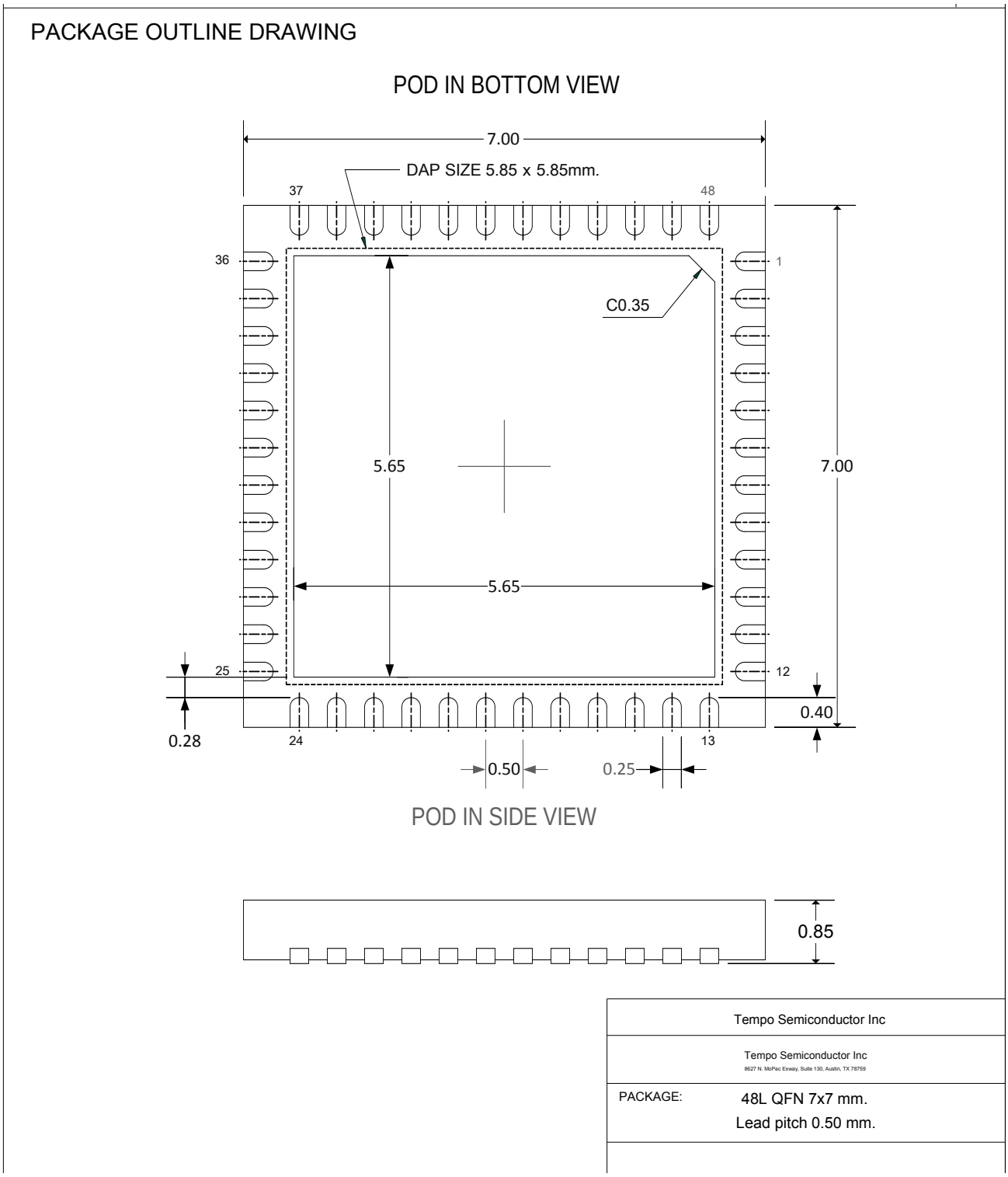

Figure 38. 48-pin QFN Package Diagram

11.2. Pb Free Process- Package Classification Reflow Temperatures

Package Thickness	Volume mm^3 <350	Volume mm^3 350 - 2000	Volume mm^3 >2000
<1.6mm	260 + 0 ^oC*	260 + 0 ^oC*	260 + 0 ^oC*
1.6mm - 2.5mm	260 + 0 ^oC*	250 + 0 ^oC*	245 + 0 ^oC*
> or = 2.5mm	250 + 0 ^oC*	245 + 0 ^oC*	245 + 0 ^oC*
*Tolerance: The device manufacturer/supplier shall assure process compatibility up to and including the stated classification temperature (this means Peak reflow temperature +0 ^oC. For example 260 ^oC+0 ^oC) at the rated MSL level.			

Table 146. Reflow Temperatures

Note: TSI's package thicknesses are <2.5mm and <350 mm^3, so 260 applies in every case.

12. APPLICATION INFORMATION

For application information, please see reference designs and application notes available on www.temposemi.com.

13. ORDERING INFORMATION

TSCS42A1X1NLGXZAX	Analog Microphone, 48 QFN package, Commercial Temp (0°C - 70°C)
TSCS42A2X1NLGXZAX	Digital Microphone, 48 QFN package - Commercial Temp (0°C - 70°C)
TSCS42A1X1NLGIZAX	Analog Microphone, 48 QFN package - Industrial Temp (-40°C - 85°F)
TSCS42A2X1NLGIZAX	Digital Microphone, 48 QFN package - Industrial Temp (-40°C - 85°F)

Please contact an TSI Sales Representative with your clock requirements for factory programming. This programming will determine the order able part number for the TSCS42xx.

14. DISCLAIMER

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15. DOCUMENT REVISION HISTORY

Revision	Date	Description of Change
0.5	May 2015	Initial release
0.8	July 2015	Updated Register set, I2S Section and Block diagram.
0.9	September 2015	Updated Register
0.95	January 2016	Updated Register and PLL information
1.0	January 2017	Pin diagram updated
1.1	February 2017	Removed sample order part number
1.2	April 2017	Corrected Treble and Bass Diagram and Output Power corrections
1.3	April 2017	Format change
1.4	December 2017	Removed Mux from Block diagram
1.5	March 2018	Updated year to 2018 and name error.
1.6	June 2020	Updated document revision, release date, and copy write year on all pages. Updated text spacing and size in Figure 34. Added industrial part numbers to updated part number table. Updated Figures 36 and 37 Titles to reflect part numbers TSCS42A1 and TSCS42A2. Updated Table 143: Class D L + was changed to pin 33 and Class DL- was changed to pin 32 in order to match the 48 QFN package pin out. Other changes include making font type consistent in note areas.
1.7	July 2020	Updated Tables 140 - 145: Removed all mentions of Pin Location associated with a 40-pin Package as this device is only offered in a 48-pin Package.

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