CS42L73

Ultralow Power Mobile Audio and Telephony CODEC

Product Overview

- Stereo analog-to-digital converter (ADC)
- Dual analog or digital mic support
- Dual mic bias generators
- Four digital-to-analog converters (DACs) coupled to five outputs
  - Ground-centered stereo headphone amp.
  - Ground-centered stereo line output
  - Mono ear speaker amplifier
  - Mono 1-W speakerphone amplifier
  - Mono speakerphone line output for stereo speakerphone expansion
- Three serial ports with asynchronous sample rate converters
- Digital audio mixing and routing

Ultralow Power Consumption

- 3.8-mW quiescent headphone playback

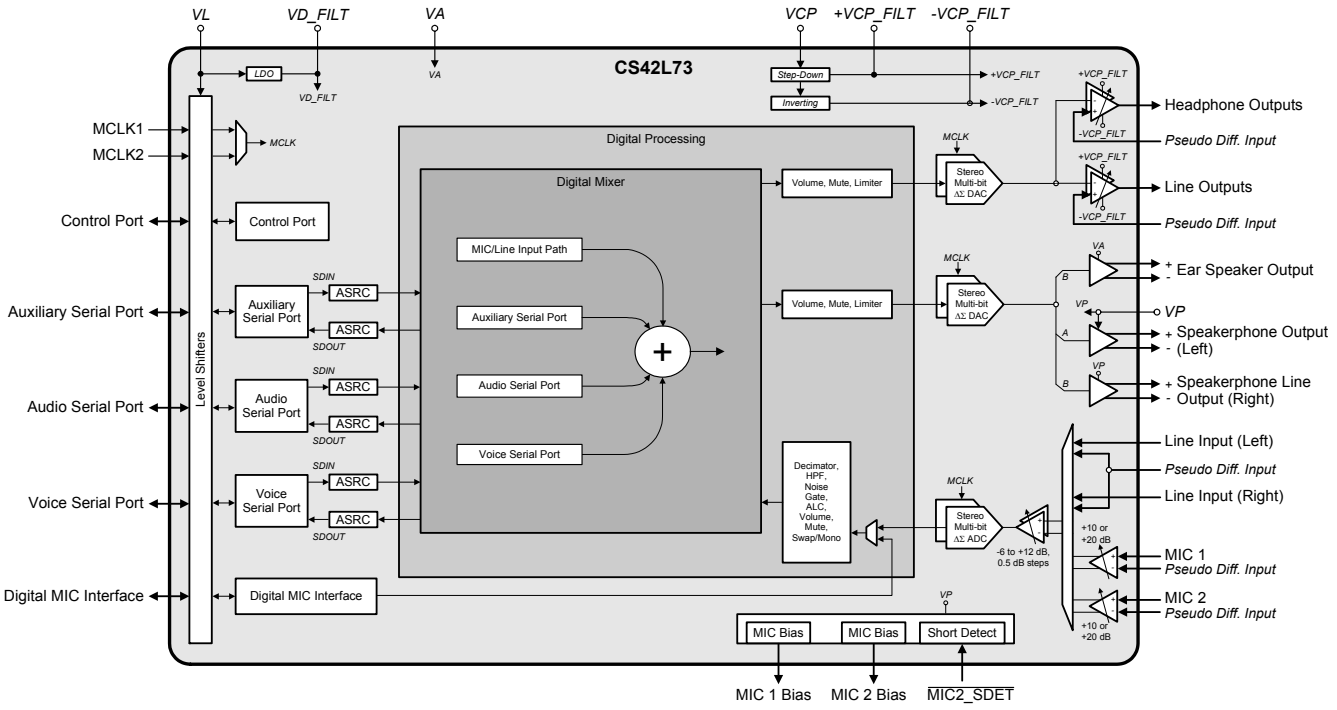
Applications

- Smart phones, ultramobile PCs, and mobile Internet devices

System Features

- Native (no PLL required) support for 6/12/24 MHz, 13/26 MHz, and 19.2/38.4 MHz master clock rates and typical audio clock rates
- Integrated high-efficiency power management reduces power consumption
  - Internal LDO regulator to reduce internal digital operating voltage to VL/2 V
  - Step-down charge pump provides low headphone/line out supply voltage
  - Inverting charge pump accommodates low system voltage by providing negative rail for HP and line amplifier
- Flexible speakerphone amplifier powering
  - 3.00–5.25 V range
  - Independent cycling
- Power-down management
  - Individual controls for ADCs, digital mic interface, mic bias generators, serial ports, and output amplifiers and associated DACs
- Programmable thermal overload notification
- High-speed I²C™ control port (400 kHz)

(**Features** continued on page 2)



Stereo Analog-to-Digital Features

- 91-db dynamic range (A-weighted)
- -85 dB THD+N
- Independent ADC channel control
- 2:1 stereo analog input MUX
- Stereo line input: Shared pseudodifferential reference input
- Dual analog mic inputs
  - Pseudodifferential or single-ended
  - Two, independent, programmable, low-noise mic bias outputs
  - Mic short detect to support headset button
- Analog programmable gain amplifier (PGA) (+12 to -6 dB in 0.5 dB steps)
- +10 dB or +20 dB analog mic boost in addition to PGA gain settings
- Programmable automatic level control (ALC)
  - Noise gate for noise suppression
  - Programmable threshold and attack/release rates

Dual Digital Microphone Interface

- Programmable clock rate: Integer divide by 2 or 4 of internal MCLK

Stereo DAC to Headphone Amplifier

- 94-dB dynamic range (A-weighted)
- -81 dB THD+N into 32 Ω
- Integrated step-down/inverting charge pump
- Class H amplifier, automatic supply adjustment
  - High efficiency
  - Low EMI
- Pseudodifferential ground-centered outputs
- High HP power output at -70/-81 dB THD+N
  - 2 x 16/8.1 mW into 16/32 Ω @ 1.8 V
- Pop and click suppression
- Analog volume control (+12 to -50 dB in 1 dB steps; to -76 dB in 2 dB steps) with zero-cross transitions
- Digital volume control (+12 to -102 dB in 0.5 dB steps) with soft-ramp transitions
- Programmable peak-detect and limiter

Stereo DAC to Line Outputs

- 97 dB dynamic range (A-weighted)
- -86 dB THD+N
- Class-H amplifier
- Pseudodifferential ground-centered outputs
- 1-V_{RMS} line output @ 1.8 V
- Pop and click suppression
- Analog volume control (+12 to -50 dB in 1 dB steps; to -76 dB in 2 dB steps) with zero-cross transitions
- Digital volume control (+12 to -102 dB in 0.5 dB steps) with soft-ramp transitions
- Programmable peak-detect and limiter

Mono DAC to Ear Speaker Amplifier

- High-power output at -70 dB (0.032%) THD+N: 45 mW into 16 Ω @ 1.8 V
- Pop and click suppression
- Digital volume control (+12 to -102 dB in 0.5 dB steps) with soft-ramp transitions
- Programmable peak-detect and limiter

Mono DAC to Speakerphone Amplifier

- High output power at ≤1% THD+N: 1.06/0.76/0.59 W into 8 Ω @ 5.0/4.2/3.7 V
- Direct battery-powered operation
- Pop and click suppression
- Digital volume control (+12 to -102 dB in 0.5 dB steps) with soft-ramp transitions
- Programmable peak-detect and limiter

Mono DAC-to-Speakerphone Line Output

- 84 dB dynamic range (A-weighted)
- -65 dB THD+N
- High voltage (2 V_{RMS} @ VA = 1.8 V, VP = 3.7 V) line output to ensure maximum output from a wide variety of external amplifiers
- Pop and click suppression
- Digital volume control (+12 to -102 dB in 0.5 dB steps) with soft-ramp transitions
- Programmable peak-detect and limiter

Serial Ports

- Three independent serial ports: auxiliary serial port (XSP), audio serial port (ASP), and voice serial port (VSP)
- 8.00, 11.025, 12.00, 16.00, 22.05, 24.00, 32.00, 44.10, and 48.00 kHz sample rates
- All ports support master or slave operation with I²S interface
- XSP and VSP support slave operation with PCM interface
- XSP and ASP are stereo-input/stereo-output to/from digital mixer
- VSP is mono-input/stereo-output to/from digital mixer
- Integrated asynchronous sample rate converters

General Description

The CS42L73 is a highly integrated, low-power, audio and telephony CODEC for portable applications such as smartphones and ultramobile personal computers.

The CS42L73 features a **flexible clocking architecture**, allowing the device to use reference clock frequencies of 6, 12, 24, 13, 26, 19.2, or 38.4 MHz, or any standard audio master clock. As many as two reference/master clock sources may be connected; either one can be selected to drive the internal clocks and processing rate of the CS42L73. Thus, multiple master clock sources within a system can be dynamically activated and deactivated to minimize system-level power consumption.

Three asynchronous bidirectional serial ports (auxiliary, audio, and voice serial ports (XSP, ASP, and VSP, respectively) support multiple clock domains of various digital audio sources or destinations. Three low-latency, fast-locking, integrated **high-performance asynchronous sample rate converters** synchronize and convert the audio samples to the internal processing rate of the CS42L73.

A stereo line input or two mono (one stereo) mic inputs are routed to a **stereo ADC**. The mic inputs may be selectively preamplified by +10 or +20 dB. Two independent, low-noise mic bias voltage supplies are also provided. A PGA is applied to the inputs before they reach the ADC.

The **stereo input path** that follows the stereo ADC begins with a multiplexer to selectively choose data from a **digital mic interface**. Following the multiplexer, the data is decimated, selectively DC high-pass filtered, channel-swapped or mono-to-stereo routed (fanned-out), and volume adjusted or muted. The volume levels can be automatically adjusted via a programmable ALC and noise gate.

A **digital mixer** is used to mix and route the CS42L73's inputs (analog inputs to ADC, digital mic, or serial ports) to outputs (DAC-fed amplifiers or serial ports). There is independent attenuation on each mixer input for each output.

The processing along the **output paths** from the digital mixer to the **two stereo DACs** includes volume adjustment and mute control. A peak-detector can be used to automatically adjust the volume levels via a programmable limiter.

The first stereo DAC feeds the **stereo headphone and line output amplifiers**, which are powered from a dedicated positive supply. An integrated **charge pump** provides a negative supply. This allows a ground-centered analog output with a wide signal swing, and eliminates external DC-blocking capacitors while reducing pops and clicks. Tri-level Class H amplification is used to reduce power consumption under low-signal-level conditions. Analog volume controls are provided on the stereo headphone and line outputs.

The second stereo DAC feeds several mono outputs. The left channel of the DAC sources a **mono, differential-drive, speakerphone amplifier** for driving the handset speakerphone. The right channel sources a **mono, differential-drive, earphone amplifier** for driving the handset earphone. The right channel is also routed to a **mono, differential-drive, speakerphone line output**, which may be connected to an external amplifier to implement a stereo speakerphone configuration when it is used in conjunction with the integrated speakerphone amplifier.

The CS42L73 implements **robust power management** to achieve ultralow power consumption. High granularity in power-down controls allows individual functional blocks to be powered down when unused. The internal low-dropout regulator (LDO) saves power by running the internal digital circuits at half the logic interface supply voltage (VL/2).

A high-speed **I^2C control port** interface capable of up to 400 kHz operation facilitates register programming.

The CS42L73 is available in space-saving 64-ball WLCSP and 65-ball FBGA packages for the commercial (-40° to +85° C) grade.

TABLE OF CONTENTS

1. PACKAGE PIN/BALL ASSIGNMENTS AND CONFIGURATIONS 12
1.1 64-Ball Wafer-Level Chip Scale Package (WLCSP) 12
1.2 65-Ball Fine-Pitch Ball Grid Array (FBGA) Package 13
1.3 Pin/Ball Descriptions 14
1.4 Digital Pin/Ball I/O Configurations 16
2. TYPICAL CONNECTION DIAGRAM 17
2.1 Low-Profile Charge-Pump Capacitors 18
2.2 Ceramic Capacitor Derating 18
3. CHARACTERISTIC AND SPECIFICATIONS 19
4. APPLICATIONS 41
4.1 Overview 41
4.1.1 Basic Architecture 41
4.1.2 Line and Microphone Inputs 41
4.1.3 Line and Headphone Outputs (Class H, Ground-Centered Amplifiers) 41
4.1.4 Digital Mixer 41
4.1.5 Power Management 41
4.2 Internal Master Clock Generation 42
4.3 Thermal Overload Notification 42
4.4 Pseudodifferential Outputs 43
4.5 Class H Amplifier 44
4.5.1 Power Control Options 44
4.5.1.1 Standard Class AB Operation (Mode 001, 010, and 011) 45
4.5.1.2 Adapt-to-Volume Settings (Mode 000) 45
4.5.1.3 Adapt-to-Output Signal (Mode 111) 46
4.5.2 Power Supply Transitions 46
4.5.3 Efficiency 49
4.6 DAC Limiter 49
4.7 Analog Output Current Limiter 51
4.8 Serial Ports 51
4.8.1 Power Management 51
4.8.2 I/O 51
4.8.3 High-impedance Mode 52
4.8.4 Master and Slave Timing 52
4.8.4.1 SCLK = MCLK Modes 53
4.8.5 Serial Port Sample Rates and Master Mode Settings 53
4.8.6 Formats 54
4.8.6.1 I²S Format 55
4.8.6.2 PCM Format 55
4.8.7 Mono/Stereo 57
4.8.8 Data Bit Depths 57
4.8.8.1 I²S Format Bit Depths 57
4.8.8.2 PCM Format Bit Depths 58
4.9 Asynchronous Sample Rate Converters (ASRCs) 59
4.10 Input Paths 59
4.10.1 Input Path Source Selection and Powering 59
4.10.2 Digital Microphone (DMIC) Interface 60
4.10.2.1 DMIC Interface Description 60
4.10.2.2 DMIC Interface Signaling 60
4.10.2.3 DMIC Interface Powering 60
4.10.2.4 DMIC Interface Clock Generation 61
4.11 Digital Mixer 61
4.11.1 Mono and Stereo Paths 63

4.11.2 Mixer Input Attenuation Adjustment 63
4.11.3 Powered-Down Mixer Inputs 64
4.11.4 Avoiding Mixer Clipping 64
4.11.5 Mixer Attenuation Values 65
4.12 Recommended Operating Procedures 65
4.12.1 Initial Power-Up Sequence 65
4.12.2 Power-Up Sequence (xSP to HP/LO) 66
4.12.3 Power-Down Sequence (xSP to HP/LO) 67
4.12.4 Recommended Sequence for Modification of the MCLK Signal 67
4.12.5 Microphone Enabling/Switching Sequence 68
4.12.6 Final Power-Down Sequence 68
4.13 Using MIC2_SDET as Headphone Plug Detect 69
4.14 Headphone Plug Detect and Mic Short Detect 70
4.15 Interrupts 70
4.16 Control Port Operation 71
4.16.1 I²C Control 71
4.17 Fast Start Mode 73
4.18 Headphone High-Impedance Mode 75
5. REGISTER QUICK REFERENCE 76
6. REGISTER DESCRIPTION 81
6.1 Fast Mode Enable (Address 00h) 81
6.1.1 Test Bits 81
6.2 Device ID A and B (Address 01h), C and D (Address 02h), and E (Address 03h) (Read Only) 81
6.2.1 Device I.D. (Read Only) 81
6.3 Revision ID (Address 05h) (Read Only) 81
6.3.1 Alpha Revision (Read Only) 81
6.3.2 Metal Revision (Read Only) 81
6.4 Power Control 1 (Address 06h) 82
6.4.1 Power Down ADC x 82
6.4.2 Power Down Digital Mic x 82
6.4.3 Discharge Filt+ Capacitor 82
6.4.4 Power Down Device 82
6.5 Power Control 2 (Address 07h) 83
6.5.1 Power Down MICx Bias 83
6.5.2 Power Down VSP 83
6.5.3 Power Down ASP SDOUT Path 83
6.5.4 Power Down ASP SDIN Path 83
6.5.5 Power Down XSP SDOUT Path 83
6.5.6 Power Down XSP SDIN Path 83
6.6 Power Control 3 and Thermal Overload Threshold Control (Address 08h) 84
6.6.1 Thermal Overload Threshold Settings 84
6.6.2 Power Down Thermal Sense 84
6.6.3 Power Down Speakerphone Line Output 84
6.6.4 Power Down Ear Speaker 84
6.6.5 Power Down Speakerphone 84
6.6.6 Power Down Line Output 85
6.6.7 Power Down Headphone 85
6.7 Charge Pump Frequency and Class H Configuration (Address 09h) 85
6.7.1 Charge Pump Frequency 85
6.7.2 Adaptive Power Adjustment 85
6.8 Output Load, Mic Bias, and MIC2 Short Detect Configuration (Address 0Ah) 86
6.8.1 VP Supply Minimum Voltage Setting 86
6.8.2 Speakerphone Light Load Mode Enable 86
6.8.3 Mic Bias Output Control 86

6.8.4 Short Detect Automatic Mute Control 86
6.9 Digital Mic and Master Clock Control (Address 0Bh) 87
6.9.1 Digital Mic Shift Clock Divide Ratio 87
6.9.2 Master Clock Source Selection 87
6.9.3 Master Clock Divide Ratio 87
6.9.4 Master Clock Disable 87
6.10 XSP Control (Address 0Ch) 88
6.10.1 Tristate XSP Interface 88
6.10.2 XSP Digital Interface Format 88
6.10.3 XSP PCM Interface Mode 88
6.10.4 XSP PCM Format Bit Order 88
6.10.5 XSP SCLK Source Equals MCLK 88
6.11 XSP Master Mode Clocking Control (Address 0Dh) 89
6.11.1 XSP Master/Slave Mode 89
6.11.2 XSP Master Mode Clock Control Dividers 89
6.12 ASP Control (Address 0Eh) 89
6.12.1 Tristate ASP Interface 89
6.12.2 ASP Sample Rate 90
6.12.3 ASP SCLK Source Equals MCLK 90
6.13 ASP Master Mode Clocking Control (Address 0Fh) 90
6.13.1 ASP Master/Slave Mode 90
6.13.2 ASP Master Mode Clock Control Dividers 90
6.14 VSP Control (Address 10h) 91
6.14.1 Tristate VSP Interface 91
6.14.2 VSP Digital Interface Format 91
6.14.3 VSP PCM Interface Mode 91
6.14.4 VSP PCM Format Bit Order 91
6.14.5 VSP SDIN Location 92
6.14.6 VSP SCLK Source Equals MCLK 92
6.15 VSP Master Mode Clocking Control (Address 11h) 92
6.15.1 VSP Master/Slave Mode 92
6.15.2 VSP Master Mode Clock Control Dividers 92
6.16 VSP and XSP Sample Rate (Address 12h) 93
6.16.1 VSP Sample Rate 93
6.16.2 XSP Sample Rate 93
6.17 Miscellaneous Input and Output Path Control (Address 13h) 94
6.17.1 Digital Swap/Mono 94
6.17.2 Input Path Channel B=A 94
6.17.3 PREAMP and PGA Channel B=A 94
6.17.4 PGA Soft-Ramp 95
6.17.5 Analog Zero Cross 95
6.17.6 Digital Soft-Ramp 96
6.17.7 Analog Output Soft Ramp 96
6.18 ADC/Input Path Control (Address 14h) 97
6.18.1 PGA x Input Select 97
6.18.2 Boost x 97
6.18.3 Invert ADCx Signal Polarity 97
6.18.4 Input Path x Digital Mute 97
6.19 Mic PreAmp and PGA Volume Control: Channel A (Mic 1, Address 15h) and Channel B (Mic 2, Address 16h) 98
6.19.1 Mic PREAMP x Volume 98
6.19.2 PGAx Volume 98
6.20 Input Path x Digital Volume Control: Channel A (Address 17h) and B (Address 18h) 99
6.20.1 Input Path x Digital Volume Control 99

6.21 Playback Digital Control (Address 19h) 100
6.21.1 Speakerphone [A], Ear Speaker/Speakerphone Line Output [B] (SES) Playback Channels B=A 100
6.21.2 Headphone/Line Output (HL) Playback Channels B=A 100
6.21.3 Limiter Soft-Ramp Disable 100
6.21.4 Ear Speaker/Speakerphone Line Output Digital Mute 100
6.21.5 Speakerphone Digital Mute 101
6.21.6 Headphone/Line Output (HL) x Digital Mute 101
6.22 Headphone/Line Output (HL) x Digital Volume Control: Channel A (Address 1Ah) and B (Address 1Bh) 101
6.22.1 Headphone/Line Output (HL) x Digital Volume Control 101
6.23 Speakerphone Out [A] Digital Volume Control (Address 1Ch) 102
6.23.1 Speakerphone Out [A] Digital Volume Control 102
6.24 Ear Speaker/Speakerphone Line Output (ESL) [B] Digital Volume Control (Address 1Dh) 102
6.24.1 Ear Speaker/Speakerphone Line Output (ESL) [B] Digital Volume Control 102
6.25 Headphone Analog Volume Control: Channel A (Address 1Eh) and B (Address 1Fh) 103
6.25.1 Headphone x Analog Mute 103
6.25.2 Headphone x Analog Volume Control 103
6.26 Line Output Analog Volume Control: Channel A (Address 20h) and B (Address 21h) 104
6.26.1 Line Output x Analog Mute 104
6.26.2 Line Output x Analog Volume Control 104
6.27 Stereo Input Path Advisory Volume (Address 22h) 105
6.27.1 Stereo Input Path Advisory Volume 105
6.28 XSP Input Advisory Volume (Address 23h) 105
6.28.1 XSP Input Advisory Volume 105
6.29 ASP Input Advisory Volume (Address 24h) 106
6.29.1 ASP Input Advisory Volume 106
6.30 VSP Input Advisory Volume (Address 25h) 106
6.30.1 VSP Input Advisory Volume 106
6.31 Limiter Attack Rate Headphone/Line Output (HL) (Address 26h) 107
6.31.1 Limiter Attack Rate HL 107
6.32 Limiter Control, Release Rate Headphone/Line Output (HL) (Address 27h) 107
6.32.1 Peak Detect and Limiter HL 107
6.32.2 Peak Signal Limit All Channels HL 107
6.32.3 Limiter Release Rate HL 107
6.33 Limiter Min/Max Thresholds Headphone/Line Output (HL) (Address 28h) 108
6.33.1 Limiter Maximum Threshold HL 108
6.33.2 Limiter Cushion Threshold HL 108
6.34 Limiter Attack Rate Speakerphone [A] (Address 29h) 108
6.34.1 Limiter Attack Rate Speakerphone [A] 108
6.35 Limiter Control, Release Rate Speakerphone [A] (Address 2Ah) 109
6.35.1 Peak Detect and Limiter Speakerphone [A] 109
6.35.2 Peak Signal Limit All Channels Speakerphone 109
6.35.3 Limiter Release Rate Speakerphone [A] 109
6.36 Limiter Min/Max Thresholds Speakerphone [A] (Address 2Bh) 110
6.36.1 Limiter Maximum Threshold Speakerphone [A] 110
6.36.2 Limiter Cushion Threshold Speakerphone [A] 110
6.37 Limiter Attack Rate Ear Speaker/Speakerphone Line Output (ESL) [B] 111
6.37.1 Limiter Attack Rate ESL [B] 111
6.38 Limiter Control, Release Rate Ear Speaker/Speakerphone Line Output (ESL) [B] (Address 2Dh) 111
6.38.1 Peak Detect and Limiter ESL [B] 111
6.38.2 Limiter Release Rate ESL [B] 111
6.39 Limiter Min/Max Thresholds Ear Speaker/Speakerphone Line Output (ESL) [B] 112

6.39.1 Limiter Maximum Threshold ESL [B] 112
6.39.2 Limiter Cushion Threshold ESL [B] 112
6.40 ALC Enable and Attack Rate AB (Address 2Fh) 113
6.40.1 ALC for Channels A and B (ALCx) 113
6.40.2 ALC Attack Rate for Channels A and B 113
6.41 ALC Release Rate AB (Address 30h) 113
6.41.1 ALC Release Rate for Channels A and B 113
6.42 ALC Threshold AB (Address 31h) 114
6.42.1 ALC Maximum Threshold for Channels A and B 114
6.42.2 ALC Minimum Threshold for Channels A and B 114
6.43 Noise Gate Control AB (Address 32h) 115
6.43.1 Noise Gate Enable for Channels A and B (NGx) 115
6.43.2 Noise gate Threshold and Boost for Channels A and B 115
6.43.3 Noise Gate Delay Timing for Channels A and B 115
6.44 ALC and Noise Gate Misc Control (Address 33h) 116
6.44.1 ALC Ganging of Channels A and B 116
6.44.2 Noise Gate Ganging of Channels A and B 116
6.44.3 ALCx Soft-Ramp Disable 116
6.44.4 ALCx Zero Cross Disable 116
6.45 Mixer Control (Address 34h) 117
6.45.1 VSP Mixer Output Stereo 117
6.45.2 XSP Mixer Output Stereo 117
6.45.3 Mixer Soft-Ramp Enable 117
6.45.4 Mixer Soft-Ramp Step Size/Period 117
6.46 Stereo Mixer Input Attenuation (Addresses 35h through 54h) 118
6.46.1 Stereo Mixer Input Attenuation 119
6.47 Mono Mixer Controls (Address 55h) 120
6.47.1 Speakerphone (SPK) Mixer, ASP Select 120
6.47.2 Speakerphone (SPK) Mixer, XSP Select 120
6.47.3 Ear Speaker/Speakerphone Line Output (ESL) Mixer, ASP Select 120
6.47.4 ESL Mixer, Auxiliary Serial Port (XSP) Select 120
6.48 Mono Mixer Input Attenuation (Addresses 56h through 5Dh) 121
6.48.1 Mono Mixer Input Attenuation 121
6.49 Interrupt Mask Register 1 (Address 5Eh) 122
6.50 Interrupt Mask Register 2 (Address 5Fh) 122
6.51 Interrupt Status Register 1 (Address 60h) 122
6.51.1 MIC2 Short Detect 122
6.51.2 Thermal Overload Detect 122
6.51.3 Digital Mixer Overflow 123
6.51.4 Input Path x Overflow 123
6.52 Interrupt Status Register 2 (Address 61h) 123
6.52.1 Voice ASRC Data Out Lock 123
6.52.2 Voice ASRC Data In Lock 123
6.52.3 Audio ASRC Data Out Lock 124
6.52.4 Audio ASRC Data In Lock 124
6.52.5 Auxiliary ASRC Data Out Lock 124
6.52.6 Auxiliary ASRC Data In Lock 124
6.53 Fast Mode 1 (Address 7Eh) 125
6.53.1 Fast Mode Bits 15:8 125
6.54 Fast Mode 2 (Address 7Fh) 125
6.54.1 Fast Mode Bits 7:0 125
7. PCB LAYOUT CONSIDERATIONS 125
7.1 Power Supply 125
7.2 Grounding 125

7.3 Layout With Fine-Pitch, Ball-Grid Packages 125
8. PERFORMANCE DATA 126
8.1 Analog Input Path Attributes 126
8.1.1 PGA Analog Volume Nonlinearity (DNL and INL) 126
8.2 Analog Mic/Line ADC and Digital Mic Input Path Attributes 127
8.2.1 Input Path Digital LPF Response 127
8.2.2 Input Path Digital HPF Response 128
8.3 Core Circuitry Attributes 129
8.3.1 ASRC Attributes 129
8.3.1.1 Response 129
8.3.1.2 Group Delay 130
8.3.1.3 Lock Time 130
8.4 Analog Output Paths Attributes 131
8.4.1 DAC Digital LPF Response 131
8.4.2 DAC HPF Response 132
8.4.3 Output Analog Volume Nonlinearity (DNL and INL) 132
8.4.4 Startup Times 133
9. PARAMETER DEFINITIONS 134
10. PACKAGE DIMENSIONS 135
10.1 WLCSP Package 135
10.2 FBGA Package 136
11. THERMAL CHARACTERISTICS 137
12. ORDERING INFORMATION 137
13. REFERENCES 137
14. REVISION HISTORY 138

LIST OF FIGURES

Figure 1.Typical Connection Diagram 17
Figure 2.MICx Dynamic Range Test Configuration 24
Figure 3.Analog Input CMRR Test Setup 24
Figure 4.LINEIN_REF/MICx_REF Input Voltage Test Setup 24
Figure 5.Headphone Output Test Configuration 28
Figure 6.Line Output Test Configuration 29
Figure 7.Ear Speaker Output Test Configuration 30
Figure 8.Speakerphone and Speakerphone Line Output Test Configuration 32
Figure 9.Power Consumption Test Configuration 34
Figure 10.Power and Reset Sequencing 36
Figure 11.Digital Mic Interface Timing 37
Figure 12.Serial Port Interface Timing—I²S Format 39
Figure 13.Serial Port Interface Timing—PCM Format 39
Figure 14.I²C Control Port Timing 40
Figure 15.Single-Ended Output Configuration 43
Figure 16.Pseudodifferential Output Configuration 43
Figure 17.Class H Operation 44
Figure 18.Class H Control - Adapt-to-Volume Mode 45
Figure 19.VCP_FILT Transitions 47
Figure 20.VCP_FILT Hysteresis 48
Figure 21.Input Power vs. Output Power 49
Figure 22.Peak Detect & Limiter 50
Figure 23.HP Short Circuit Setup 51
Figure 24.Line Short Circuit Setup 51
Figure 25.Serial Port Busing when Mastering Timing 52
Figure 26.Serial Port Busing When Slave Timed 52

Figure 27.I²S Format 55
Figure 28.PCM Format—Mode 0 56
Figure 29.PCM Format—Mode 1 56
Figure 30.PCM Format—Mode 2 57
Figure 31.Digital Mic Interface Signaling 60
Figure 32.Digital Mixer Diagram 62
Figure 33.Connection Diagram for Using MIC2_SDET as Headphone Detect 69
Figure 34.Flow Diagram Showing the INT Pin State in Response to MIC2_SDET State Changes 69
Figure 35.Connection Diagram for Headphone Detect with Additional Short Detect 70
Figure 36.Example of Rising-Edge Sensitive, Sticky, Interrupt Status Bit Behavior 71
Figure 37.Control Port Timing, I²C Writes with Autoincrement 72
Figure 38.Control Port Timing, I²C Reads with Autoincrement 72
Figure 39.Control Port Timing, I²C Reads with Preamble and Autoincrement 73
Figure 40.Fast Start Pop 74
Figure 41.Start Up Transition Diagram 75
Figure 42.PGA DNL 126
Figure 43.PGA INL 126
Figure 44.PGA + Preamp (+10 dB) DNL 126
Figure 45.PGA + Preamp (+10 dB) INL 126
Figure 46.PGA + Preamp (+20 dB) DNL 127
Figure 47.PGA + Preamp (+20 dB) INL 127
Figure 48.Input Path LPF Frequency Response 127
Figure 49.Input Path LPF Stopband Rejection 127
Figure 50.Input Path LPF Transition Band 128
Figure 51.Input Path LPF Transition Band Detail 128
Figure 52.Input Path HPF Frequency Response 128
Figure 53.ASRC Frequency Response 129
Figure 54.ASRC Passband Frequency Response 129
Figure 55.ASRC Group Delay vs. Serial Port and Internal Sample Rates 130
Figure 56.DAC LPF Frequency Response 131
Figure 57.DAC LPF Stopband Rejection to 1x Fs 131
Figure 58.DAC LPF Stopband Rejection to 3x Fs 131
Figure 59.DAC HPF Frequency Response 132
Figure 60.HPOUTx DNL (-50 to +12 dB) 132
Figure 61.HPOUTx DNL (-76 to -52 dB) 132
Figure 62.HPOUTx INL (-50 to +12 dB) 132
Figure 63.HPOUTx INL (-76 to -52 dB) 132
Figure 64.LINEOUTx DNL (-50 to +12 dB) 133
Figure 65.LINEOUTx DNL (-76 to -52 dB) 133
Figure 66.LINEOUTx INL (-50 to +12 dB) 133
Figure 67.LINEOUTx INL (-76 to -52 dB) 133

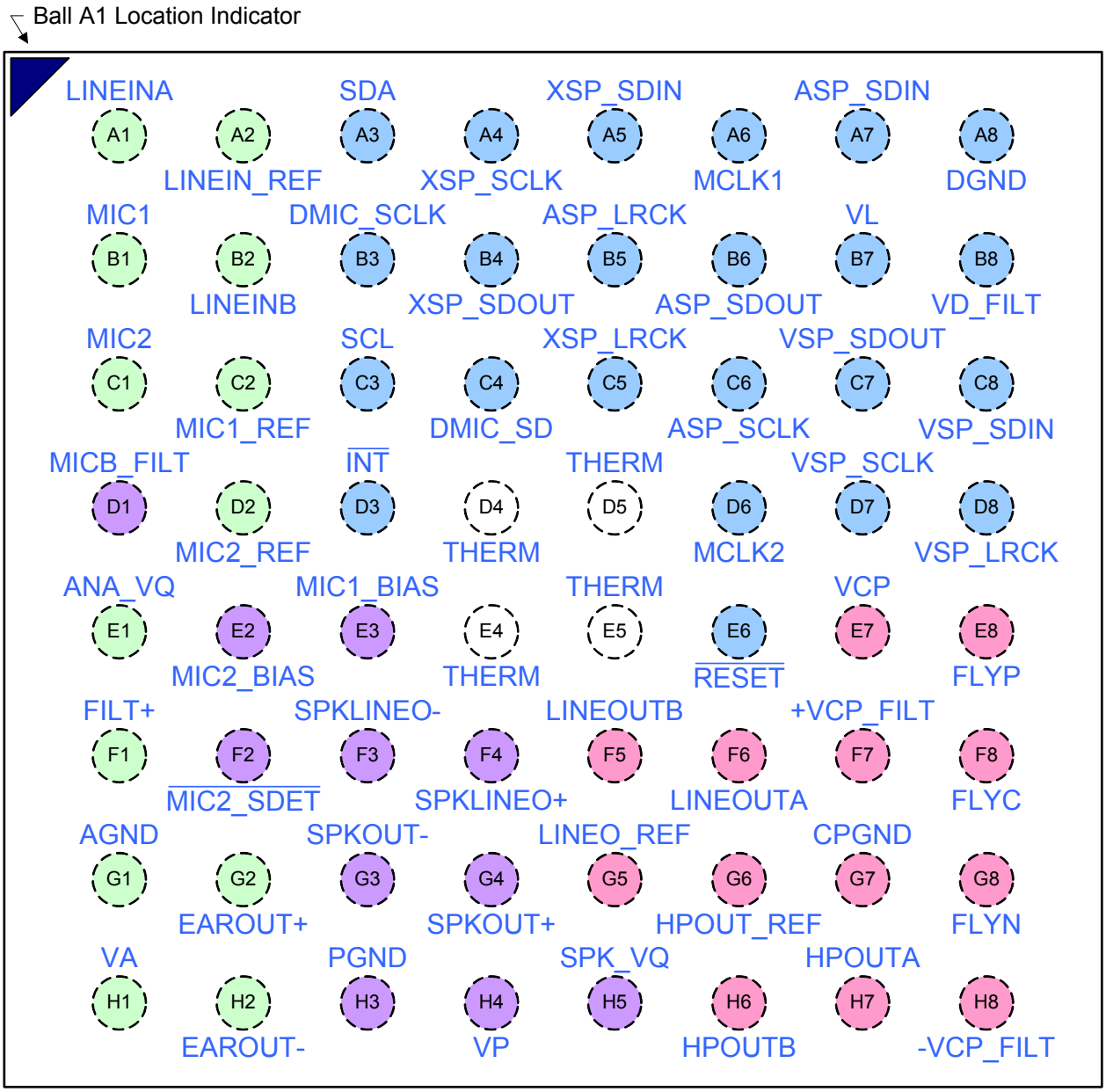
LIST OF TABLES

Table 1. Internal Master Clock Generation 42
Table 2. Example of Impedance in Reference Path 44
Table 3. Current through VCP with Varying Short Circuits 51
Table 4. Supported MCLK1/MCLK2 Rates for Pre-MCLK Mode 53
Table 5. Serial Port Rates and Master Mode Settings 53
Table 6. Actual xSP_LRCK Rate/Deviation Selector for Note 3 54
Table 7. Supported Serial Port Formats 54
Table 8. Input Path Source Select and Digital Power States 59
Table 9. Digital Mic Interface Power States 60
Table 10. Digital Microphone Interface Clock Generation 61

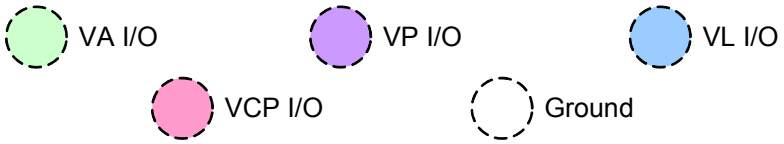
Table 11. Digital Mixer Soft Ramp Rates 63
Table 12. Digital Mixer Nonclipping Attenuation Settings 64
Table 13. Start Up Times 73
Table 14. Start Up Transition Values 75
Table 15. ASRC Lock Times 130
Table 16. Analog Output Startup Times 133
Table 17. WLCSP Package Dimensions 135
Table 18. FBGA Package Dimensions 136

1. PACKAGE PIN/BALL ASSIGNMENTS AND CONFIGURATIONS

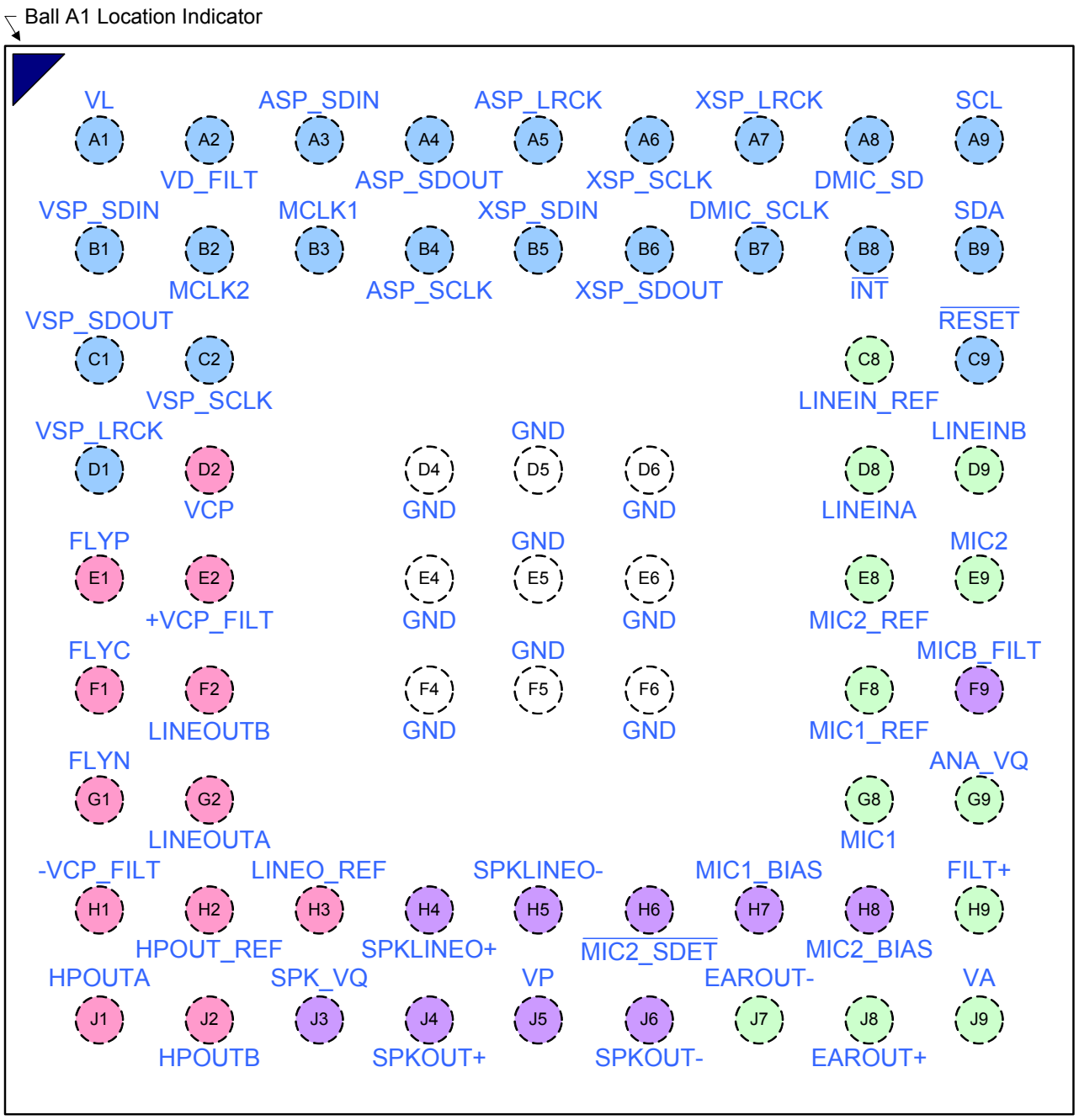
1.1 64-Ball Wafer-Level Chip Scale Package (WLCSP)



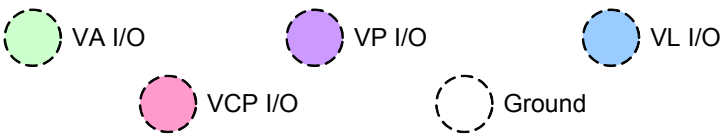
Top-Down
(Though Package)
View



1.2 65-Ball Fine-Pitch Ball Grid Array (FBGA) Package



Top-Down
(Though Package)
View



1.3 Pin/Ball Descriptions

Name	Location		Description
	WLCSP	FBGA	
MCLK1 MCLK2	A6 D6	B3 B2	**High Speed Clock** (*Input*). Potential clock sources for the converters and the device core. Clock source for optional serial port mastering.
$\overline{\text{RESET}}$	E6	C9	**Reset** (*Input*). The device enters a low-power mode when this pin is driven low.
SCL	C3	A9	**Serial Control Port Clock** (*Input*). Serial clock for the I²C control interfaces.
SDA	A3	B9	**Serial Control Data** (*Input*/*Output*). SDA is the bidirectional data pin for the I²C control interface.
$\overline{\text{INT}}$	D3	B8	**Interrupt Request** (*Output*). Open-drain active low interrupt request output.
LINEINA LINEINB	A1 B2	D8 D9	**Analog Line Inputs, A and B (LEFT and RIGHT)** (*Input*). The full-scale level is specified in the Analog Input Characteristics specification table.
LINEIN_REF	A2	C8	**Analog Line Input Pseudodifferential Reference** (*Input*). Ground reference for the analog line input buffers LINEINA and LINEINB.
MIC1 MIC2	B1 C1	G8 E9	**Microphone Inputs 1 and 2** (*Input*). The handset (MIC1) and headset (MIC2) microphone signal inputs. The full-scale level is specified in the Analog Input Characteristics specification table.
MIC1_REF MIC2_REF	C2 D2	F8 E8	**Microphone Inputs 1 and 2 Pseudodifferential References** (*Input*). Ground references for the microphone inputs MIC1 and MIC2.
MIC1_BIAS MIC2_BIAS	E3 E2	H7 H8	**Microphone Bias Voltages 1 and 2** (*Output*). Bias voltage for the microphones MIC1 and MIC2.
$\overline{\text{MIC2\_SDET}}$	F2	H6	**Microphone 2 Short Detect** (*Input*). Transitions on this input can be configured to cause interrupts that represent the pressing and releasing of a button that shorts the headset microphone to ground.
DMIC_SCLK	B3	B7	**Digital Mic Serial Clock** (*Output*). The high-speed clock output to the digital microphone(s).
DMIC_SD	C4	A8	**Digital Mic Serial Data** (*Input*). The serialized data input from the digital microphone(s).
XSP_SCLK	A4	A6	**Auxiliary Serial Port (XSP), Serial Clock** (*Input/Output*). Serial shift clock for the interface.
XSP_LRCK	C5	A7	**XSP, Left/Right Clock** (*Input/Output*). Identifies the start of each serialized PCM data word. When the I²S interface format is selected, this signal also indicates which channel, Left or Right, is currently active on the serial PCM audio data lines.
XSP_SDIN	A5	B5	**XSP, Data Input** (*Input*). Input for two's complement serial PCM audio data.
XSP_SDOUT	B4	B6	**XSP, Data Output** (*Output*). Output for two's complement serial PCM audio data.
ASP_SCLK	C6	B4	**Audio Serial Port (ASP), Serial Clock** (*Input/Output*). Serial shift clock for the interface.
ASP_LRCK	B5	A5	**ASP, Left/Right Clock** (*Input/Output*). Identifies the start of each serialized PCM data word and indicates which channel, Left or Right, is currently active on the serial PCM audio data lines.
ASP_SDIN	A7	A3	**ASP, Data Input** (*Input*). Input for two's complement serial PCM audio data.
ASP_SDOUT	B6	A4	**ASP, Data Output** (*Output*). Output for two's complement serial PCM audio data.
VSP_SCLK	D7	C2	**Voice Serial Port (VSP), Serial Clock** (*Input/Output*). Serial shift clock for the interface.
VSP_LRCK	D8	D1	**Voice Serial Port, Left/Right Clock** (*Input/Output*). Identifies the start of each serialized PCM data word. When the I²S interface format is selected, this signal also indicates which channel, Left or Right, is currently active on the serial PCM audio data lines.
VSP_SDIN	C8	B1	**VSP, Data Input** (*Input*). Input for two's complement serial PCM audio data.
VSP_SDOUT	C7	C1	**VSP, Data Output** (*Output*). Output for two's complement serial PCM audio data.
HPOUTA HPOUTB	H7 H6	J1 J2	**Headphone Audio Output** (*Output*). The full-scale output level is specified in the HP Output Characteristics specification table.
HPOUT_REF	G6	H2	**Pseudodifferential Headphone Output Reference** (*Input*). Ground reference for the headphone amplifiers.

Name	Location		Description
	WLCSP	FBGA	
LINEOUTA LINEOUTB	F6 F5	G2 F2	**Line Audio Output** (*Output*). The full-scale output level is specified in the Line Output Characteristics specification table.
LINEO_REF	G5	H3	**Pseudodifferential Line Output Reference** (*Input*). Ground reference for the line amplifiers.
EAROUT+ EAROUT-	G2 H2	J8 J7	**Ear Speaker Audio Output** (*Output*). The full-scale output level is specified in the Ear Speaker Output Characteristics specification table.
SPKOUT+ SPKOUT-	G4 G3	J4 J6	**Speakerphone Audio Output** (*Output*). The full-scale output level is specified in the Speakerphone Output Characteristics specification table.
SPKLINEO+ SPKLINEO-	F4 F3	H4 H5	**Speakerphone Audio Line Output** (*Output*). The full-scale output level is specified in the Speakerphone Line Output Characteristics specification table.
VA	H1	J9	**Analog Power** (*Input*). Power supply for the internal analog section.
VP	H4	J5	**Speakerphone Power** (*Input*). Power supply for the speakerphone output amplifier and mic bias generators.
VCP	E7	D2	**Step-down Charge Pump Power** (*Input*). Power supply for the step-down charge pump.
VL	B7	A1	**Digital Interface/Core Power** (*Input*). Power Supply for the serial PCM audio ports, I²C control port, and digital mic interface. Power supply for the digital core logic step-down regulator.
+VCP_FILT	F7	E2	**Step-down Charge Pump Filter Connection** (*Output*). Power supply from the step-down charge pump that provides the positive rail for the headphone and line amplifiers.
-VCP_FILT	H8	H1	**Inverting Charge Pump Filter Connection** (*Output*). Power supply from the inverting charge pump that provides the negative rail for the headphone and line amplifiers.
FLYP	E8	E1	**Charge Pump Cap Positive Node** (*Output*). Positive node for the headphone and line amplifiers' step-down charge pump's flying capacitor.
FLYC	F8	F1	**Charge Pump Cap Common Node** (*Output*). Common positive node for the headphone and line amplifiers' step-down and inverting charge pumps' flying capacitors.
FLYN	G8	G1	**Charge Pump Cap Negative Node** (*Output*). Negative node for the headphone and line amplifiers' inverting charge pump's flying capacitor.
VD_FILT	B8	A2	**Regulator Filter Connection** (*Output*). Power supply filter connection for the step-down regulator that provides the low voltage power to the digital section.
ANA_VQ	E1	G9	**Quiescent Voltage, Analog** (*Output*). Filter connection for the internal VA quiescent voltage.
SPK_VQ	H5	J3	**Quiescent Voltage, Speaker** (*Output*). Filter connection for the internal VP quiescent voltage.
FILT+	F1	H9	**Positive Voltage Reference** (*Output*). Positive reference voltage for the internal sampling circuits.
MICB_FILT	D1	F9	**Microphone Bias Source Voltage Filter** (*Output*). Filter connection for the internal quiescent voltage used for the MICx_BIAS outputs.
AGND	G1	N/A	**Analog Ground** (*Input*). Ground reference for the internal analog section.
PGND	H3	N/A	**Speakerphone Ground** (*Input*). Ground reference for the speakerphone and speakerphone line output amplifiers. Connect to ground plane(s) on board to conduct heat away from the part.
CPGND	G7	N/A	**Charge Pump Ground** (*Input*). Ground reference for the internal headphone and line amplifiers charge pump.
DGND	A8	N/A	**Digital Ground** (*Input*). Ground reference for the internal digital section.
GND	N/A	D4, D5, D6, E4, E5, E6, F4, F5, F6	**Ground**. Ground reference for internal analog (AGND), speakerphone and speakerphone line output amplifiers (PGND), internal headphone and line amplifiers (CPGND), and the internal digital section (DGND). These balls also provide thermal relief for the device. Connect to the Ground plane of the circuit board.
THERM	D4, D5, E4, E5	N/A	**Thermal Relief Balls**. Connect to the Ground plane of the circuit board. The Thermal Relief Balls are not electrically connected to the device.
NC	-	-	**No Connect**. No connection is required for these pins.

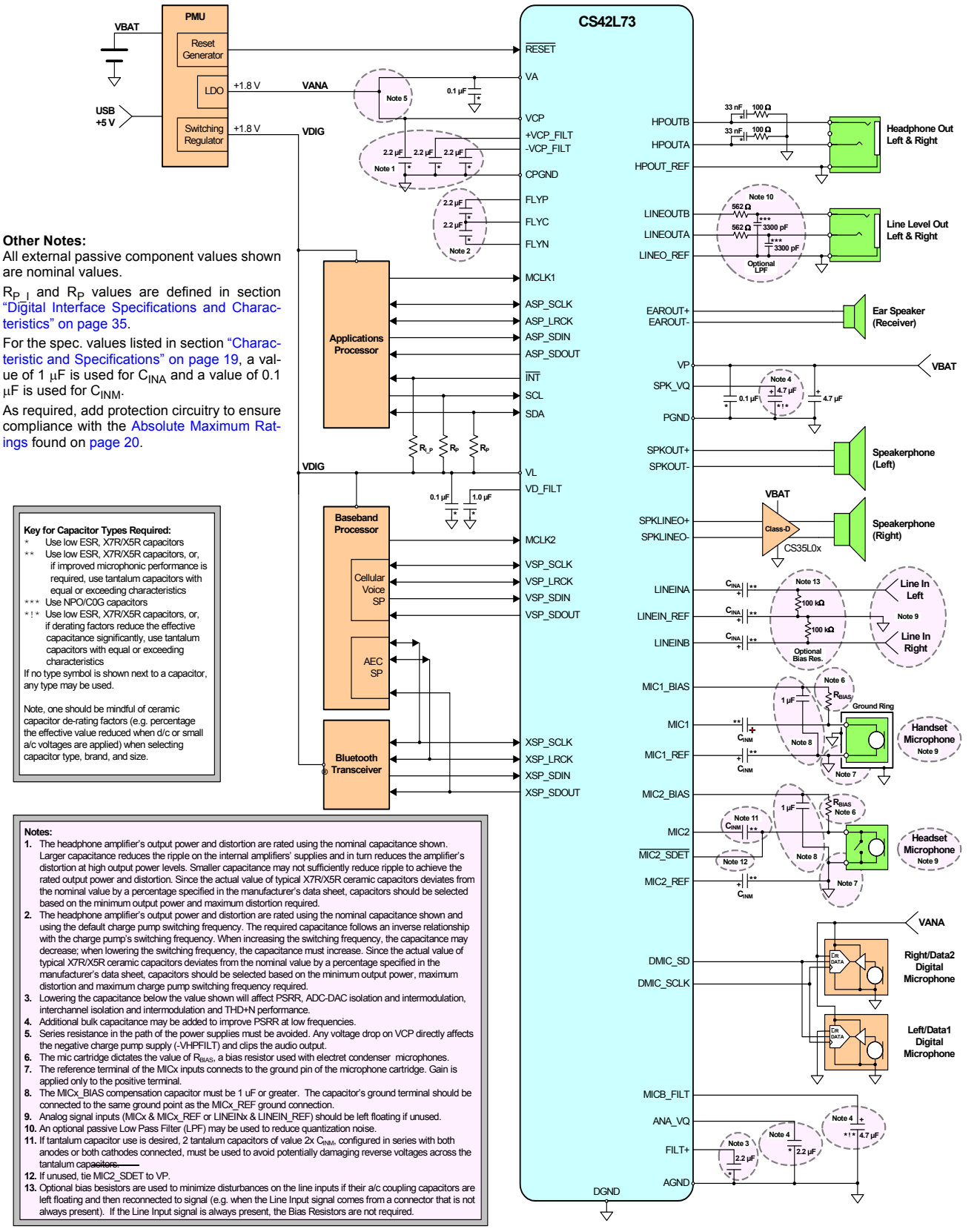
1.4 Digital Pin/Ball I/O Configurations

Power Supply	I/O Name	Direction	Internal Connections	Configuration
VL	MCLK1	Input	Weak Pull-down	Hysteresis on CMOS Input
	MCLK2	Input	Weak Pull-down	Hysteresis on CMOS Input
	$\overline{\text{RESET}}$	Input	-	Hysteresis on CMOS Input
	SCL	Input	-	Hysteresis on CMOS Input
	SDA	Input/Output	-	Hysteresis on CMOS Input/ CMOS Open-drain Output
	$\overline{\text{INT}}$	Output	Weak Pull-up	CMOS Open-drain Output
	XSP_SCLK	Input/Output	Weak Pull-down	Hysteresis on CMOS Input/CMOS Output
	XSP_LRCK	Input/Output	Weak Pull-down	Hysteresis on CMOS Input/CMOS Output
	XSP_SDIN	Input	Weak Pull-down	Hysteresis on CMOS Input
	XSP_SDOUT	Output	Weak Pull-down	Tristateable CMOS Output
	ASP_SCLK	Input/Output	Weak Pull-down	Hysteresis on CMOS Input/ CMOS Output
	ASP_LRCK	Input/Output	Weak Pull-down	Hysteresis on CMOS Input/CMOS Output
	ASP_SDIN	Input	Weak Pull-down	Hysteresis on CMOS Input
	ASP_SDOUT	Output	Weak Pull-down	Tristateable CMOS Output
	VSP_SCLK	Input/Output	Weak Pull-down	Hysteresis on CMOS Input/CMOS Output
	VSP_LRCK	Input/Output	Weak Pull-down	Hysteresis on CMOS Input/CMOS Output
	VSP_SDIN	Input	Weak Pull-down	Hysteresis on CMOS Input
	VSP_SDOUT	Output	Weak Pull-down	Tristateable CMOS Output
	DMIC_SCLK	Output	-	CMOS Output
	DMIC_SD	Input	Weak Pull-down	Hysteresis on CMOS Input

Notes:

- All outputs are disabled when $\overline{\text{RESET}}$ is active.
- Internal weak pull up/down minimum and typical resistances are 550 kΩ and 1 MΩ.
- Typical hysteresis is 500 mV within the 650 mV to 1.15 V window.
- The xSP_SCLK, xSP_LRCK, and xSP_SDOUT (x = X, A, or V) outputs may be disabled via register controls as described in sections “High-impedance Mode” on page 52 and “Master and Slave Timing” on page 52.
- Refer to specification table “Digital Interface Specifications and Characteristics” on page 35 for details on the digital I/O DC characteristics (output voltages/load-capacity, input switching threshold voltages, etc.). Inputs without integrated pull-ups/downs must not be left floating. All inputs must be driven or pulled (internally and/or externally) to a valid high or low level, as defined in the specification table.
- Refer to specification tables “Switching Specifications—Serial Ports—I²S Format” on page 38 on page 47, “Switching Specifications—Serial Ports—PCM Format” on page 39, and “Switching Specifications—Control Port” on page 40 for digital I/O AC characteristics (timing specifications).
- I/O voltage levels must not exceed the I/O’s corresponding power supply voltage. I/O voltage levels must not exceed the voltage listed in “Absolute Maximum Ratings” on page 20.

2. TYPICAL CONNECTION DIAGRAM



Other Notes:
All external passive component values shown are nominal values.

R_{P_I} and R_P values are defined in section "Digital Interface Specifications and Characteristics" on page 35.

For the spec. values listed in section "Characteristic and Specifications" on page 19, a value of 1 µF is used for C_{INA} and a value of 0.1 µF is used for C_{INM}.

As required, add protection circuitry to ensure compliance with the Absolute Maximum Ratings found on page 20.

Key for Capacitor Types Required:
* Use low ESR, X7R/X5R capacitors
** Use low ESR, X7R/X5R capacitors, or, if improved microphonic performance is required, use tantalum capacitors with equal or exceeding characteristics
*** Use NPO/C0G capacitors
! Use low ESR, X7R/X5R capacitors, or, if derating factors reduce the effective capacitance significantly, use tantalum capacitors with equal or exceeding characteristics
If no type symbol is shown next to a capacitor, any type may be used.

Note, one should be mindful of ceramic capacitor de-rating factors (e.g. percentage the effective value reduced when d/c or small a/c voltages are applied) when selecting capacitor type, brand, and size.

Notes:
1. The headphone amplifier's output power and distortion are rated using the nominal capacitance shown. Larger capacitance reduces the ripple on the internal amplifiers' supplies and in turn reduces the amplifier's distortion at high output power levels. Smaller capacitance may not sufficiently reduce ripple to achieve the rated output power and distortion. Since the actual value of typical X7R/X5R ceramic capacitors deviates from the nominal value by a percentage specified in the manufacturer's data sheet, capacitors should be selected based on the minimum output power and maximum distortion required.
2. The headphone amplifier's output power and distortion are rated using the nominal capacitance shown and using the default charge pump switching frequency. The required capacitance follows an inverse relationship with the charge pump's switching frequency. When increasing the switching frequency, the capacitance may decrease; when lowering the switching frequency, the capacitance must increase. Since the actual value of typical X7R/X5R ceramic capacitors deviates from the nominal value by a percentage specified in the manufacturer's data sheet, capacitors should be selected based on the minimum output power, maximum distortion and maximum charge pump switching frequency required.
3. Lowering the capacitance below the value shown will affect PSRR, ADC-DAC isolation and intermodulation, interchannel isolation and intermodulation and THD+N performance.
4. Additional bulk capacitance may be added to improve PSRR at low frequencies.
5. Series resistance in the path of the power supplies must be avoided. Any voltage drop on VCP directly affects the negative charge pump supply (-VHPFILT) and clips the audio output.
6. The mic cartridge dictates the value of R_{BIAS}, a bias resistor used with electret condenser microphones.
7. The reference terminal of the MICx inputs connects to the ground pin of the microphone cartridge. Gain is applied only to the positive terminal.
8. The MICx_BIAS compensation capacitor must be 1 uF or greater. The capacitor's ground terminal should be connected to the same ground point as the MICx_REF ground connection.
9. Analog signal inputs (MICx & MICx_REF or LINEINx & LINEIN_REF) should be left floating if unused.
10. An optional passive Low Pass Filter (LPF) may be used to reduce quantization noise.
11. If tantalum capacitor use is desired, 2 tantalum capacitors of value 2x C_{INM}, configured in series with both anodes or both cathodes connected, must be used to avoid potentially damaging reverse voltages across the tantalum capacitors.
12. If unused, tie MIC2_SDET to VP.
13. Optional bias besistors are used to minimize disturbances on the line inputs if their a/c coupling capacitors are left floating and then reconnected to signal (e.g. when the Line Input signal comes from a connector that is not always present). If the Line Input signal is always present, the Bias Resistors are not required.

Figure 1. Typical Connection Diagram

2.1 Low-Profile Charge-Pump Capacitors

The "Typical Connection Diagram" on page 17 shows that the recommended capacitor values for the charge pump circuitry are all 2.2 µF and the types are all X7R/X5R. Applications that require low-profile versions of these capacitors may use the following parts with a nominal height of only 0.5 mm:

Description: 2.2 µF ±20%, 6.3 V, X5R, 0402, Height = 0.5 mm
Manufacturer, Part Number:

- KEMET, C0402C225M9PAC

2.2 Ceramic Capacitor Derating

The Typical Connection Diagram Capacitor Key highlights that ceramic capacitor derating factors can significantly affect the in-circuit capacitance value and thus the performance of the CS42L73.

As is noted on the Typical Connection Diagram, the 4.7 µF ceramic capacitors used for ANA_VQ or SPKR_VQ affect low-frequency PSRR performance. Numerous types and brands of ceramic capacitors, under typical conditions, exhibit effective capacitances well below their tolerance of ±20%, with some being derated by as much as -50%. These same capacitors, when tested by a multimeter, read much closer to their rated value. A similar derating effect has not been observed with tantalum capacitors.

The amount of derating observed varied with manufacturer and physical size; larger capacitors performed better as did ones from Kemet Electronics Corp. and TDK Corp. of any size. This derating effect is described in datasheets and applications notes from capacitor manufacturers. For instance, as DC and AC voltages are varied from the standard test points (applied DC and AC voltages for standard test points vs. PSRR test are 0 V and 1 V_{RMS} @ 1 kHz vs. 0.9 V and ~1 mV_{RMS} @ 20 Hz to 20 kHz), it is documented that the capacitance varies significantly.

Based on these tests, the following ANA_VQ/SPKR_VQ capacitor parts are recommended for applications that require ceramic capacitors with the smallest PCB footprint:

Description: 4.7 µF ±20%, 6.3V, X5R, 0603
Manufacturer, Part Number:

- KEMET, C0603C475M9PAC
- TDK, C1608X5R0J475M

3. CHARACTERISTIC AND SPECIFICATIONS

RECOMMENDED OPERATING CONDITIONS

Test Conditions: GND = AGND = PGND = CPGND = DGND = 0 V; all voltages are with respect to ground (GND).

Parameters (Note 1) (Note 2)	Symbol	Min	Nom	Max	Units	Equivalent Tolerance from Nominal
DC Power Supplies						
Analog	VA	1.66	1.80	1.94	V	±7.8%
Speakerphone Amplifiers, Mic Bias Generators (Note 3) Mic Bias with High Voltage Selected and VP_MIN = 1b Otherwise	VP	 3.20 3.00	 - -	 5.25 5.25	 V	 - -
Charge Pump (Headphone and Lineout Amplifiers)	VCP	1.66	1.80	1.94	V	±7.8%
Digital Core, Serial/Control/Digital-Mic Interfaces	VL	1.66	1.80	1.94	V	±7.8%
Temperature						
Ambient Temperature (local to device) Commercial: CWZR	T_A	-40	-	+85	°C	-

Notes:

1. Device functional operation is guaranteed within these limits. Functionality is not guaranteed or implied outside of these limits. Operation outside of these limits may adversely affect device reliability.
2. "Parameter Definitions" on page 134 describes some parameters in detail.
3. The recommended operation range of the VP supply depends on how the CS42L73 is configured. If either mic bias is enabled (PDN_MIC1_BIAS = 0b or PDN_MIC2_BIAS = 0b) and the mic bias generators are set for their higher voltage (MIC_BIAS_CTRL = 1b), either VP must be held above 3.2 V or VP_MIN must be set to 0b. With this configuration and a VP level between 3.00 and 3.20 V, VP_MIN must be set to 0b to ensure the bias generators bypass one of their two LDO stages, ensuring there is enough headroom to avoid dropout. Refer to "Mic BIAS Characteristics" on page 26 for details on how much setting VP_MIN to 0b reduces PSRR performance.

ABSOLUTE MAXIMUM RATINGS

Test Conditions: GND = AGND = PGND = CPGND = DGND = 0 V; all voltages are with respect to ground (GND).

Parameters (Note 2)		Symbol	Min	Max	Units
DC Power Supply					
Analog, Charge Pump, Digital Core (LDO fed),		VA, VCP, VL	-0.3	2.22	V
Serial/Control/Digital-Mic Interfaces		VP (Note 4)	-0.3	5.6	V
Speakerphone Amplifiers, Mic Bias Generators					
Input Current	(Note 5)	I_{in}	-	±10	mA
Voltages Applied to I/Os					
External Voltage Applied	LINEINx, MICx, x_REF	V_{IN-AI}	AGND – 0.3	VA + 0.3	V
to Analog Input (Note 6)	$\overline{\text{MIC2\_SDET}}$	$V_{IN-AI-SD}$	PGND – 0.3	VP + 0.3	V
External Voltage	HPOUT, LINEOUT	V_{FLT-HP_LINE}	-VCP_FILT – 0.3	+VCP_FILT + 0.3	V
Applied to Analog	EAROUT	$V_{FLT-EAR}$	AGND – 0.3	VA + 0.3	V
Output (Note 7)	SPKOUT, SPKLINEO, MICx_BIAS	$V_{FLT-SPK\_MB}$	PGND – 0.3	VP + 0.3	V
External Voltage Applied to Digital Input	(Note 6)	V_{IN-DI}	-0.3	VL + 0.3	V
External Voltage Applied to Digital Output	(Note 7)	V_{FLT-DO}	-0.3	VL + 0.3	V
Temperature					
Ambient Operating Temperature	Commercial: CWZR	T_A	-50	+110	°C
(local to device, power applied)					
Storage Temperature (no power applied)		T_{stg}	-65	+150	°C

WARNING: Operation at or beyond these limits may result in permanent damage to the device. Normal operation is not guaranteed at these extremes.

Notes:

4. VP must be applied before VA is applied. VP must be removed after VA is removed.
5. Any pin except supplies. Transient currents of up to ±100 mA on the analog input pins will not cause SCR latch-up.
6. The maximum over/under voltage is limited by the input current.
7. • V_{FLT-x} is the applied voltage that causes a contention fault condition between its source and the CS42L73 output.
   • ±VCPFILT are specified in "DC Electrical Characteristics" on page 21.
   • The specification applies to both the signal and pseudodifferential reference pins, where applicable.

DC ELECTRICAL CHARACTERISTICS

Test Conditions: Connections to the CS42L73 are shown in the "Typical Connection Diagram" on page 17; GND = AGND = PGND = CPGND = DGND = 0 V; all voltages are with respect to ground (GND); VA = VCP = 1.80 V, VP = 3.70 V; T_A = +25 °C.

Parameters (Note 2)		Min	Typ	Max	Units
ANA_VQ Characteristics					
Nominal Voltage		-	VA/2	-	V
SPK_VQ Characteristics					
Nominal Voltage		-	VP/2	-	V
VCPFILT Characteristics (Note 8)					
VCP Mode	+VCPFILT	-	VCP	-	V
	-VCPFILT	-	-VCP	-	V
VCP/2 Mode	+VCPFILT	-	VCP/2	-	V
	-VCPFILT	-	-VCP/2	-	V
VCP/3 Mode	+VCPFILT	-	VCP/3	-	V
	-VCPFILT	-	-VCP/3	-	V
FILT+ Characteristics					
Nominal Voltage		-	VA	-	V
VD_FILT Characteristics					
Nominal Voltage		-	0.9	-	V
MICB_FILT Characteristics					
Nominal Voltage	MIC_BIAS_CTRL = 0b	-	2.00	-	V
	MIC_BIAS_CTRL = 1b	-	2.75	-	V
Analog Output Current Limiter Characteristics					
Current Limiter On Threshold (Note 9)		100	120	150	mA

Notes:

8. No load (from specification tables "Serial Port to Stereo HP Output Characteristics" on page 27 and "Serial Port to Stereo Line Output Characteristics" on page 29, R_L = ∞ Ω and C_L = 0 pF) connected to Headphone and Line Outputs (HPOUTx and LINEOUTx). Headphone Zobel Network remains connected.
9. See "Analog Output Current Limiter" on page 51.

ANALOG INPUT TO SERIAL PORT CHARACTERISTICS

Test Conditions (unless otherwise specified): Connections to the CS42L73 are shown in the "Typical Connection Diagram" on page 17; Input is a 1-kHz sine wave through the passive input filter shown in Figure 1; GND = AGND = PGND = CPGND = DGND = 0 V; all voltages are with respect to ground (GND); VA = 1.80 V; T_A = +25 °C; Measurement Bandwidth is 20 Hz to 20 kHz; Fs = 48 kHz (Note 10); ASP is used and is in slave mode with Fs_{ext} = 48 kHz; MIC_PREAMPx = +10 dB, PGAxVOL = 0 dB; Mixer Attenuation and Digital Volume = 0 dB, Digital Mute is disabled.

Parameters (Note 2) (Note 11)		Min	Typ	Max	Units
LINEINA/LINEINB to PGA to ADC					
Dynamic Range					
PGA Setting: 0 dB	A-weighted	85	91	-	dB
	unweighted	82	88	-	dB
PGA Setting: +12 dB	A-weighted	78	84	-	dB
	unweighted	75	81	-	dB
Total Harmonic Distortion + Noise					
PGA Setting: 0 dB	-1 dBFS	-	-85	-79	dB
	-60 dBFS	-	-28	-22	dB
PGA Setting: +12 dB	-1 dBFS	-	-81	-75	dB
Common Mode Rejection (Note 12)		-	40	-	dB
MIC1/MIC2 to PREAMP to PGA to ADC, MIC_PREAMPx = +10 dB Gain					
Dynamic Range (Note 13)					
PGA Setting: 0 dB	A-weighted	-	88	-	dB
	unweighted	-	86	-	dB
PGA Setting: +12 dB	A-weighted	-	78	-	dB
	unweighted	-	75	-	dB
Total Harmonic Distortion + Noise					
PGA Setting: 0 dB	-1 dBFS	-	-77	-	dB
PGA Setting: +12 dB	-1 dBFS	-	-64	-	dB
Common Mode Rejection (Note 12)		-	40	-	dB
MIC1/MIC2 to PREAMP to PGA to ADC, MIC_PREAMPx = +20 dB Gain					
Dynamic Range (Note 13)					
PGA Setting: 0 dB	A-weighted	-	82	-	dB
	unweighted	-	79	-	dB
PGA Setting: +12 dB	A-weighted	-	70	-	dB
	unweighted	-	67	-	dB
Total Harmonic Distortion + Noise					
PGA Setting: 0 dB	-1 dBFS	-	-71	-	dB
PGA Setting: +12 dB	-1 dBFS	-	-63	-	dB
Common Mode Rejection (Note 12)		-	40	-	dB
DC Accuracy					
Interchannel Gain Mismatch		-	0.2	-	dB
Gain Drift		-	±100	-	ppm/°C
Offset Error		-	352	-	LSB

ANALOG INPUT TO SERIAL PORT CHARACTERISTICS (CONTINUED)

Test Conditions (unless otherwise specified): Connections to the CS42L73 are shown in the "Typical Connection Diagram" on page 17; Input is a 1-kHz sine wave through the passive input filter shown in Figure 1; GND = AGND = PGND = CPGND = DGND = 0 V; all voltages are with respect to ground (GND); VA = 1.80 V; T_A = +25 °C; Measurement Bandwidth is 20 Hz to 20 kHz; Fs = 48 kHz (Note 10); ASP is used and is in slave mode with Fs_{ext} = 48 kHz; MIC_PREAMPx = +10 dB, PGAxVOL = 0 dB; Mixer Attenuation and Digital Volume = 0 dB, Digital Mute is disabled.

Parameters (Note 2) (Note 11)		Min	Typ	Max	Units
Input					
Interchannel Isolation (1 kHz)	LINEINA to LINEINB, PGAxVOL = +12 dB	-	90	-	dB
	MIC1 to MIC2, MIC_PREAMPx = +20 dB, PGAxVOL = +12 dB	-	80	-	
HP Amp to Analog Input Isolation (Note 14)	R_L = 3 kΩ	84	90	-	dB
	R_L = 16 Ω	77	83	-	dB
Full-scale Signal Input Voltage LINEINA/LINEINB (Note 15)	PGAxVOL = 0 dB	0.78•VA	0.82•VA	0.86•VA	V_{PP}
	PGAxVOL = +12 dB	-	0.198•VA	-	V_{PP}
Full-scale Signal Input Voltage MIC1/MIC2 (Note 15)	MIC_PREAMPx = +10 dB, PGAxVOL = +0 dB	-	0.258•VA	-	V_{PP}
	MIC_PREAMPx = +20 dB, PGAxVOL = +0 dB	-	0.081•VA	-	V_{PP}
	MIC_PREAMPx = +10 dB, PGAxVOL = +12 dB	-	0.064•VA	-	V_{PP}
	MIC_PREAMPx = +20 dB, PGAxVOL = +12 dB	-	0.020•VA	-	V_{PP}
LINEIN_REF/MICx_REF Input Voltage (Note 16)		-	-	0.300	V_{PP}
Input Impedance (Note 17), LINEINA/LINEINB	1 kHz	-	50	-	kΩ
Input Impedance (Note 17), MIC1/MIC2	1 kHz	-	1.0	-	MΩ
DC Voltage at Analog Input (Pin Floating)		-	0.50•VA	-	V
LINEINA/LINEINAB PSRR					
- 100 mV_{PP} signal AC-coupled to VA supply (Note 18)	217 Hz	-	50	-	dB
- LINEINA and LINEINB connected to LINEIN_REF	1 kHz	-	65	-	dB
- PGAxVOL = 0 dB	20 kHz	-	40	-	dB
MIC1/MIC2 PSRR					
- 100 mV_{PP} signal AC-coupled to VA supply (Note 18)	217 Hz	-	50	-	dB
- MICx connected to MICx_REF	1 kHz	-	65	-	dB
- MIC_PREAMPx = +20 dB, PGAxVOL = +12 dB	20 kHz	-	35	-	dB

Notes:

10. Fs is the sampling frequency used by the core and the A/D and D/A converters. For specifications, a default value of 48 kHz is used. Refer to section "Applications" on page 41 for a description of how Fs relates to the CS42L73's clock inputs.
11. Measures are referred to the applicable typical full-scale voltages. Applies to all THD+N and dynamic range values in the table.
12. Refer to Figure 3 below.
13. Includes noise from MICx_BIAS output through series 2.21 kΩ series resistor to MICx. Refer to Figure 2 below. Input signal is -60 dB down from corresponding full-scale voltage.
14. Measurement taken with the following analog gain settings:
    - LINEINA/LINEINB: PGAxVOL = +12 dB
    - MIC1/MIC2: MIC_PREAMPx= + 20 dB, PGAxVOL = +12 dB
    - HPxAVOL = +2 dB for R_L = 3 kΩ, -4 dB for R_L = 16 Ω
15. The full-scale input voltages given refer to the maximum voltage difference between the LINEINx/MICx and LINEIN_REF/MICx_REF pins. Providing an input signal at these pins that exceeds the full-scale input voltage will result in the clipping of the analog signal.
16. The PGA output clips if the voltage difference between the LINEINx/MICx and LINEIN_REF/MICx_REF signals exceeds the full-scale voltage specification. If the LINEIN_REF/MICx_REF signal level exceeds the specified maximum value, PGA linearity may be degraded and analog input performance may be adversely affected. Refer to Figure 4 below.
17. Measured between LINEINx/MICy and AGND. Input impedance can vary from nominal value by ±20%.
18. The PGA is biased with ANA_VQ, created by a resistor divider from the VA supply. Increasing the capacitance on ANA_VQ will increase the PSRR at low frequencies.

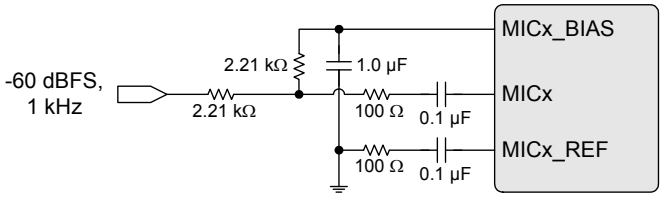


Figure 2. MICx Dynamic Range Test Configuration

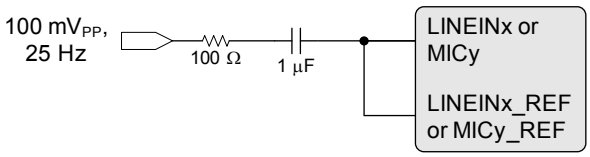


Figure 3. Analog Input CMRR Test Setup

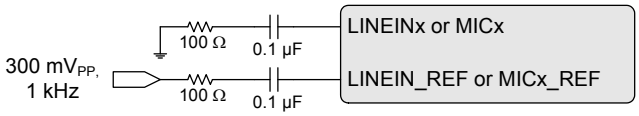


Figure 4. LINEIN_REF/MICx_REF Input Voltage Test Setup

STEREO-ADC AND DUAL-DIGITAL-MIC DIGITAL FILTER CHARACTERISTICS

Test Conditions (unless otherwise specified): Fs = 48 kHz (Note 10), f_{DMIC_SCLK} = 3.072 MHz (Note 19).

Parameters (Note 2)		Min	Typ	Max	Units
Low-Pass Filter Characteristics (Note 20)					
Frequency Response (20 Hz to 20 kHz)		-0.07	-	+0.02	dB
Passband	to -0.05 dB corner	-	0.41	-	Fs
	to -3.0 dB corner	-	0.49	-	Fs
Stopband (Note 21)		0.60	-	-	Fs
Stopband Attenuation		33	-	-	dB
Total Input Path Digital Filter Group Delay		-	4.3/Fs	-	s
High-Pass Filter Characteristics (Note 20) (Note 22)					
Passband	to -3.0 dB corner	-	$4.10x10^{-5}$	-	Fs
	to -0.05 dB corner	-	$3.57x10^{-4}$	-	Fs
Passband Ripple		-	-	0.01	dB
Phase Deviation @ 20 Hz		-	5.30	-	Deg
Filter Settling Time (input signal goes to 95% of its final value)		-	$12.2x10^{3}$/Fs	-	s

Notes:

19. Refer to section "Digital Microphone (DMIC) Interface" on page 60 for a description of how the digital mic shift clock frequency (f_{DMIC_SCLK}) relates to the CS42L73's internal master clock rate.
20. Responses are clock-dependent and will scale with Fs. Note that the response plots (Figures 48 to 52 on pages 127 and 128) have been normalized to Fs and can be denormalized by multiplying the X-axis scale by Fs.
21. Measurement Bandwidth is from Stopband to 3 Fs.
22. High-pass filter is applied after low-pass filter.

THERMAL OVERLOAD DETECT CHARACTERISTICS

Test Conditions: Connections to the CS42L73 are shown in the "Typical Connection Diagram" on page 17; GND = AGND = PGND = CPGND = DGND = 0 V; all voltages are with respect to ground (GND); VA = 1.80 V.

Parameters		Min	Typ	Max	Units
Thermal Overload Detect Threshold Characteristics					
Threshold Junction Temperature (T_J) (Note 23)	THMOVLD_THLD[1:0] = 00b	-	150	-	°C
	THMOVLD_THLD[1:0] = 01b	-	132	-	°C
	THMOVLD_THLD[1:0] = 10b	-	115	-	°C
	THMOVLD_THLD[1:0] = 11b	-	98	-	°C

Notes:

23. The thermal overload detect threshold temperature level can vary from the nominal value by ±10 °C.

ASRC DIGITAL FILTER CHARACTERISTICS

Test Conditions (unless otherwise specified): Fs = 48 kHz (Note 10); Fs_{ext} = 48 kHz (Note 24).

Parameters (Note 2) (Note 25)		Min	Typ	Max	Units
Low-Pass Filter Characteristics (Note 24)					
Frequency Response (0 Hz to 20 kHz)		-0.07	-	+0.04	dB
Passband	to -0.05 dB corner	-	0.48	-	Fs_{ext}
	to -3.0 dB corner	-	0.50	-	
Stopband		0.55	-	-	Fs_{ext}
Stopband Attenuation		125	-	-	dB
Total ASRC Group Delay		-	(Note 26)	-	s

Notes:

24. Fs_{ext} is the sample rate of the serial port (XSP, ASP, or VSP) interface.
25. Refer to Response plots in Figures 53 and 54 on page 129.
26. The equations for the group delay through the sample rate converters are:
    - Input (from the serial ports to the core): $6.9/Fs_{ext} + 3.0/Fs$
    - Output (from the core to the serial ports): $2.6/Fs_{ext} + 14.1/Fs$.

    A plot of ASRC group delay values for the extreme supported internal sample rates (Fs) and standard audio sample rates is found in section "Group Delay" on page 130.

MIC BIAS CHARACTERISTICS

Test Conditions (unless otherwise specified): Connections to the CS42L73 are shown in the "Typical Connection Diagram" on page 17; GND = AGND = PGND = CPGND = DGND = 0 V; all voltages are with respect to ground (GND); VA = 1.80 V, VP = 3.70 V; T_A = +25 °C; I_{OUT} = 500 µA; only one bias output is powered up at a time; VP_MIN = 1b, MIC_BIAS_CTRL = 1b.

Parameters (Note 2)		Min	Typ	Max	Units
MIC1_BIAS and MIC2_BIAS Characteristics					
Output Voltage (Note 27)	MIC_BIAS_CTRL = 0b MIC_BIAS_CTRL = 1b	1.85 2.59	2.00 2.75	2.15 2.89	V V
DC Output Current (I_{OUT}) (Note 28)	Per output Total for both outputs	- -	- -	3.0 5.0	mA mA
Output Resistance (R_{OUT})		-	35	-	Ω
Dropout Voltage (Note 29)		-	-	340	mV
PSRR with 100 mV_{PP} signal AC-coupled to VA supply					
	217 Hz	-	105	-	dB
	1 kHz	-	100	-	dB
	20 kHz	-	90	-	dB
PSRR with 100 mV_{PP} signal AC-coupled to VP supply VP_MIN = 0b, VP = 3.10 V (Note 3)					
	217 Hz	-	90	-	dB
	1 kHz	-	90	-	dB
	20 kHz	-	70	-	dB
PSRR with 1 V_{PP} signal AC-coupled to VP supply VP_MIN = 1b, VP = 3.70 V					
	217 Hz	-	110	-	dB
	1 kHz	-	105	-	dB
	20 kHz	-	90	-	dB

Notes:

27. The output voltage includes attenuation due to the Mic Bias Output Resistance (R_{OUT}).
28. Specifies use limits for the normal operation and MIC2 short conditions.
29. Dropout Voltage indicates the point where an output's voltage starts to vary significantly with reductions to its supply voltage. When the VP supply voltage drops below the programmed MIC2_BIAS output voltage plus the Dropout Voltage, the MIC2_BIAS output voltage will progressively decrease as its supply decreases. Dropout Voltage is measured by reducing the VP supply until MIC2_BIAS drops 10 mV from its initial voltage with the default typical test condition VP voltage (= 3.80 V from table heading above). The difference between the VP supply voltage and the MIC2_BIAS voltage at this point is the dropout voltage. For instance, if the initial MIC2_BIAS output is 2.86 V when VP = 3.80 V and VP = 3.19 V when MIC2_BIAS drops to 2.85 V (-10mV), the Dropout Voltage is 340 mV (3.19 V – 2.85 V).

SERIAL PORT TO STEREO HP OUTPUT CHARACTERISTICS

Test conditions (unless otherwise specified): "Typical Connection Diagram" on page 17 shows CS42L73 connections (including Zobel Networks on outputs); Input test signal is a 24-bit full-scale 997-Hz sine wave with 1 LSB of triangular PDF dither applied; GND = AGND = PGND = CPGND = DGND = 0 V; all voltages are with respect to ground (GND); VA = VCP = 1.80 V; T_A = +25 °C; VCP Mode; Measurement Bandwidth is 20 Hz to 20 kHz; Fs = 48 kHz (Note 10); ASP is used and is in slave mode with Fs_{ext} = 48 kHz; test loading is configured as per Figure 5 on page 28 (R_L and C_L (= $C_{L(Max)}$) as indicated in the table below); Mixer Attenuation and Digital Volume = 0 dB, Digital and Analog Mutes are disabled.

Parameters (Note 2)		Min	Typ	Max	Units
Load R_L = 16 Ω (Analog Gain = -4 dB) (Note 30)					
Dynamic Range					
18 to 24-Bit	A-weighted	87	93	-	dB
	unweighted	84	90	-	dB
16-Bit	A-weighted	85	91	-	dB
	unweighted	82	88	-	dB
Total Harmonic Distortion + Noise (Note 31) (Note 32)	0 dBFS	-	-70	-60	dB
Full-scale Output Voltage (Note 32)		0.73•VA	0.79•VA	0.85•VA	V_{PP}
Output Power (Full-scale, Per Channel) (P_{OUT}) (Note 32)					
2 Channels Driven, THD+N ≤ -60 dB (0.1%)	Analog Vol. = -4 dB, Dig. Vol. = 0 dB	-	16	-	mW
2 Channels Driven, THD+N ≤ -40 dB (1%)	Analog Vol. = -3 dB, Dig. Vol. = 0 dB	-	20	-	mW
2 Channels Driven, THD+N ≤ -20 dB (10%)	Analog Vol. = -1 dB, Dig. Vol. = 0 dB	-	27	-	mW
1 Channel Driven, THD+N ≤ -60 dB (0.1%)	Analog Vol. = -2 dB, Dig. Vol. = 0 dB	-	25	-	mW
1 Channel Driven, THD+N ≤ -40 dB (1%)	Analog Vol. = -1 dB, Dig. Vol. = 0 dB	-	32	-	mW
1 Channel Driven, THD+N ≤ -20 dB (10%)	Analog Vol. = +1 dB, Dig. Vol. = 0 dB	-	44	-	mW
Load R_L = 32 Ω (Analog Gain = -4 dB) (Note 30)					
Dynamic Range					
18 to 24-Bit	A-weighted	88	94	-	dB
	unweighted	85	91	-	dB
16-Bit	A-weighted	86	92	-	dB
	unweighted	83	89	-	dB
Total Harmonic Distortion + Noise (Note 31) (Note 32)	0 dBFS	-	-81	-75	dB
Full-scale Output Voltage (Note 32)		0.74•VA	0.80•VA	0.86•VA	V_{PP}
Output Power (Full-scale, Per Channel) (P_{OUT}) (Note 32)					
2 Channels Driven, THD+N ≤ -75 dB (0.018%)	Analog Vol. = -4 dB, Dig. Vol. = 0 dB	-	8.1	-	mW
2 Channels Driven, THD+N ≤ -60 dB (0.1%)	Analog Vol. = -1 dB, Dig. Vol. = 0 dB	-	16	-	mW
2 Channels Driven, THD+N ≤ -40 dB (1%)	Analog Vol. = 0 dB, Dig. Vol. = -0.5 dB	-	17	-	mW
2 Channels Driven, THD+N ≤ -20 dB (10%)	Analog Vol. = +2 dB, Dig. Vol. = -0.5 dB	-	25	-	mW
1 Channel Driven, THD+N ≤ -75 dB (0.018%)	Analog Vol. = 0 dB, Dig. Vol. = 0 dB	-	20	-	mW
1 Channel Driven, THD+N ≤ -60 dB (0.1%)	Analog Vol. = +1 dB, Dig. Vol. = -0.5 dB	-	23	-	mW
1 Channel Driven, THD+N ≤ -40 dB (1%)	Analog Vol. = +1 dB, Dig. Vol. = 0 dB	-	25	-	mW
1 Channel Driven, THD+N ≤ -20 dB (10%)	Analog Vol. = +3 dB, Dig. Vol. = -0.5 dB	-	35	-	mW
Load R_L = 3 kΩ (Analog Gain = +2 dB) (Note 30)					
Dynamic Range					
18 to 24-Bit	A-weighted	90	96	-	dB
	unweighted	87	93	-	dB
16-Bit	A-weighted	88	94	-	dB
	unweighted	85	91	-	dB
Total Harmonic Distortion + Noise (Note 31) (Note 32)					
18 to 24-Bit	0 dBFS	-	-85	-79	dB
	-20 dBFS	-	-73	-	dB
	-60 dBFS	-	-33	-27	dB
16-Bit	0 dBFS	-	-83	-77	dB
	-20 dBFS	-	-71	-	dB
	-60 dBFS	-	-31	-25	dB

SERIAL PORT TO STEREO HP OUTPUT CHARACTERISTICS (CONTINUED)

Test conditions (unless otherwise specified): "Typical Connection Diagram" on page 17 shows CS42L73 connections (including Zobel Networks on outputs); Input test signal is a 24-bit full-scale 997-Hz sine wave with 1 LSB of triangular PDF dither applied; GND = AGND = PGND = CPGND = DGND = 0 V; all voltages are with respect to ground (GND); VA = VCP = 1.80 V; T_A = +25 °C; VCP Mode; Measurement Bandwidth is 20 Hz to 20 kHz; Fs = 48 kHz (Note 10); ASP is used and is in slave mode with Fs_{ext} = 48 kHz; test loading is configured as per Figure 5 on page 28 (R_L and C_L (= $C_{L(Max)}$) as indicated in the table below); Mixer Attenuation and Digital Volume = 0 dB, Digital and Analog Mutes are disabled.

Parameters (Note 2)		Min	Typ	Max	Units
Full-scale Output Voltage (Note 32)		1.56•VA	1.64•VA	1.73•VA	V_{PP}
Other Characteristics for R_L = 16 Ω, 32 Ω, or 3 kΩ (Note 33)					
Interchannel Isolation (Note 34)		-	90	-	dB
Interchannel Gain Mismatch (Note 34)		-	±0.1	±0.25	dB
Output Offset Voltage (DAC to HPOUTx)	Analog mute enabled	-	±0.1	±1.0	mV
	0 dB analog gain	-	±0.3	±2.0	mV
Gain Drift		-	±100	-	ppm/°C
Load Resistance (R_L) (Note 35)		16	-	-	Ω
Load Capacitance (C_L) (Note 35)		-	-	150	pF
PSRR with 100 mV_{PP} signal AC-coupled to VA supply	217 Hz	-	75	-	dB
- Analog Gain = 0 dB; Input test signal held low (all zeros data)	1 kHz	-	75	-	dB
(Note 8) (Note 36)	20 kHz	-	70	-	dB
PSRR with 100 mV_{PP} signal AC-coupled to VCP supply	217 Hz	-	85	-	dB
- Analog Gain = 0 dB; Input test signal held low (all zeros data)	1 kHz	-	85	-	dB
(Note 8) (Note 36)	20 kHz	-	70	-	dB
Output Impedance	High-Impedance Mode (Note 37)	3.0	3.14	-	kΩ

Notes:

30. Analog Gain setting (refer to "Headphone x Analog Volume Control" on page 103 or "Line Output x Analog Volume Control" on page 104) must be configured as indicated to achieve specified output characteristics.
31. If the VCP supply level is less than the VA supply level, clipping may occur as the audio signal is handed from the VA to the VCP powered circuits in the output amplifier. This clipping would occur as the audio signal approaches full-scale, maximum power output and could prevent achievement of THD+N performance.
32. Full-scale output voltage and power are determined by analog gain settings. Full-scale output voltage values here refer to the maximum voltage difference achievable on the analog output pins, measured between the HPOUTx/LINEOUTx and HPOUT_REF/LINEO_REF pins. Modifying internal gain settings to increase peak-to-peak voltage may cause analog output signal clipping, degrading THD+N performance.
33. Unless otherwise specified, measurement is taken for each load resistance test case with the gain set as indicated for the dynamic range, etc., performance specifications at the given load resistances.
34. Measured between stereo pairs (HPOUTA to HPOUTB or LINEOUTA to LINEOUTB).
35. Figure 5 on page 28 and Figure 6 on page 29 shows headphone and line output test configurations.
36. Valid with the recommended capacitor values on FILT+ and ANA_VQ. Increasing capacitance on FILT+ and ANA_VQ increases the PSRR at low frequencies.
37. High-impedance state enabled as described in Section 4.18.

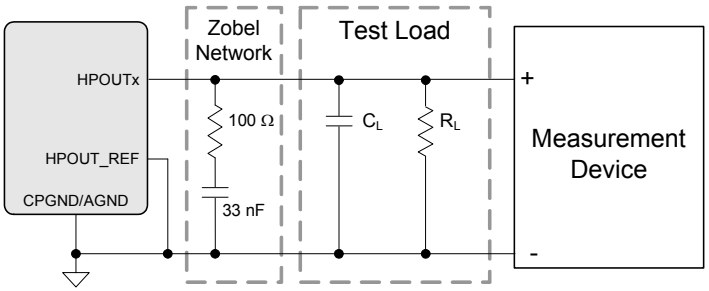


Figure 5. Headphone Output Test Configuration

SERIAL PORT TO STEREO LINE OUTPUT CHARACTERISTICS

Test conditions (unless otherwise specified): Connections to the CS42L73 are shown in the "Typical Connection Diagram" on page 17; Input test signal is a 24-bit full-scale 997 Hz sine wave with 1 LSB of triangular PDF dither applied; GND = AGND = PGND = CPGND = DGND = 0 V; all voltages are with respect to ground (GND); VA = VCP = 1.80 V; T_A = +25 °C; VCP Mode; Measurement Bandwidth is 20 Hz to 20 kHz; Fs = 48 kHz (Note 10); ASP is used and is in slave mode with Fs_{ext} = 48 kHz; test loading is configured as per Figure 6 on page 29 (R_L and C_L as indicated in the table below for $R_{L(Min)}$ and $C_{L(Max)}$); Mixer Attenuation and Digital Volume = 0 dB, Analog Gain = +2 dB; Digital and Analog Mutes are disabled.

Parameters (Note 2)		Min	Typ	Max	Units
(Analog Gain = +2 dB) (Note 30)					
Dynamic Range					
18 to 24-Bit	A-weighted	91	97	-	dB
	unweighted	88	94	-	dB
16-Bit	A-weighted	88	94	-	dB
	unweighted	85	91	-	dB
Total Harmonic Distortion + Noise (Note 31) (Note 32)					
18 to 24-Bit	0 dBFS	-	-86	-80	dB
	-20 dBFS	-	-74	-	dB
	-60 dBFS	-	-34	-28	dB
16-Bit	0 dBFS	-	-84	-78	dB
	-20 dBFS	-	-71	-	dB
	-60 dBFS	-	-31	-25	dB
Full-scale Output Voltage (Note 32) (Note 38)		1.50•VA	1.58•VA	1.66•VA	V_{PP}
Other Characteristics					
Interchannel Isolation (Note 34)		-	90	-	dB
Interchannel Gain Mismatch (Note 34)		-	±0.1	±0.25	dB
Output Offset Voltage (DAC to LINEOUTx)	Analog mute enabled	-	±0.1	±0.5	mV
	0 dB analog gain	-	±0.3	±1.0	mV
Gain Drift		-	±100	-	ppm/°C
Output Resistance (R_{OUT})		-	100	-	Ω
Load Resistance (R_L) (Note 35)		3	-	-	kΩ
Load Capacitances (C_L) (Note 35)		-	-	150	pF
PSRR with 100 mV_{PP} signal AC-coupled to VA supply	217 Hz	-	70	-	dB
- Analog Gain = 0 dB; Input test signal held low (all zeros data)	1 kHz	-	70	-	dB
(Note 8) (Note 36)	20 kHz	-	70	-	dB
PSRR with 100 mV_{PP} signal AC-coupled to VCP supply	217 Hz	-	85	-	dB
- Analog Gain = 0 dB; Input test signal held low (all zeros data)	1 kHz	-	85	-	dB
(Note 8) (Note 36)	20 kHz	-	65	-	dB

Notes:

38. The full-scale output voltage includes attenuation due to the Stereo Line Output Resistance (R_{OUT}).

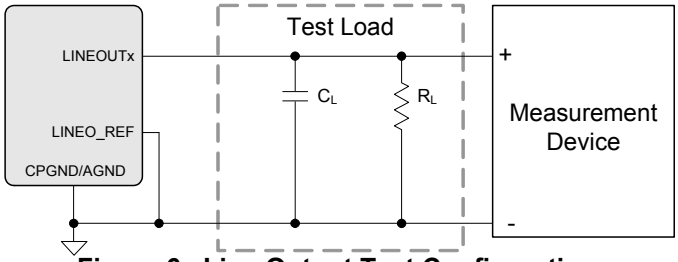


Figure 6. Line Output Test Configuration

SERIAL PORT TO MONO EAR SPEAKER OUTPUT CHARACTERISTICS

Test conditions (unless otherwise specified): Connections to the CS42L73 are shown in the "Typical Connection Diagram" on page 17; Input test signal is a 24-bit full-scale 997 Hz sine wave with 1 LSB of triangular PDF dither applied; GND = AGND = PGND = CPGND = DGND = 0 V; all voltages are with respect to ground (GND); VA = 1.80 V; T_A = +25 °C; Measurement Bandwidth is 20 Hz to 20 kHz; Fs = 48 kHz (Note 10); ASP is used and is in slave mode with Fs_{ext} = 48 kHz; test loading is configured as per Figure 7 on page 30 (R_L, C_{L1}, and C_{L2} as indicated in the table below for $R_{L(Min)}$, $C_{L1(Max)}$, and $C_{L2(Max)}$); Mixer Attenuation = 0 dB, Digital Volume = -2.5 dB, Digital Mute is disabled.

Parameters (Note 2)		Min	Typ	Max	Units
Dynamic Range					
16 to 24-Bit	A-weighted	82	88	-	dB
	unweighted	79	85	-	dB
Total Harmonic Distortion + Noise (Note 39), 16 to 24-Bit					
	0 dBFS, P_{OUT} = 45 mW	-	-70	-65	dB
Full-scale Output Voltage (Note 39)	(Diff. EAROUT ±, see Note 40)	1.24•VA	1.34•VA	1.44•VA	V_{PP}
Output Power (Full-scale) (P_{OUT}) (Note 39)					
THD+N ≤ -65 dB (0.056%)	Dig. Vol. = -2.5 dB	-	45	-	mW
THD+N ≤ -60 dB (0.1%)	Dig. Vol. = -2.0 dB	-	51	-	mW
THD+N ≤ -40 dB (1%)	Dig. Vol. = -1.5 dB	-	56	-	mW
THD+N ≤ -20 dB (10%)	Dig. Vol. = -0.5 dB	-	66	-	mW
Other Characteristics					
Output Offset Voltage	(DC offset of diff. EAROUT ±, see Note 40)	-	±2.5	±4.0	mV
Gain Drift		-	±100	-	ppm/°C
Load Resistance (R_L) (Note 41)		16	-	-	Ω
Load Capacitances	C_{L1} across outputs	-	-	150	pF
(Note 41)	C_{L2} from each output to ground	-	-	50	pF
PSRR with 100 mV_{PP} signal AC-coupled to VA supply	217 Hz	-	70	-	dB
- Input test signal held low (all zeros data)	1 kHz	-	70	-	dB
(Note 36)	20 kHz	-	70	-	dB

Notes:

39. Modifying internal gain settings to achieve a higher peak-to-peak voltage may result in clipping the analog output signal, degrading the THD+N performance.
40. Differential peak-to-peak voltage is measured from the extremes (peaks) of the waveform that represents the difference between the positive and negative signals of the differential pair [i.e. the voltage between the maximum and minimum of V_{Diff} (= $V_+ - V_-$)].
41. Refer to Figure 7 on page 30 and Figure 8 on page 32 to observe Ear-Speaker, Speakerphone, and Speakerphone Line-Output test configurations.

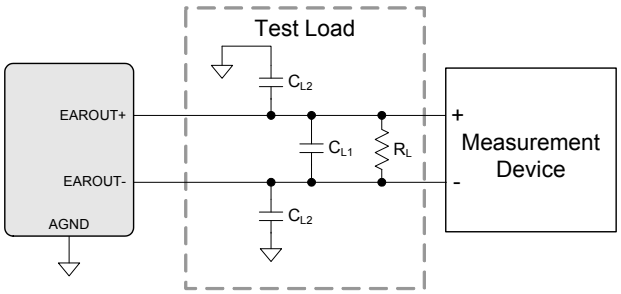


Figure 7. Ear Speaker Output Test Configuration

SERIAL PORT-TO-MONO SPEAKERPHONE OUTPUT CHARACTERISTICS

Test conditions (unless otherwise specified): "Typical Connection Diagram" on page 17 shows CS42L73 connections; Input test signal is a 24-bit full-scale 997 Hz sine wave with 1 LSB of triangular PDF dither applied; GND = AGND = PGND = CPGND = DGND = 0 V; all voltages are with respect to ground (GND); VA = 1.80 V, VP = 3.70 V; T_A = +25 °C; Measurement Bandwidth is 20 Hz to 20 kHz; Fs = 48 kHz (Note 10); ASP is used and is in slave mode with Fs_{ext} = 48 kHz; test loading is configured as per Figure 8 on page 32 (R_L, C_{L1} (= $C_{L1(Max)}$) and C_{L2} (= $C_{L2(Max)}$) as indicated in the table below); Mixer Attenuation = 0 dB, Digital Volume = -5.5 dB, Digital Mute is disabled, SPK_LITE_LOAD = 0b and 1b for R_L = 8 Ω and 50 kΩ, respectively.

Parameters (Note 2)		Min	Typ	Max	Units
Load R_L = 8 Ω (SPK_LITE_LOAD = 0)					
Dynamic Range					
16 to 24-Bit	A-weighted	80	86	-	dB
	unweighted	77	83	-	dB
Total Harmonic Distortion + Noise (Note 39) (Note 42), 16- to 24-Bit					
0 dBFS, P_{OUT} = 0.48 W		-	-65	-62	dB
		-	0.056	0.079	%
Full-scale Output Voltage (Note 39)	(Diff. SPKOUT ±, see Note 40)	2.85•VA	3.09•VA	3.33•VA	V_{PP}
Output Power (P_{OUT}) (Continuous Average) (Note 39)					
THD+N ≤ -62 dB (0.079%)	VP = 3.70 V, Dig. Vol. = -5.5 dB	-	0.48	-	W
THD+N ≤ -40 dB (1.0%)	VP = 3.70 V, Dig. Vol. = -4.5 dB	-	0.59	-	W
	VP = 4.20 V, Dig. Vol. = -3.5 dB	-	0.76	-	W
	VP = 5.00 V, Dig. Vol. = -2.0 dB	-	1.06	-	W
THD+N ≤ -20 dB (10%)	VP = 3.70 V, Dig. Vol. = -2.5 dB	-	0.75	-	W
	VP = 4.20 V, Dig. Vol. = -1.5 dB	-	0.95	-	W
	VP = 5.00 V, Dig. Vol. = 0.0 dB	-	1.36	-	W
Load R_L = 50 kΩ (SPK_LITE_LOAD = 1)					
Dynamic Range					
16 to 24-Bit	A-weighted	78	84	-	dB
	unweighted	75	81	-	dB
Total Harmonic Distortion + Noise (Note 39) (Note 42)	16 to 24-Bit 0 dBFS	-	-65	-60	dB
Full-scale Output Voltage (Note 39)	(Diff. SPKOUT ±, see Note 40)	2.99•VA	3.23•VA	3.47•VA	V_{PP}
Other Characteristics for R_L = 8 Ω or 50 kΩ (Note 33)					
Output Offset Voltage	(DC offset of diff. SPKOUT ±, see Note 40)	-	±5.0	±10.0	mV
Gain Drift		-	±100	-	ppm/°C
Load Resistance (R_L) (Note 41)	SPK_LITE_LOAD = 0b	6.5	8	100	Ω
	SPK_LITE_LOAD = 1b	3.0	50	-	kΩ
Load Capacitances	C_{L1} across outputs/load	-	-	150	pF
(Note 41) (Note 43)	C_{L2} from each output to ground	-	-	50	pF
PSRR with 100 mV_{PP} signal AC-coupled to VA supply	217 Hz	-	70	-	dB
- Input test signal held low (all zeros data)	1 kHz	-	70	-	dB
(Note 36)	20 kHz	-	70	-	dB
PSRR with 100 mV_{PP} signal AC-coupled to VP supply	217 Hz	-	70	-	dB
- Input test signal held low (all zeros data)	1 kHz	-	80	-	dB
(Note 44)	20 kHz	-	60	-	dB

Notes:

42. When the VP supply level is low and the VA supply level is high, clipping may occur as the audio signal is handed off from the VA to the VP powered circuits within the output amplifier. This clipping would occur as the audio signal approached full-scale, maximum power output and could result in the specified THD+N performance not being achieved.
43. The maximum speakerphone capacitance across the load is specified as C_{L1}. If more load capacitance is desired, contact Cirrus Logic for alternatives using additional external circuitry.
44. Valid with the recommended capacitor values on FILT+ and SPK_VQ. Increasing the capacitance on FILT+ and SPK_VQ will increase the PSRR at low frequencies.

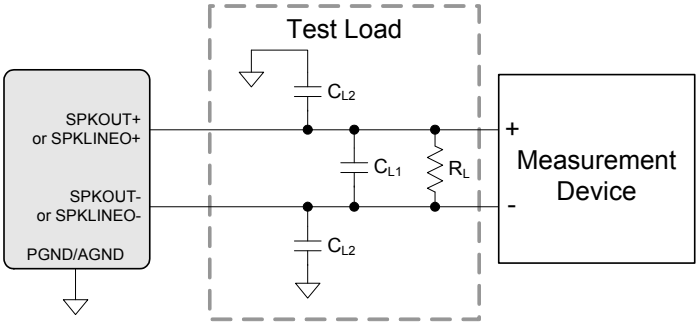


Figure 8. Speakerphone and Speakerphone Line Output Test Configuration

SERIAL PORT TO MONO SPEAKERPHONE LINE OUTPUT CHARACTERISTICS

Test conditions (unless otherwise specified): Connections to the CS42L73 are shown in the "Typical Connection Diagram" on page 17; Input test signal is a 24-bit full-scale 997-Hz sine wave with 1 LSB of triangular PDF dither applied; GND = AGND = PGND = CPGND = DGND = 0 V; all voltages are with respect to ground (GND); VA = 1.80 V, VP = 3.70 V; T_A = +25 °C; Measurement Bandwidth is 20 Hz to 20 kHz; Fs = 48 kHz (Note 10); ASP is used and is in slave mode with Fs_{ext} = 48 kHz; test loading is configured as per Figure 8 on page 32 (R_L, C_{L1}, and C_{L2} as indicated in the table below for $R_{L(Typ)}$, $C_{L1(Max)}$, and $C_{L2(Max)}$); Mixer Attenuation = 0 dB, Digital Mute is disabled.

Parameters (Note 2)		Min	Typ	Max	Units
Digital Volume = -5.5 dB					
Dynamic Range					
16 to 24-Bit	A-weighted	78	84	-	dB
	unweighted	75	81	-	dB
Total Harmonic Distortion + Noise (Note 39) (Note 42), 16 to 24-Bit	0 dBFS	-	-65	-60	dB
Full-scale Output Voltage (Note 39) (Note 45)	(Diff. SPKLINEO±, see Note 40)	2.97•VA	3.21•VA	3.45•VA	V_{PP}
Other Characteristics					
Output Offset Voltage	(DC offset of diff. SPKLINEO±, see Note 40)	-	±5.0	±10.0	mV
Gain Drift		-	±100	-	ppm/°C
Output Resistance (R_{OUT}) (Note 46)		-	100	-	Ω
Load Resistance (R_L) (Note 41)		3	50	-	kΩ
Load Capacitances	C_{L1} across outputs/load	-	-	150	pF
(Note 41) (Note 43)	C_{L2} from each output to ground	-	-	50	pF
PSRR with 100 mV_{PP} signal AC-coupled to VA supply	217 Hz	-	70	-	dB
–Input test signal held low (all zeros data)	1 kHz	-	70	-	dB
(Note 36)	20 kHz	-	70	-	dB
PSRR with 100 mV_{PP} signal AC-coupled to VP supply	217 Hz	-	70	-	dB
–Input test signal held low (all zeros data)	1 kHz	-	80	-	dB
(Note 44)	20 kHz	-	75	-	dB

Notes:

45. The full-scale output voltage includes attenuation due to the Speakerphone Line Output Resistance (R_{OUT}).
46. The specified output resistance is present on each of the SPKLINEO pins.

STEREO/MONO DAC INTERPOLATION AND ON-CHIP DIGITAL/ANALOG FILTER CHARACTERISTICS

Test Conditions (unless otherwise specified): Fs = 48 kHz (Note 10).

Parameters (Note 2), (Note 47)		Min	Typ	Max	Units
DAC Low-Pass Filter Characteristics—EAR/HP/LINE					
Frequency Response ($0.453x10^{-3}$ x Fs to 0.453 x Fs)		-0.02	-	0.10	dB
Passband	to -0.05 dB corner	-	0.48	-	Fs
	to -3.0 dB corner	-	0.50	-	Fs
DAC Low-Pass Filter Characteristics—SPK/SPKLINEOUT					
Frequency Response ($0.453x10^{-3}$ x Fs to 0.453 x Fs)		-0.29	-	0.02	dB
Passband	to -0.05 dB corner	-	0.36	-	Fs
	to -3.0 dB corner	-	0.50	-	Fs
DAC Low-Pass Filter Characteristics—All					
Stopband @ -60 dB (Note 21)		0.55	-	-	Fs
Total DAC Group Delay		-	3.8/Fs	-	s
Post-DAC High-Pass Filter Characteristics					
Passband	to -3.0 dB corner	-	$5.2x10^{-6}$	-	Fs
	to -0.05 dB corner	-	$4.4x10^{-5}$	-	Fs
Passband Ripple		-	-	0.01	dB
Phase Deviation @ 20 Hz		-	0.61	-	Deg
Filter Settling Time (input signal goes to 95% of its final value)		-	10^5/Fs	-	s

Notes:

47. Responses are clock dependent and scale with Fs. Note that the response plots (Figures 56 to 59 on pages 131 and 132) have been normalized to Fs and can be denormalized by multiplying the X-axis scale by Fs.

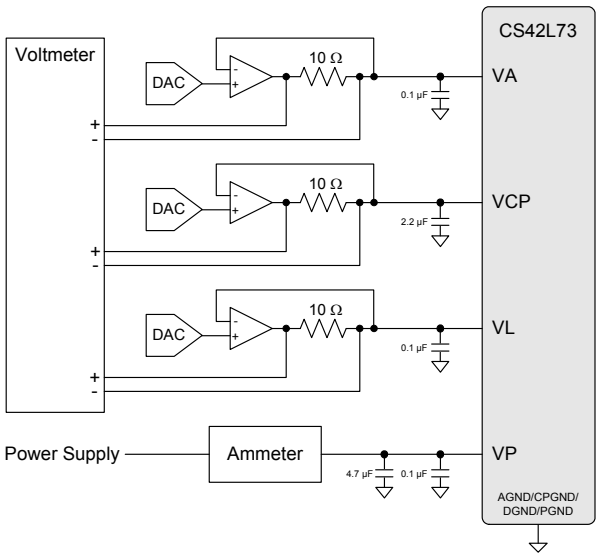
POWER CONSUMPTION

Test conditions (unless otherwise specified): Connections to the CS42L73 are shown in the "Typical Connection Diagram" on page 17; GND = AGND = PGND = CPGND = DGND = 0 V; all voltages are with respect to ground (GND); VA = VCP = VL = 1.80 V, VP = 3.70 V; T_A = +25 °C; $\overline{\text{RESET}}$ pin inactive; f_{MCLK1} = 6.144 MHz, MCLK2 is held low; Internal MCLK enabled and derived from MCLK1 input (/1) (thus, Fs = 48 kHz); f_{DMIC_SCLK} = 1.536 MHz (/4); silence on analog inputs, microphones are not attached; SCL inactive, SDA held high; XSP and ASP are in I²S slave mode, VSP is in PCM mode, Fs_{ext}(XSP and ASP) = 48 kHz, Fs_{ext}(VSP) = 8 kHz; XSP, ASP, and VSP clocks are held low unless port is in use; XSP, ASP, VSP and DMIC data inputs are held low (all zeros data), XSP, ASP, and VSP data output lines are not driven by other devices in system; no load on analog outputs except for HP Zobel; PDN, PDN_ADCx, PDN_BIASx, PDN_DMICx, PDN_XSPSDOUT, PDN_XSPSDIN, PDN_ASPSDOUT, PDN_ASPSDIN, PDN_VSP, PDN_HP, PDN_EAR, PDN_LO, PDN_SPK, PDN_SPKLO, PDN_THMS = 1b; all other register controls as per defaults; Figure 9 on page 34 describes the current-measuring method.

	Case/Configuration	Class H Mode	Typical Current (µA)				Total Power (µW)
			i_{VA}	i_{VP}	i_{VCP}	i_{VL}	
1	**Reset** $\overline{\text{RESET}}$ and MCLK1 held low, PDN* = x, MCLKDIS = x	-	1	2	1	4	18
2	**Standby** (PDN = 1b) MCLK1 held low, MCLK from MCLK1, MCLKDIS = 1b	-	1	4	1	7	31
3	**Stereo Play to HP**: ASP to HP PDN_ASPSDIN, PDN_HP = 0b	VCP/3	611	4	468	1029	3809
		VCP/2	611	4	556	1029	3968
		VCP	611	4	913	1029	4610
4	**Stereo Play to HP - 32 Ω load** (Note 48): ASP to HP PDN_ASPSDIN, PDN_HP = 0b	VCP/3	611	4	1598	1338	6399
		VCP/2	611	4	2238	1338	7551
		VCP	611	4	4255	1338	11182
5	**Handset Voice Call (Digital Mic)**: DMICA to VSP, Mono VSP to EAR PDN_DMICA, PDN_VSP, PDN_EAR = 0b	-	1233	4	1	1232	4454
6	**Handset Voice Call (Analog Mic)**: MIC1 to VSP, Mono VSP to EAR PDN_BIAS1, PDN_ADCA, PDN_VSP, PDN_EAR = 0b	-	2377	107	1	1160	6764
7	**Headset Voice Call**: MIC2 to VSP, Mono VSP to Stereo HP PDN_BIAS2, PDN_ADCB, PDN_VSP, PDN_HP = 0b	VCP/3	1794	112	475	1280	6803
8	**Headset Voice Call - 32 Ω load** (Note 48): MIC2 to VSP, Mono VSP to Stereo HP PDN_BIAS2, PDN_ADCB, PDN_VSP, PDN_HP = 0b	VCP/3	1794	112	1598	1329	8912

Notes:

48. In accordance with the JEITA CP-2905B standard, 0.1 mW per channel is delivered to headphone loads via a 1 kHz sine wave. The popular 32 Ω headphone loading is used.



Note: The current draw on each CS42L73 power supply pin, except VP, is derived from the measured voltage drop across a 10-Ω series resistor between the associated supply source and each voltage supply pin. Given the larger currents that are possible on the VP supply, an ammeter is used on that rail.

Figure 9. Power Consumption Test Configuration

DIGITAL INTERFACE SPECIFICATIONS AND CHARACTERISTICS

Test Conditions (unless otherwise specified): Connections to the CS42L73 are shown in the "Typical Connection Diagram" on page 17; GND = AGND = PGND = CPGND = DGND = 0 V; all voltages are with respect to ground (GND); VA = VL = 1.80 V, VP = 3.70 V; T_A = +25 °C.

Parameters (Note 2)		Symbol	Min	Max	Units
Input Leakage Current (Note 49) (Note 50)	Inputs with pull-up/downs	I_{in}	-	±4000	nA
	Inputs without pull-up/downs		-	±800	nA
Input Capacitance (Note 49)		-	-	10	pF
SDA Pull-up Resistance (Note 51)		R_P	500	-	Ω
$\overline{INT}$ Pull-up Resistance (Note 51)		R_{P_I}	2	-	kΩ
Logic I/Os					
High-Level Output Voltage (I_{OH} = -100 µA)		V_{OH}	VL – 0.2	-	V
Low-Level Output Voltage	All outputs (I_{OL} = 100 µA)	V_{OL}	-	0.2	V
	SDA and $\overline{INT}$ (I_{OL} as per $R_{P(min)}$ and $R_{P\_I(min)}$) (Note 51)		-	0.2•VL	V
High-Level Input Voltage		V_{IH}	0.70•VL	-	V
Low-Level Input Voltage		V_{IL}	-	0.30•VL	V
$\overline{MIC2\_SDET}$ Input					
High-Level Input Voltage		V_{IH-SD}	0.55	-	V
Low-Level Input Voltage		V_{IL-SD}	-	0.35	V

Notes:

49. Specification is per pin.
50. Specification includes current through internal pull up/down resistors, where applicable (as defined in section "Digital Pin/Ball I/O Configurations" on page 16).
51. The minimum values of the pull-up resistors R_P and R_{P_I} (as shown in the "Typical Connection Diagram" on page 17 and specified in "Digital Interface Specifications and Characteristics" on page 35) are determined using the maximum level of VL, the minimum sink current strength of their respective output, and the maximum low-level output voltage (V_{OL} in "Digital Interface Specifications and Characteristics" on page 35). The maximum values of R_P and R_{P_I} may be determined by the how fast their associated signals must transition (e.g., the lower the value of R_P, the faster the I²C bus will be able to operate for a given bus load capacitance). Refer to "Switching Specifications—Control Port" on page 40 and to the I²C bus specification (see section "References" on page 137) for more details.

SWITCHING SPECIFICATIONS—POWER, RESET, AND MASTER CLOCKS

Test conditions (unless otherwise specified): Connections to the CS42L73 are shown in the "Typical Connection Diagram" on page 17; GND = AGND = PGND = CPGND = DGND = 0 V; all voltages are with respect to ground (GND); VA = VCP = VL = 1.80 V, VP = 3.70 V; T_A = +25 °C; Inputs: Logic 0 = GND, Logic 1 = VL.

Parameters (Note 2)	Symbol	Min	Max	Units
Power Supplies (Note 52)				
Power Supply Ramp Up/Down	$t_{pwr\text{-}rud}$	-	100	ms
Power Supply Ramp Skew	$t_{pwr\text{-}rs}$	-	1	s
Reset (Note 52)				
$\overline{\text{RESET}}$ low (Logic 0) Pulse Width	t_{rlpw}	1	-	ms
$\overline{\text{RESET}}$ Hold Time After Power Supplies Ramp Up	$t_{rh(PWR\text{-}RH)}$	1	-	ms
$\overline{\text{RESET}}$ Setup Time Before Power Supplies Ramp Down	$t_{rs(RL\text{-}PWR)}$	1	-	ms
Master Clocks				
MCLK1 or MCLK2 Frequency (Note 53)	-	-	38.5	MHz
MCLK1 or MCLK2 Duty Cycle	-	45	55	%

Notes:

52. Refer to Figure 10 on page 36.
53. Maximum frequency for highest supported nominal rate is indicated. The supported nominal MCLK1/MCLK2 rates and their associated configurations are found in section "Internal Master Clock Generation" on page 42. Likewise, the supported nominal serial port sample rates are found in section "Serial Port Sample Rates and Master Mode Settings" on page 53.

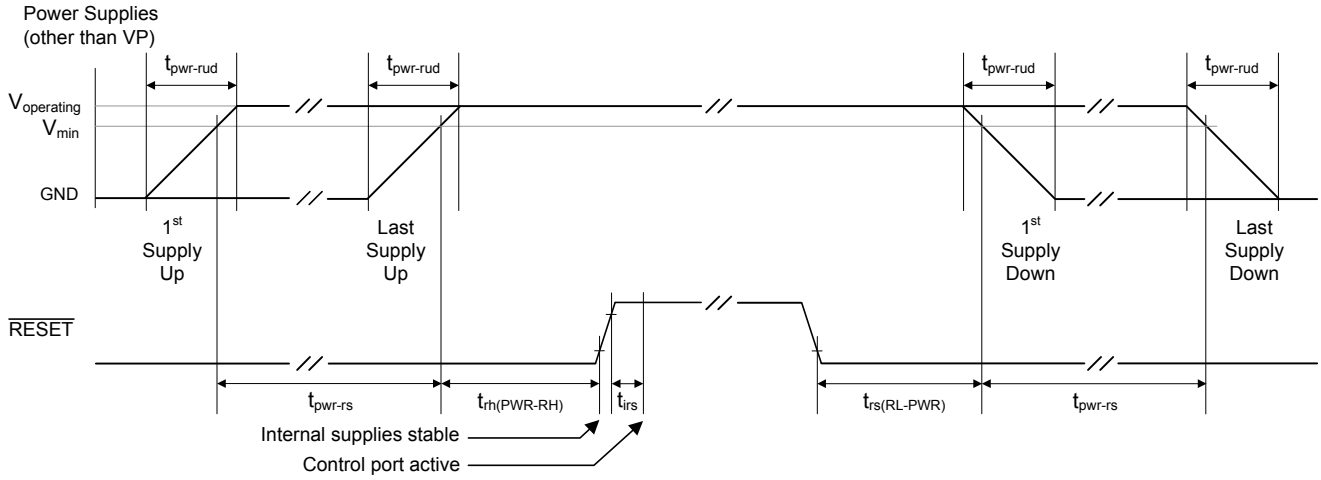


Figure 10. Power and Reset Sequencing

SWITCHING SPECIFICATIONS—DIGITAL MIC INTERFACE

Test conditions: Inputs: Logic 0 = GND = DGND = 0 V, Logic 1 = VL; T_A = +25 °C; C_{LOAD} = 30 pF.

Parameters (Note 2)	Symbol	Min	Max	Units
Output Clock (DMIC_CLK) Frequency	$1/t_P$	-	(Note 54)	kHz
DMIC_CLK Duty Cycle	-	45	55	%
DMIC_CLK Rise Time (10% to 90% of VL)	t_r	-	22	ns
DMIC_CLK Fall Time (90% to 10% of VL)	t_f	-	10	ns
DMIC_SD Setup Time Before DMIC_CLK Rising Edge (Note 55)	$t_{s(SD-CLKR)}$	10	-	ns
DMIC_SD Hold Time After DMIC_CLK Rising Edge (Note 55)	$t_{h(CLKR-SD)}$	0	-	ns
DMIC_SD Setup Time Before DMIC_CLK Falling Edge (Note 55)	$t_{s(SD-CLKF)}$	10	-	ns
DMIC_SD Hold Time After DMIC_CLK Falling Edge (Note 55)	$t_{h(CLKF-SD)}$	600	-	ps

Notes:

54. The output clock frequency will follow the master clock (MCLK) rate divided down as per the tables in sections "Digital Microphone (DMIC) Interface" on page 60. Any deviation of the Master Clock source from the nominal supported rates will be directly imparted to the output clock rate by the same factor (e.g., +100 ppm offset in the frequency of MCLK1/MCLK2 will become a +100 ppm offset in DMIC_CLK).
55. Data is valid at the high-level input voltage (V_{IH}) and the low-level input voltage (V_{IL}), which are specified in "Digital Interface Specifications and Characteristics" on page 35.

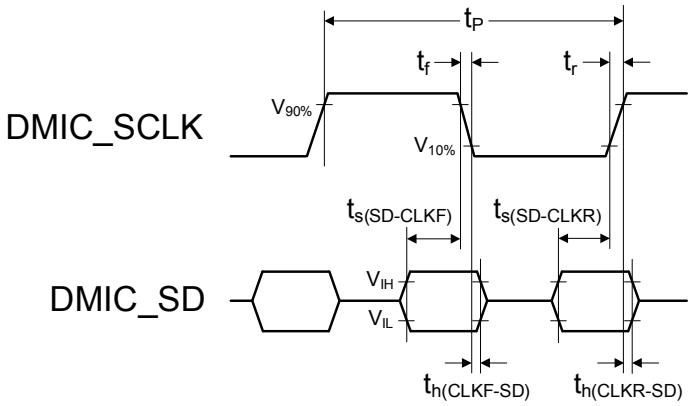


Figure 11. Digital Mic Interface Timing

SWITCHING SPECIFICATIONS—SERIAL PORTS—I²S FORMAT

Test conditions: Inputs: Logic 0 = GND = DGND = 0 V, Logic 1 = VL; T_A = +25 °C; x = X, A, or V; xSP_LRCK, xSP_SCLK, xSP_SDOUT; C_{LOAD} = 15 pF.

Parameters (Note 2)		Symbol	Min	Max	Units
Slave Mode					
Input sample Rate (xSP_LRCK)	(Note 24) (Note 53)	Fs_{ext-s}	-	50	kHz
xSP_LRCK Duty Cycle		-	45	55	%
xSP_SCLK Frequency	(Note 10)	$1/t_{Ps}$	-	68•Fs	Hz
xSP_SCLK Duty Cycle		-	45	55	%
xSP_LRCK Setup Time Before xSP_SCLK Rising Edge		$t_{ss(LK-SK)}$	40	-	ns
xSP_LRCK Hold Time After xSP_SCLK Rising Edge		$t_{hs(SK-LK)}$	20	-	ns
xSP_SDOUT Setup Time Before xSP_SCLK Rising Edge		$t_{ss(SDO-SK)}$	20	-	ns
xSP_SDOUT Hold Time After xSP_SCLK Rising Edge		$t_{hs(SK-SDO)}$	30	-	ns
xSP_SDIN Setup Time Before xSP_SCLK Rising Edge		$t_{ss(SDI-SK)}$	20	-	ns
xSP_SDIN Hold Time After xSP_SCLK Rising Edge		$t_{hs(SK-SDI)}$	20	-	ns
Master Mode					
Output Sample Rate (xSP_LRCK)	(Note 24)	Fs_{ext-m}	-	(Note 56)	kHz
xSP_LRCK Duty Cycle		-	45	55	%
xSP_SCLK Frequency	“SCLK ≠ MCLK” Mode	$1/t_{Pm}$	-	68•Fs_{ext-m}	Hz
	“SCLK = MCLK” Mode		-	MCLK	Hz
	“SCLK = Pre-MCLK” Mode (Note 57)		-	12.1	MHz
xSP_SCLK Duty Cycle		-	45	55	%
xSP_LRCK Setup Time Before xSP_SCLK Rising Edge		$t_{sm(LK-SK)}$	35	-	ns
xSP_LRCK Hold Time After xSP_SCLK Rising Edge		$t_{hm(SK-LK)}$	20	-	ns
xSP_SDOUT Setup Time Before xSP_SCLK Rising Edge		$t_{sm(SDO-SK)}$	20	-	ns
xSP_SDOUT Hold Time After xSP_SCLK Rising Edge		$t_{hm(SK-SDO)}$	30	-	ns
xSP_SDIN Setup Time Before xSP_SCLK Rising Edge		$t_{sm(SDI-SK)}$	20	-	ns
xSP_SDIN Hold Time After xSP_SCLK Rising Edge		$t_{hm(SK-SDI)}$	20	-	ns

Notes:

56. In Master Mode, the output sample rate follows the Master Clock source (MCLK1 or MCLK2) rate divided down per “Internal Master Clock Generation” on page 42 and “Serial Port Sample Rates and Master Mode Settings” on page 53. Any Master Clock source deviation from the nominal supported rates is directly imparted to the output sample rate by the same factor (e.g., +100 ppm offset in the MCLK1/MCLK2 frequency becomes a +100 ppm xSP_LRCK offset).
57. Maximum frequency for highest supported nominal rate is indicated. The supported nominal rates are described in section “SCLK = MCLK Modes” on page 53.

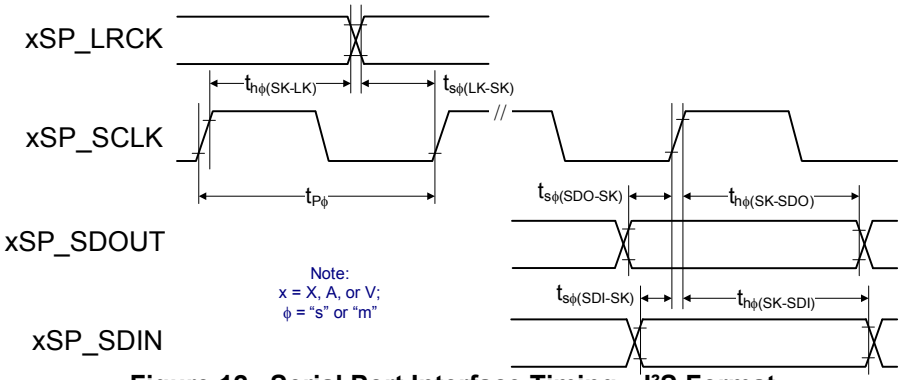


Figure 12. Serial Port Interface Timing—I²S Format

SWITCHING SPECIFICATIONS—SERIAL PORTS—PCM FORMAT

Test condition: Inputs: Logic 0 = GND = DGND = 0 V, Logic 1 = VL; T_A = +25 °C; x = X or V; xSP_LRCK, xSP_SCLK, xSP_SDOUT; C_{LOAD} = 15 pF.

Parameters (Note 2)	Symbol	Min	Max	Units
Slave Mode				
Input Sample Rate (xSP_LRCK) (Note 24) (Note 53)	Fs_{ext-s}	-	50	kHz
xSP_LRCK Duty Cycle	-	45	55	%
xSP_SCLK Frequency (Note 10)	$1/t_{Ps}$	-	68•Fs	Hz
xSP_SCLK Duty Cycle	-	45	55	%
xSP_LRCK Setup Time Before xSP_SCLK Falling Edge	$t_{ss(LK-SK)}$	40	-	ns
xSP_LRCK Hold Time After xSP_SCLK Falling Edge	$t_{hs(SK-LK)}$	20	-	ns
xSP_SDOUT Setup Time Before xSP_SCLK Falling Edge	$t_{ss(SDO-SK)}$	20	-	ns
xSP_SDOUT Hold Time After xSP_SCLK Falling Edge	$t_{hs(SK-SDO)}$	30	-	ns
xSP_SDIN Setup Time Before xSP_SCLK Falling Edge	$t_{ss(SDI-SK)}$	20	-	ns
xSP_SDIN Hold Time After xSP_SCLK Falling Edge	$t_{hs(SK-SDI)}$	20	-	ns

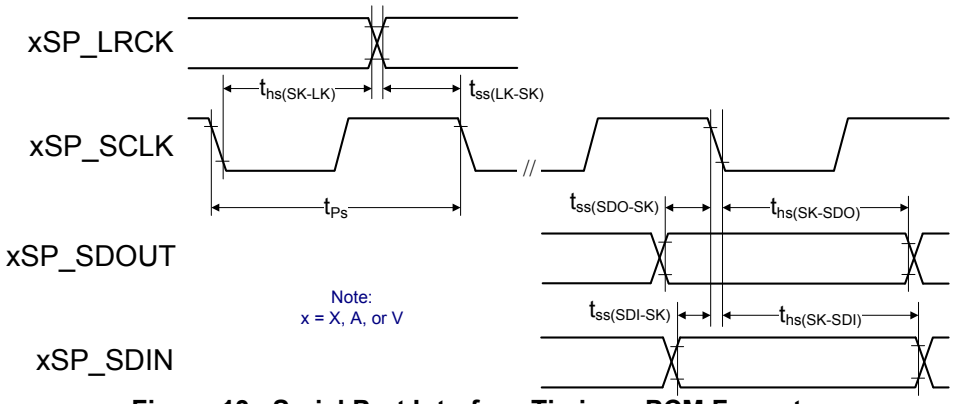


Figure 13. Serial Port Interface Timing—PCM Format

SWITCHING SPECIFICATIONS—CONTROL PORT

Test conditions: Inputs: Logic 0 = GND = DGND = 0 V, Logic 1 = VL; T_A = +25 °C; SDA load capacitance equal to maximum value of C_b specified below (Note 58); minimum SDA pull-up resistance ($R_{P(min)}$) (Note 51).

Parameters (Note 2)	Symbol	Min	Max	Unit
$\overline{\text{RESET}}$ Rising Edge to Start (Note 52)	t_{irs}	500	-	ns
SCL Clock Frequency	f_{scl}	-	550	kHz
Start Condition Hold Time (prior to first clock pulse)	t_{hdst}	0.6	-	µs
Clock Low time	t_{low}	1.3	-	µs
Clock High Time	t_{high}	0.6	-	µs
Setup Time for Repeated Start Condition	t_{sust}	0.6	-	µs
SDA Input Hold Time from SCL Falling (Note 59)	t_{hddi}	0	0.9	µs
SDA Output Hold Time from SCL Falling	t_{hddo}	0.2	0.9	µs
SDA Setup Time to SCL Rising	t_{sud}	100	-	ns
Rise Time of SCL and SDA	t_r	-	300	ns
Fall Time SCL and SDA	t_f	-	300	ns
Setup Time for Stop Condition	t_{susp}	0.6	-	µs
Bus Free Time Between Transmissions	t_{buf}	1.3	-	µs
SDA Bus Load Capacitance (Note 51)	C_b	-	400	pF

Notes:

58. All specifications are valid for the signals at the pins of the CS42L73 with the specified load capacitance.
59. Data must be held for sufficient time to bridge the transition time, t_f, of SCL.

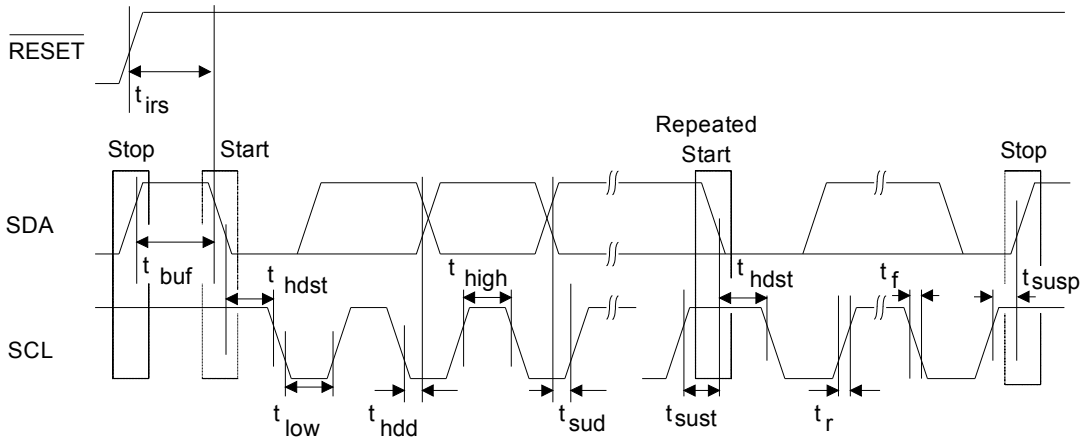


Figure 14. I²C Control Port Timing

4. APPLICATIONS

4.1 Overview

4.1.1 Basic Architecture

The CS42L73 is a highly integrated, ultralow power, 24-bit audio CODEC comprising a stereo ADC and two stereo DAC converters. The ADC is fed by pseudodifferential inputs. The DACs feed two stereo pseudodifferential output amplifiers and three mono (or one mono and one stereo, depending on configuration) full-differential amplifiers. The ADC and DAC are designed using multibit delta-sigma techniques. Both converters operate at a low oversampling ratio, maximizing power savings while maintaining high performance.

The serial data interface ports of the CS42L73 may operate at standard audio sample rates as either the master or slave of timing. The timing of the core of the CS42L73 is flexibly sourced, without the need of a PLL, by clocks with typical audio clock rates (N x 5.6448 or 6.1440 MHz; N = 1 or 2), USB rates (6, 12, or 24 MHz), or common cell phone reference rates (N x 13.0 or 19.2 MHz; N = 1 or 2).

Designed with a very low voltage digital core and low voltage Class H amplifiers (powered from an integrated LDO regulator and a step-down/inverting charge pump, respectively), the CS42L73 provides significant reduction in overall power consumption.

4.1.2 Line and Microphone Inputs

The analog input portion of the CODEC allows selection from stereo line-level or mic sources. The selected source is fed into a microphone preamplifier (when applicable) and then a PGA, before entering the stereo ADC.

When used, the pseudodifferential analog input configuration provides noise-rejection for single-ended analog inputs to the CS42L73.

4.1.3 Line and Headphone Outputs (Class H, Ground-Centered Amplifiers)

The analog output portion of the CODEC includes separate pseudodifferential headphone and line out Class H amplifiers. An on-chip step-down/inverting charge pump creates a positive and negative voltage equal to the input, one-half the input, or one-third the input supply for the amplifiers, allowing an adaptable, full-scale output swing centered around ground. The inverting architecture eliminates the need for large DC-blocking capacitors and allows the amplifier to deliver more power to headphone loads at lower supply voltages. The step-down architecture allows the amplifier's power supply to adapt to the required output signal. This adaptive power supply scheme converts traditional Class AB amplifiers into more power-efficient Class H amplifiers.

4.1.4 Digital Mixer

The digital mixer facilitates the mixing and routing of the ADC and serial port audio data to the device analog and Serial port outputs. All routes from inputs to outputs are supported.

All paths have selectable attenuation before being mixed to allow relative volume control and to avoid clipping.

4.1.5 Power Management

Several control registers and bits provide independent power down control of the analog and digital sections of the CS42L73, allowing operation in select applications with minimal power consumption.

4.2 Internal Master Clock Generation

Table 1 outlines the supported internal Master Clock (MCLK) nominal frequencies and how they are derived from the supported frequencies of the external MCLK sources (MCLK1 and MCLK2).

Table 1. Internal Master Clock Generation

MCLK1/MCLK2 Rate (MHz)	Required Divide Ratio	MCLK Rate (MHz)	Internal Fs (kHz)	Settings for MCLKDIV[2:0] (Note 1)
5.6448	1	5.6448	44.100	000
11.2896	2			010
6.0000	1	6.0000	46.875	000
12.0000	2			010
24.0000	4			100
6.1440	1	6.1440	48.000	000
12.2880	2			010
13.0000	2	6.5000	50.781	010
26.0000	4			100
19.2000	3	6.4000	50.000	011
38.4000	6			101

Notes:

1. The MCLKDIV[2:0] register control is described in section "Master Clock Divide Ratio" on page 87.
2. To save power, MCLK may be disabled using the MCLKDIS register control (refer to section "Master Clock Disable" on page 87).
3. Refer to section "SCLK = MCLK Modes" on page 53 for a description of the frequency limitations on MCLK1 and MCLK2 when using the "SCLK = MCLK" or "SCLK = Pre-MCLK" modes.

4.3 Thermal Overload Notification

The CS42L73 can be configured to notify the system processor when its die temperature is too high. The processor can use this notification prevent possible damage to the CS42L73 and other devices in the system. When notified, the processor should react by powering down CS42L73 (and/or other devices in the system) partially or entirely, depending on the extent to which the CS42L73's power dissipation is the cause of its excessive die temperature. Note, the Speakerphone output, when used, accounts for the vast majority of the power dissipation from the CS42L73.

To use thermal overload notification:

1. Enable the thermal sense circuitry by programming PDN_THMS.
2. Configure the threshold temperature (via control bits THMOVLD_THLD[1:0]), over which the Thermal Overload Interrupt Status bit will be set.
3. If an interrupt is desired when the Thermal Overload Detect (THMOVLD) bit toggles from 0b to 1b, set the M_THMOVLD control to 1b. If polling is desired, set it to 0b.
4. Monitor (read after interrupt or poll) THMOVLD and react accordingly.

Referenced Control	Register Location
PDN_THMS	"Power Down Thermal Sense" on page 84,
THMOVLD_THLD[1:0]	"Thermal Overload Threshold Settings" on page 84
M_THMOVLD	"Interrupt Mask Register 1 (Address 5Eh)" on page 122
THMOVLD	"Thermal Overload Detect" on page 122

4.4 Pseudodifferential Outputs

The CS42L73 provides access to the headphone and line output amplifiers' reference inputs via the HPOUT_REF and LINEO_REF pins. These pins may be connected to either the ground at the device, or the ground return pin of each amplifier's corresponding output connector. By routing HPOUT_REF and LINEO_REF to the ground at the device, the respective amplifier's common mode is dictated by the ground local to the device. An equivalent circuit is shown in Figure 15 where the ground-noise voltages developed local to the device and the jack are modeled as voltage sources $V_{N-LOCAL}$ and V_{N-JACK}, respectively. $V_{N-LOCAL}$ is transferred to the output of the amplifier. $V_{N-LOCAL}$ - V_{N-JACK} is then presented across the load (V_{N-LOAD}). For PCB designs in which the headphone or line output signals traverse long distances to an output connector, $V_{N-LOCAL}$ and V_{N-JACK} can be different. As such, V_{N-LOAD} may be significant and can compromise dynamic range performance.

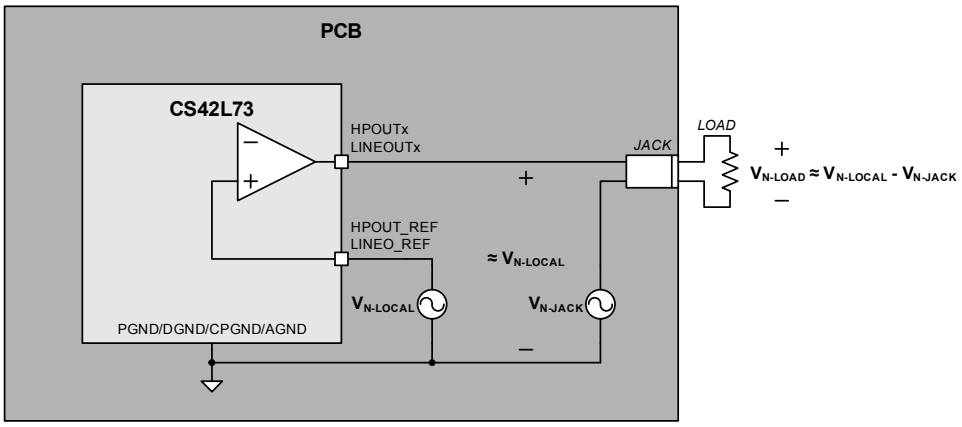


Figure 15. Single-Ended Output Configuration

By routing HPOUT_REF and LINEO_REF to the corresponding output connector as shown in Figure 16, however, the amplifier's common mode is dictated by the ground local to the jack. This connection is useful in systems for which, as described above, the ground noise local to the device differs from the ground noise at the jack. As this noise voltage couples to HPOUT_REF and LINEO_REF, it is also transferred through the amplifier to its output. This behavior allows the ground noise at the jack to be seen as common mode at the load, and as a result, V_{N-LOAD} is minimized.

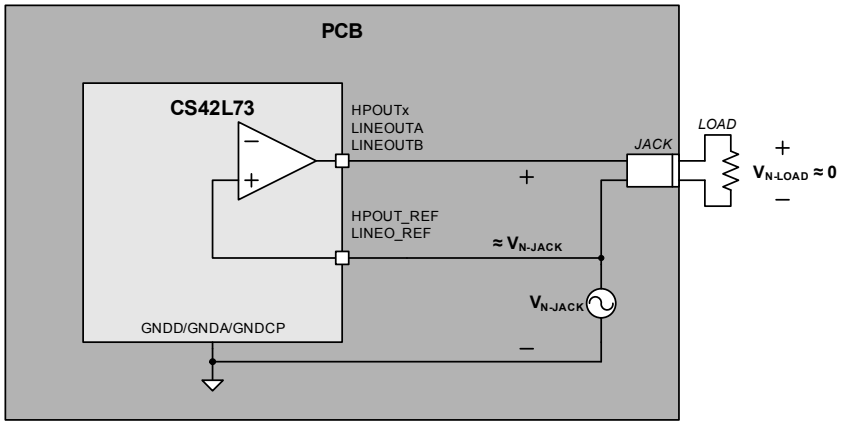


Figure 16. Pseudodifferential Output Configuration

Minimize any impedance from the HPOUT_REF and LINEO_REF pins to the corresponding load ground (typically the connector ground). Impedance in this path affects analog output attenuation of the output amplifier, which affects output offset and step deviation. Table 2 shows the effects of impedance on the reference pin with regard to output attenuation:

External Impedance (Ω)	Maximum Attenuation Possible @ -76 dB Setting (dB)
0	-76.0
1	-74.5
10	-72.3
50	-64.8
100	-60.0

Table 2. Example of Impedance in Reference Path

4.5 Class H Amplifier

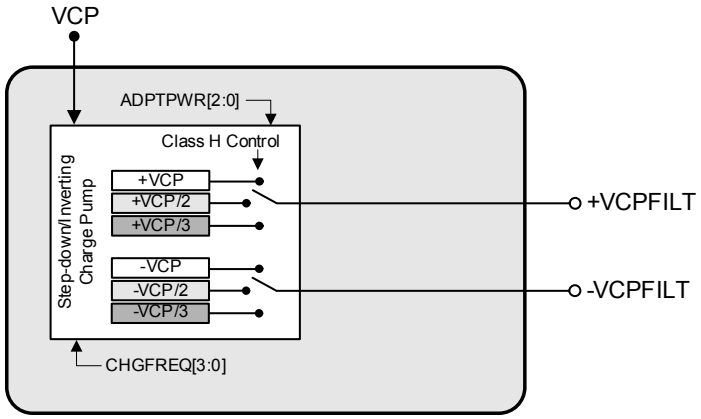


Figure 17. Class H Operation

Referenced Control	Register Location
Analog Output ADPTPWR[2:0]	"Adaptive Power Adjustment" on page 85

The CS42L73 headphone and line output amplifiers use a patent-pending Cirrus Logic Tri-Modal Class H technology. This technology maximizes operating efficiency of the typical Class AB amplifier while maintaining high performance. In a Class H amplifier design, the rail voltages supplied to the amplifier vary with the needs of the music passage that is being amplified. This prevents unnecessarily wasting energy during low power passages of program material or when the program material is played back at a low volume level.

The central component of the Tri-Modal Class H technology found in the CS42L73 is the internal charge pump, which creates the rail voltages for the headphone and line amplifiers of the device. The charge pump receives its input voltage from the voltage present on the CS42L73 VCP pin. From this input voltage, the charge pump creates three sets of the differential rail voltages that are supplied to the amplifier output stages: ±VCP, ±VCP/2, and ±VCP/3.

4.5.1 Power Control Options

The method by which the CS42L73 selects the set of rail voltages supplied to the amplifier output stages depends on the settings of the ADPTPWR[2:0] bits found in "Charge Pump Frequency and Class H Configuration (Address 09h)" section on page 85. There are five possible settings for these bits: Mode 000, 001, 010, 011, and 111.

Referenced Control	Register Location
ADPTPWR[2:0]	"Adaptive Power Adjustment" on page 85

4.5.1.1 Standard Class AB Operation (Mode 001, 010, and 011)

When the ADPTPWR is set to 001, 010, or 011, the rail voltages supplied to the amplifiers will be held to ±VCP, ±VCP/2, or ±VCP/3, respectively. For these settings, the rail voltages supplied to the output stages are held constant, regardless of the signal level, internal volume settings, or the settings of the advisory volume registers. In these settings, the amplifiers in the CS42L73 simply operate in a traditional Class AB configuration.

Note: In the 010 or 011 setting, clipping can occur if the input signal level exceeds the headroom of the output amplifier.

4.5.1.2 Adapt-to-Volume Settings (Mode 000)

If the Adaptive Power bits are set to 000, the CS42L73 determines which set of rail voltages to send to the amplifiers based upon the gain and attenuation levels of all active internal processing blocks. To adjust for digital (DSP) input volume settings, it also takes into account the settings of the advisory volume registers. The combined effect of all volume settings is shown in Figure 18.

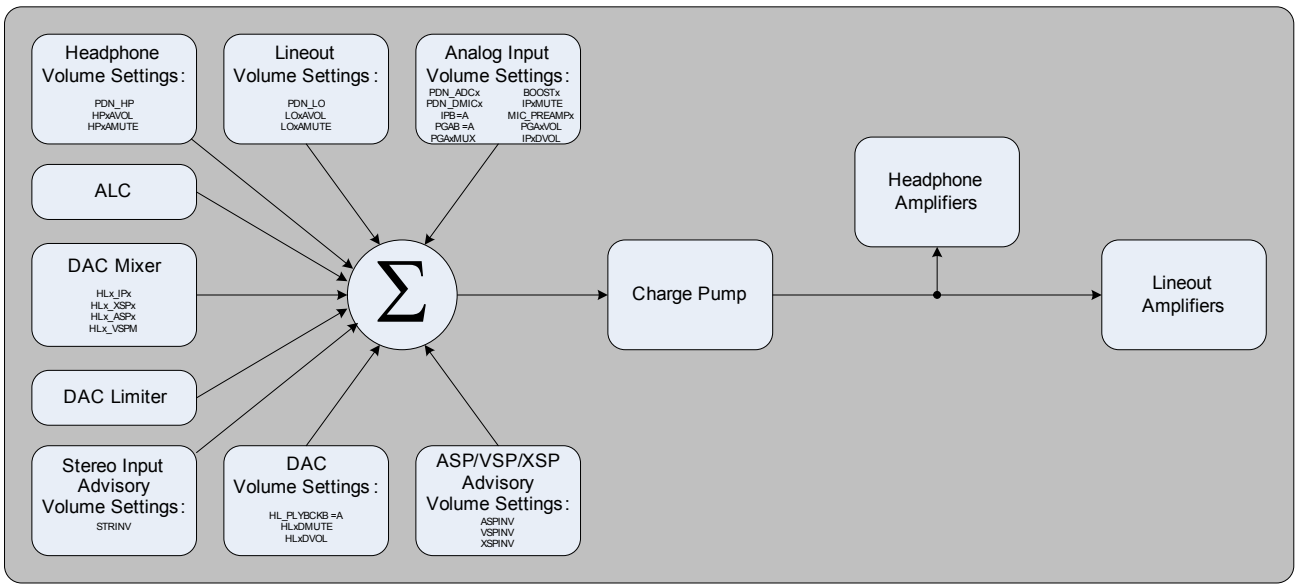


Figure 18. Class H Control - Adapt-to-Volume Mode

If the total gain and attenuation set in the volume control registers would cause the amplifiers to clip the signal with the lowest voltage setting (±VCP/3), the control logic instructs the charge pump to provide the next higher set of the rail voltages (±VCP/2, then ±VCP) to the amplifiers until the signal is no longer clipped or the charge pump is in its highest mode (±VCP).

Note that the A and B channels of each respective volume control must both cross the threshold to trigger a change to a lower VCP mode. If either channel crossed the threshold in an upward direction, the charge pump will switch to a higher VCP mode. The control logic also monitors various functions (listed in the following table) that may affect the total gain and attenuation of the signal applied to the amplifiers.

Referenced Control	Register Location
PDN_HP	“Power Down Headphone” on page 85
HPxAVOL	“Headphone x Analog Volume Control” on page 103
HPxAMUTE	“Headphone x Analog Mute” on page 103
PDN_LO	“Power Down Line Output” on page 85
LOxAVOL	“Line Output x Analog Volume Control” on page 104
LOxAMUTE	“Line Output x Analog Mute” on page 104
HL_PLYBCKB=A	“Headphone/Line Output (HL) Playback Channels B=A” on page 100
HLxDMUTE	“Headphone/Line Output (HL) x Digital Mute” on page 101
HLxDVOL	“Headphone/Line Output (HL) x Digital Volume Control” on page 101
PDN_ADCx	“Power Down ADC x” on page 82
PDN_DMICx	“Power Down Digital Mic x” on page 82
IPB=A	“Input Path Channel B=A” on page 94
PGAB=A	“PREAMP and PGA Channel B=A” on page 94
PGAxMUX	“PGA x Input Select” on page 97
BOOSTx	“Boost x” on page 97
IPxMUTE	“Input Path x Digital Mute” on page 97
MIC_PREAMPx	“Mic PREAMP x Volume” on page 98
PGAxVOL	“PGAx Volume” on page 98
IPxDVOL	“Input Path x Digital Volume Control” on page 99
HLx_IPx	“Stereo Mixer Input Attenuation” on page 119
HLx_XSPx	“Stereo Mixer Input Attenuation” on page 119
HLx_ASPx	“Stereo Mixer Input Attenuation” on page 119
HLx_VSPM	“Stereo Mixer Input Attenuation” on page 119
STRINV	“Stereo Input Path Advisory Volume” on page 105
ASPINV	“ASP Input Advisory Volume” on page 106
VSPINV	“VSP Input Advisory Volume” on page 106
XSPINV	“XSP Input Advisory Volume” on page 105

4.5.1.3 Adapt-to-Output Signal (Mode 111)

If the Adaptive Power bits are set to 111, the CS42L73 determines which of the three sets of rail voltages to send to the amplifiers based solely on the level of the signal being sent to the amplifiers. If that signal would cause the amplifiers to clip when operating on the lower set of rail voltages, the control logic instructs the charge pump to provide the next higher set of rail voltages to the amplifiers. If that signal would not cause the amplifiers to clip when operating on the lower set of rail voltages, control logic instructs the charge pump to provide the lower set of rail voltages to the amplifiers. This mode eliminates the need to advise the CS42L73 of volume settings external to the device.

4.5.2 Power Supply Transitions

Charge pump transitions from the lower to the higher set of rail voltages occur on the next FLYN/FLYP clock cycle. Despite the system’s fast response time, the capacitive elements on the VCP_FILT pins prevent the rail voltages from changing instantaneously. Instead, the voltages ramp up from the lower to the higher supply, based on the time constant created by the output impedance of the charge pump and the capacitor on the VCP_FILT pin (the transition time is approximately 20 µs). The behavior of ±VCP/2 to and from ±VCP is shown in Figure 19. During this charging transition, a high dv/dt transient on the inputs may briefly clip the outputs before the rail voltages charge to the full higher supply level. This transitory clipping has been found to be inaudible in listening tests.

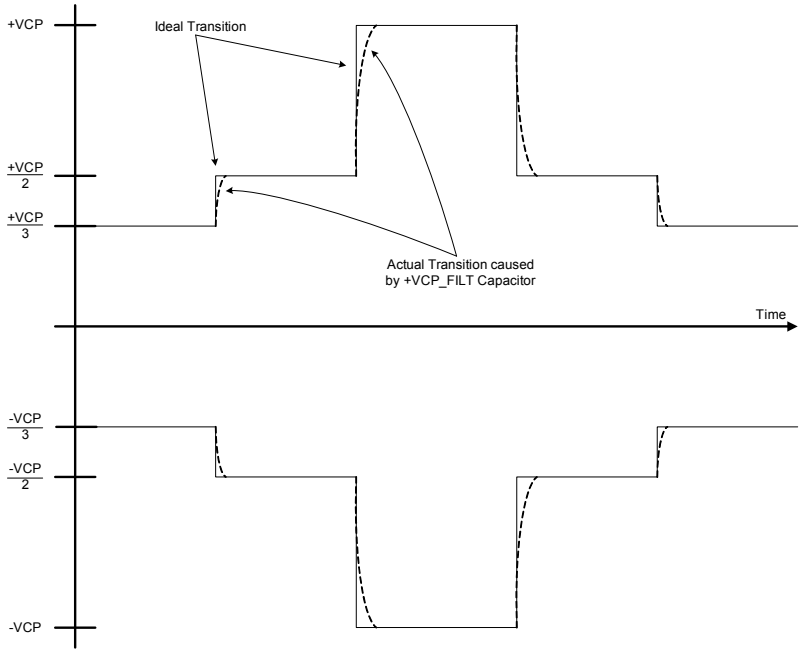


Figure 19. VCP_FILT Transitions

When the charge pump transitions from the higher set of rail voltages to the lower set, there is a 2-second delay before the charge pump supplies the lower rail voltages to the amplifiers. This hysteresis ensures that the charge pump does not toggle between the two rail voltages as signals approach the clip threshold. It also prevents clipping in the instance of repetitive high-level transients in the input signal. The timing diagram of ±VCP/2 to/from ±VCP for this transitional behavior is detailed in Figure 20.

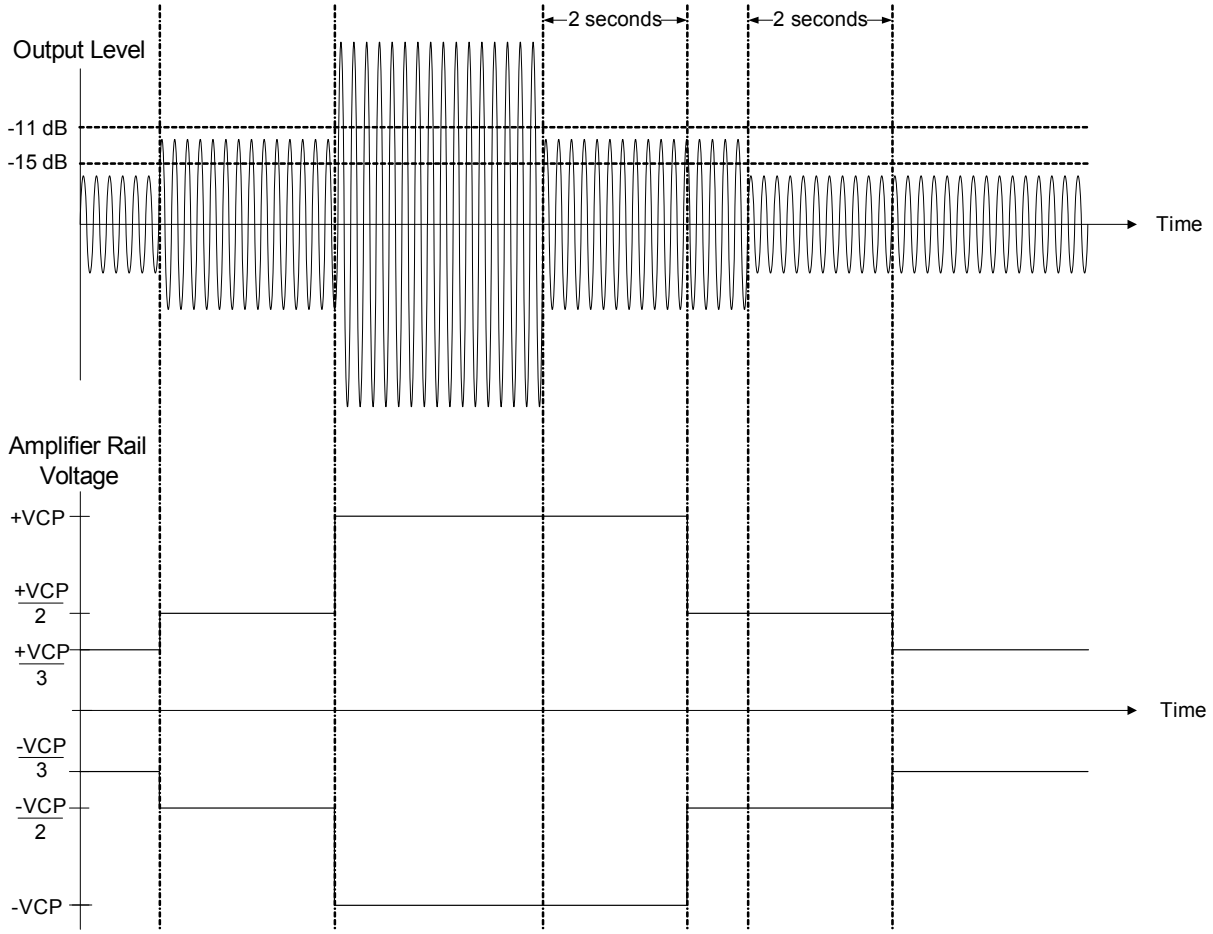


Figure 20. VCP_FILT Hysteresis

4.5.3 Efficiency

As discussed in previous sections, the HPOUTx and LINEOUTx amplifiers may operate from one of three pairs of rail voltages based on the amplitude of the output signal or the relevant volume settings in the signal path. Figure 21 shows total power drawn by the device vs. power delivered to two headphone loads when the rails are held constant at each of the three available settings, or when the Class H controller is set to Adapt-to-Volume mode.

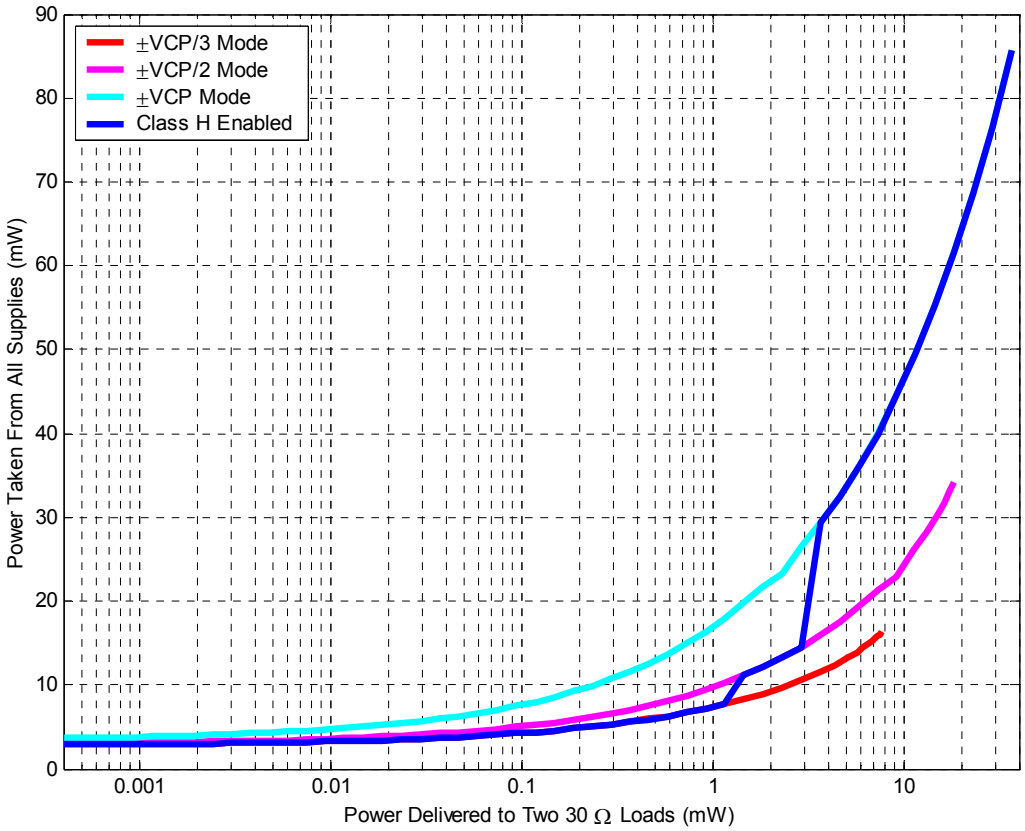


Figure 21. Input Power vs. Output Power

Note: Test Conditions: VCP = VL = VA = 1.8 V, VP = 5 V; MCLK = 6 MHz, LRCK = 44.118 kHz; full-scale input signal applied through HPOUTA and HPOUTB.

If rail voltages are set to ±VCP Mode, the output amplifiers operate in their least-efficient mode for low-level signals. When rail voltages are held at ±VCP/2 or ±VCP/3, the amplifiers operate in a more efficient mode, but clip when amplifying a full-scale signal.

The blue trace in Figure 21 shows the benefit of the Tri-Modal Class H design. At lower output levels, the output of the amplifiers is represented by the ±VCP/3 or ±VCP/2 curves, depending on the signal level. At higher output levels, the output is represented by the ±VCP curve. The duration in which the amplifiers operate within any of the three rail pairs (±VCP/3, ±VCP/2, or ±VCP) depends on both the content and the output level of the program material being amplified. The highest efficiency results from maintaining an output level that is close to, but does not exceed, the clip threshold of a particular supply curve.

4.6 DAC Limiter

When enabled, the limiter monitors the digital input signal before the DAC modulators, detects when levels exceed the maximum threshold settings and lowers the volume at a programmable attack rate below the

maximum threshold. When the input signal level falls below the maximum threshold, the AOUT volume returns to its original level set in the HP/LO Volume Control register at a programmable release rate. Attack and release rates are affected by the DAC soft ramp settings and sample rate, Fs. Limiter soft ramp dependency may be independently enabled/disabled using the LIMSRDIS.

Note that the limiter maintains the output signal between the CUSHSPK, CUSHHL, CUSHESL and LMAXSPK, LMAXHL, LMAXESL thresholds. As the digital input signal level changes, the level-controlled output may not always be the same, but always falls within the thresholds.

Recommended settings: Best limiting performance may be realized with the fastest attack and slowest release setting with soft ramp enabled in the control registers. The CUSHx bits allow the user to set a threshold slightly below the maximum threshold for hysteresis control—this cushions the sound as the limiter attacks and releases.

Referenced Control	Register Location
Limiter Rates	"Limiter Attack Rate HL" on page 107, "Limiter Release Rate HL" on page 107 "Limiter Attack Rate Speakerphone [A]" on page 108, "Limiter Release Rate Speakerphone [A]" on page 109 "Limiter Attack Rate ESL [B]" on page 111, "Limiter Release Rate ESL [B]" on page 111
Limiter Thresholds	"Limiter Cushion Threshold HL" on page 108, "Limiter Maximum Threshold HL" on page 108 "Limiter Cushion Threshold Speakerphone [A]" on page 110, "Limiter Maximum Threshold Speakerphone [A]" on page 110 "Limiter Cushion Threshold ESL [B]" on page 112, "Limiter Maximum Threshold ESL [B]" on page 112
LIMSRDIS	"Limiter Soft-Ramp Disable" on page 100
Volume Controls	"Headphone/Line Output (HL) x Digital Volume Control" on page 101 "Speakerphone Out [A] Digital Volume Control" on page 102 "Ear Speaker/Speakerphone Line Output (ESL) [B] Digital Volume Control" on page 102 "Speakerphone Out [A] Digital Volume Control" on page 102

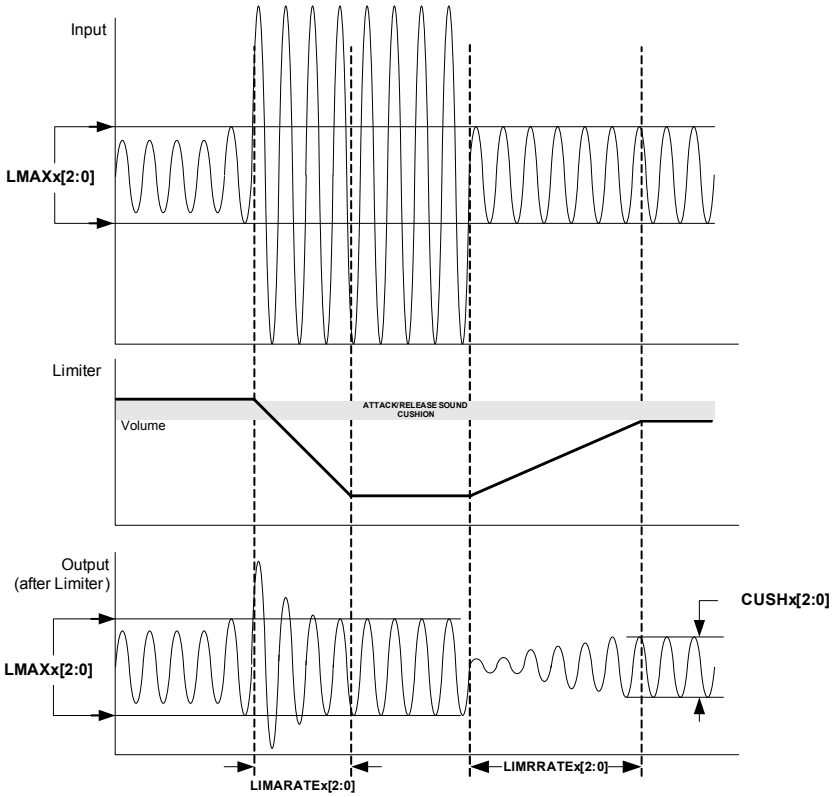


Figure 22. Peak Detect & Limiter

4.7 Analog Output Current Limiter

The CS42L73 features built-in current-limit protection for both the headphone and line output amplifiers. The approximate current through VCP during the short circuit conditions shown in Figure 23 and Figure 24 is described in Table 3.

Note: 100 Ω is always required in series with the line-output amplifiers. These amplifiers must never be shorted directly to ground.

While the values in Table 3 show that the device is protected from permanent damage during a short circuit, they do not represent maximum specification. See "DC Electrical Characteristics" on page 21.

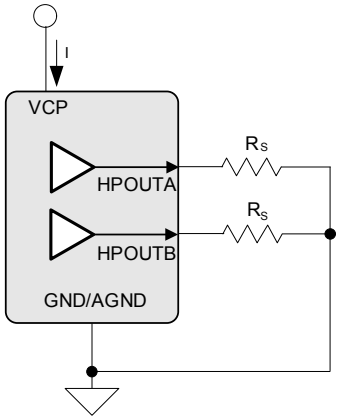


Figure 23. HP Short Circuit Setup

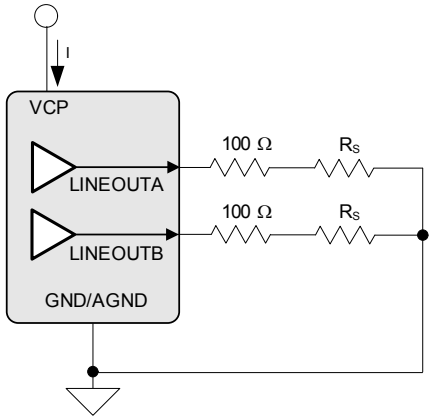


Figure 24. Line Short Circuit Setup

Amplifier	R_S (Ω)	Maximum Current (mA)
HPOUTx	3.3	120
	0	120
LINEOUTx	0	120

Table 3. Current through VCP with Varying Short Circuits

4.8 Serial Ports

The three independent, highly configurable, serial ports XSP, ASP, and VSP communicate audio and voice data to and from other devices in the system, such as application processors, Bluetooth transceivers, and cell-phone modems.

4.8.1 Power Management

The XSP and ASP have separate power-down controls (PDN_XSP_SDOUT, PDN_XSP_SDIN, PDN_ASP_SDOUT, and PDN_ASP_SDIN) for their input and output data paths. Separating power state controls minimizes power consumption if only monoplex communication is required (e.g., music playback).

The VSP, being targeted for duplex voice communication, has a single power-down control, PDN_VSP.

4.8.2 I/O

Each serial port interface consists to four signals (x = X, A, or V):

- xSP_SCLK Serial data shift clock
- xSP_LRCK Left/right clock
  - Identifies the start of each serialized data word
  - Identifies where each channel (left or right) is located within the data word when I²S format (refer to section "I²S Format" on page 55) is used

- Toggles at external sample rate (Fs_{ext})

- xSP_SDIN Serial data input
- xSP_SDOUT Serial data output

4.8.3 High-impedance Mode

The serial ports may be placed on a clock/data bus that allows multiple masters, without the need for external buffers. The 3ST_XSP, 3ST_ASP, and 3ST_VSP bits place the internal buffers for the respective serial port interface signals in a high-impedance state, allowing another device to transmit clocks and data without bus contention. When the CS42L73 serial port is a timing slave, its xSP_SCLK and xSP_LRCK I/Os are always inputs and are thus unaffected by the 3ST_xSP control.

Figure 25 and Figure 26 show the busing of the serial port interface for both the master and slave timing CS42L73 serial port use cases.

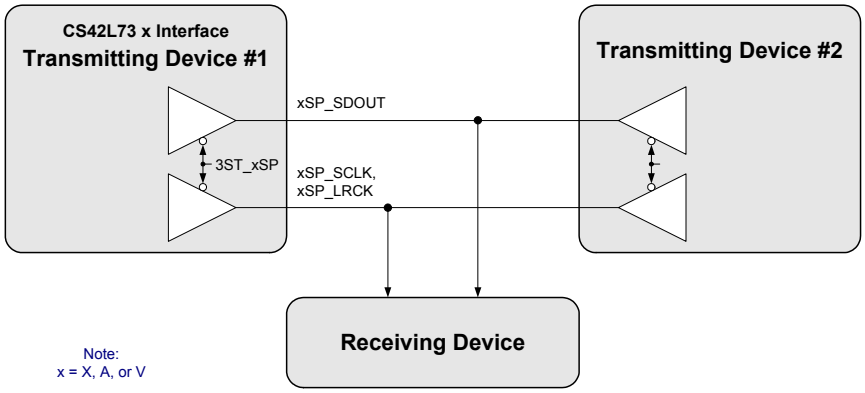


Figure 25. Serial Port Busing when Mastering Timing

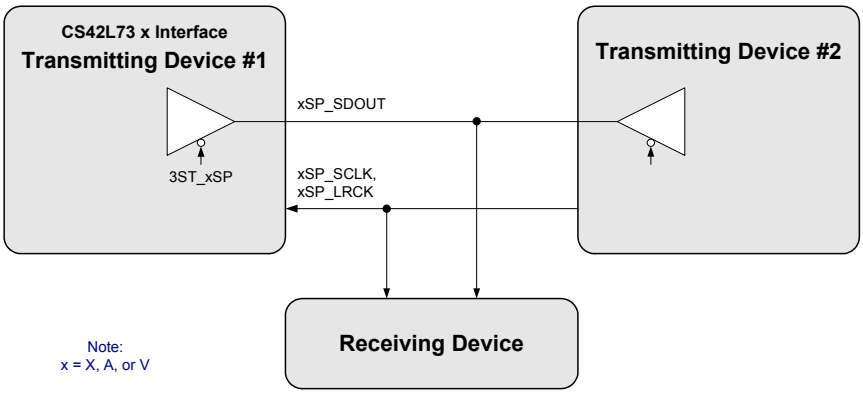


Figure 26. Serial Port Busing When Slave Timed

4.8.4 Master and Slave Timing

The serial ports can independently operate as either the master of timing or a slave to another device's timing. When mastering, xSP_SCLK and xSP_LRCK are outputs, when slaved, they are inputs. Master/Slave mode is configured by the X_M/S, A_M/S, and V_M/S bits. Note, master mode is not supported when the PCM interface format is selected (refer to section "PCM Format" on page 55).

In master mode, the xSP_SCLK and xSP_LRCK clock outputs are derived from either the internal MCLK (MCLK) or (for a subset of SCLK = MCLK modes, refer to section "SCLK = MCLK Modes" on page 53) directly from its source, MCLK1 or MCLK2.

When in slave mode, the supported interface sample rates (Fs_{ext}) are as is shown for MCLK = 6.000 MHz in the table "Serial Port Rates and Master Mode Settings" on page 53.

The master mode supported rates for each supported MCLK are listed in the aforementioned table. The table also documents how to program the X_MMCC[5:0], A_MMCC[5:0], and V_MMCC[5:0] registers to derive the desired master mode Fs_{ext} and how much the derived Fs_{ext} rate deviates from the desired rate.

4.8.4.1 SCLK = MCLK Modes

The frequency of the Serial Clock (xSP_SCLK) is programmable in master mode using the register controls X_SCLK = MCLK[1:0], A_SCLK=MCLK[1:0], and V_SCLK = MCLK[1:0]. It can be either automatically derived to approximate 64 cycles per xSP_LRCK period, be equal to MCLK, or it can be set to be equal to Pre-MCLK, the predivided version of MCLK (MCLK1 or MCLK2 as per register control MCLK-SEL).

When in MCLK mode, all the MCLK1/MCLK2 rates and corresponding supported MCLK rates shown in Table 1. “Internal Master Clock Generation” on page 42 are supported. When in Pre-MCLK mode, the supported MCLK1/MCLK2 rates are as is shown in the following table.

Table 4. Supported MCLK1/MCLK2 Rates for Pre-MCLK Mode

MCLK1/MCLK2 = xSP_SCLK Rate (MHz)
5.6448
11.2896
6.0000
12.0000
6.1440

4.8.5 Serial Port Sample Rates and Master Mode Settings

Table 5 illustrates the supported serial port nominal audio sample rates (Fs_{ext}) and the settings required to generate them when in master mode. See the notes on the following page.

Table 5. Serial Port Rates and Master Mode Settings

MCLK Rate (MHz) (Note 1)	Standard Audio Sample Rate (kHz)	Actual Master Mode xSP_LRCK Rate (Fs_{ext}) (kHz)	Deviation (%)	Settings for x_MMCC[5:0] (Note 2)
5.6448	11.0250	11.0250	0.00	11 0000
	22.0500	22.0500	0.00	10 0000
	44.1000	44.1000	0.00	01 0000
6.0000 (Note 3)	8.0000	8.0000	0.00	11 1001
	11.0250	11.0294	0.04	11 0011
	12.0000	12.0000	0.00	11 0001
	16.0000	16.0000	0.00	10 1001
	22.0500	22.0588	0.04	10 0011
	24.0000	24.0000	0.00	10 0001
	32.0000	32.0000/31.9149	0.00/–0.27	01 1001
	44.1000	44.1176	0.04	01 0011
	48.0000	48.0000	0.00	01 0001
6.1440	8.0000	8.0000	0.00	11 1000
	12.0000	12.0000	0.00	11 0000
	16.0000	16.0000	0.00	10 1000
	24.0000	24.0000	0.00	10 0000
	32.0000	32.0000	0.00	01 1000
	48.0000	48.0000	0.00	01 0000

Table 5. Serial Port Rates and Master Mode Settings

MCLK Rate (MHz) (Note 1)	Standard Audio Sample Rate (kHz)	Actual Master Mode xSP_ LRCK Rate (Fs_{ext}) (kHz)	Deviation (%)	Settings for x_ MMCC[5:0] (Note 2)
6.5000	8.0000	7.9951	-0.06	11 1100
	11.0250	11.0169	-0.07	11 0101
	12.0000	11.9926	-0.06	11 0100
	16.0000	15.9902	-0.06	10 1100
	22.0500	22.0339	-0.07	10 0101
	24.0000	23.9852	-0.06	10 0100
	32.0000	31.9803	-0.06	01 1100
	44.1000	44.0678	-0.07	01 0101
	48.0000	47.9705	-0.06	01 0100
6.4000	8.0000	8.0000	0.00	11 1110
	11.0250	11.0345	0.09	11 0111
	12.0000	12.0000	0.00	11 0110
	16.0000	16.0000	0.00	10 1110
	22.0500	22.0690	0.09	10 0111
	24.0000	24.0000	0.00	10 0110
	32.0000	32.0000	0.00	01 1110
	44.1000	44.1379	0.09	01 0111
	48.0000	48.0000	0.00	01 0110

Notes:

1. Refer to section "Internal Master Clock Generation" on page 42.
2. See "XSP Master Mode Clock Control Dividers" on page 89, "ASP Master Mode Clock Control Dividers" on page 90, and "VSP Master Mode Clock Control Dividers" on page 92 for details regarding MMCC control.
3. For this row, the xSP_LRCK rate and resulting deviation varies based on the programming of MCLKDIV and x_SCLK=MCLK. The values given, ValueA/ValueB, are applicable according to the rule set in Table 6.

Table 6. Actual xSP_LRCK Rate/Deviation Selector for Note 3

MCLKDIV[2:0]	MCLK Divide Ratio	x_SCLK=MCLK	SCLK=MCLK Mode	Applicable Value
xxx	x	00b	SCLK ≠ MCLK	ValueA
xxx	x	10b	SCLK = MCLK	ValueB
000	1	11b	SCLK = Pre-MCLK	ValueB
010	2	11b	SCLK = Pre-MCLK	ValueA

4.8.6 Formats

Table 7 lists formats supported on the CS42L73 serial ports:

Table 7. Supported Serial Port Formats

Serial Port	I²S Format	PCM Format
XSP	√	√
ASP	√	x
VSP	√	√

The XSPDIF and VSPDIF register bits are used to select the format for the XSP and VSP. There is no selector for the ASP, since it always uses I²S format.

4.8.6.1 I²S Format

Selecting I²S format provides the following behavior:
- Up to 24 bits/sample of stereo data can be transported (see "Data Bit Depths" on page 57)
- Master or Slave timing may be selected
- xSP_LRCK identifies the start of a new sample word and the active stereo channel (A or B)
- Data is clocked into the xSP_SDIN input using the rising edge of xSP_SCLK
- Data is clocked out of the xSP_SDOUT output using the falling edge of xSP_SCLK
- Bit order is MSB to LSB

Refer to section "Mono/Stereo" on page 57 for details on how the stereo nature of the I²S format impacts the operation of the VSP.

The signaling for I²S format is shown in Figure 27.

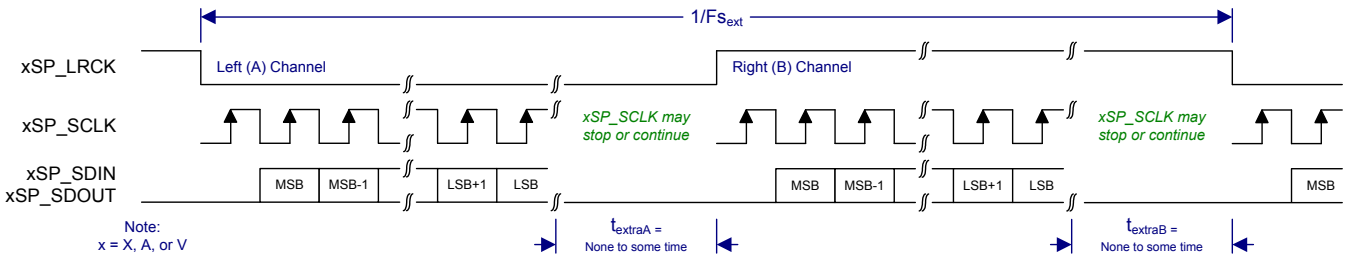


Figure 27. I²S Format

4.8.6.2 PCM Format

If PCM format is selected:
- 16 bits/sample of mono data can be transported (refer to "Data Bit Depths" on page 57)
- Slave timing is supported
- xSP_LRCK (aka WA) identifies the start of a new sample word, acting as a Word-Aligner
- Data is clocked into the xSP_SDIN input using the falling edge of xSP_SCLK
- Data is clocked out of the xSP_SDOUT output using the rising edge of xSP_SCLK
- Bit order may selected as MSB-to-LSB or LSB-to-MSB
- The PCM Mode must be selected

PCM Format supports word bit-order reversal (LSB-to-MSB vs. MSB-to-LSB) via the XPCM_BIT_ORDER and VPCM_BIT_ORDER bits. If enabled, the data in the location (refer to the signaling waveforms in Figures 28 to 30) normally occupied by the data's MSB bit is occupied by the data's LSB bit, the location normally occupied by the data's MSB-1 bit is occupied by the data's LSB+1 bit, and so on.

The X_PCM_MODE[1:0] and V_PCM_MODE[1:0] fields select how WA (xSP_LRCK) may vary in width and location vs. the data.

Mode 0:
- WA may be one or two xSP_SCLK periods wide
- 1st data bit is transported in the cycle following WA
- No data is sampled into the CS42L73 during WA
- When WA is 2 xSP_SCLK periods wide, the first data bit is output from the CS42L73 for 2 cycles, during the last active cycle of WA and during the bit that follows WA (as usual)

Section "Mono/Stereo" on page 57 describes how the mono nature of the PCM format affects operation.

Signaling for all PCM format modes, with xPCM_BIT_ORDER = 0b (MSB-to-LSB), are shown in Figures 28 to 30).

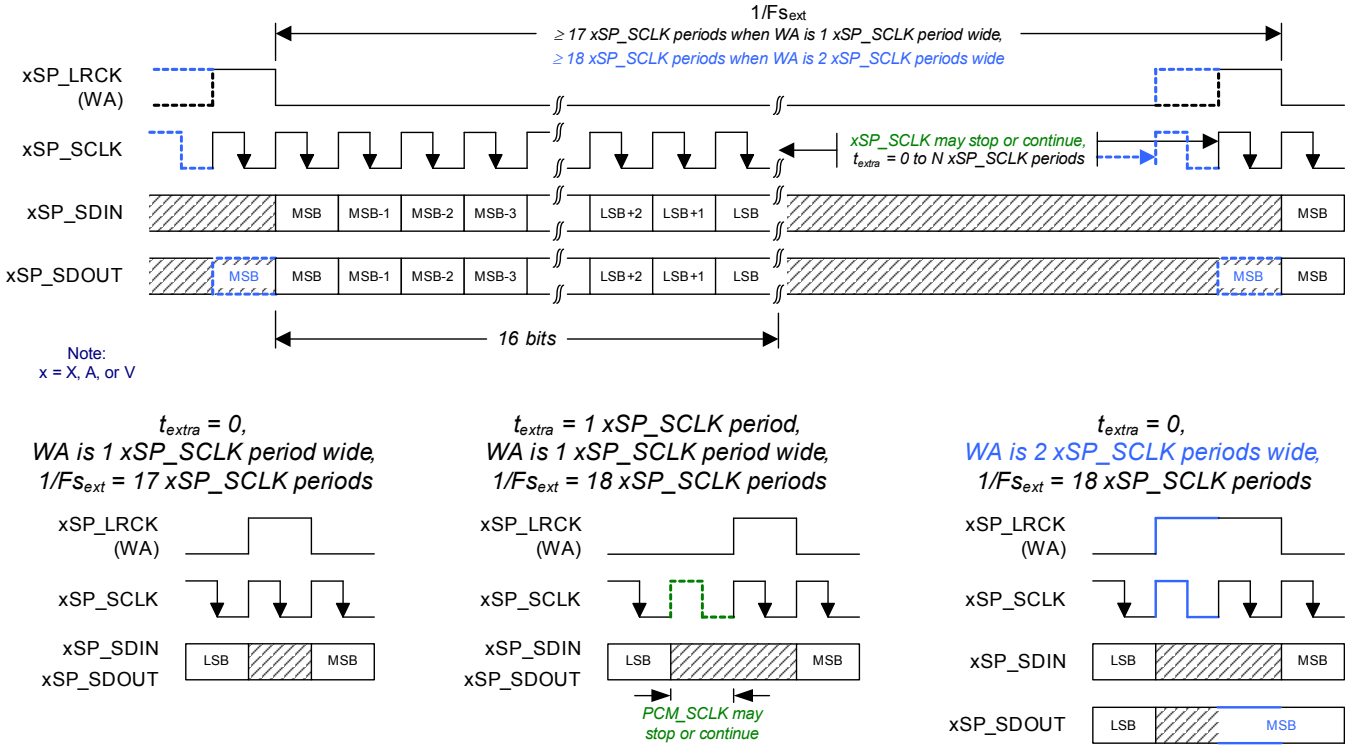


Figure 28. PCM Format—Mode 0

Mode 1:

- WA may be one or up to all-but-one xSP_SCLK periods wide
- 1st data bit is aligned to WA

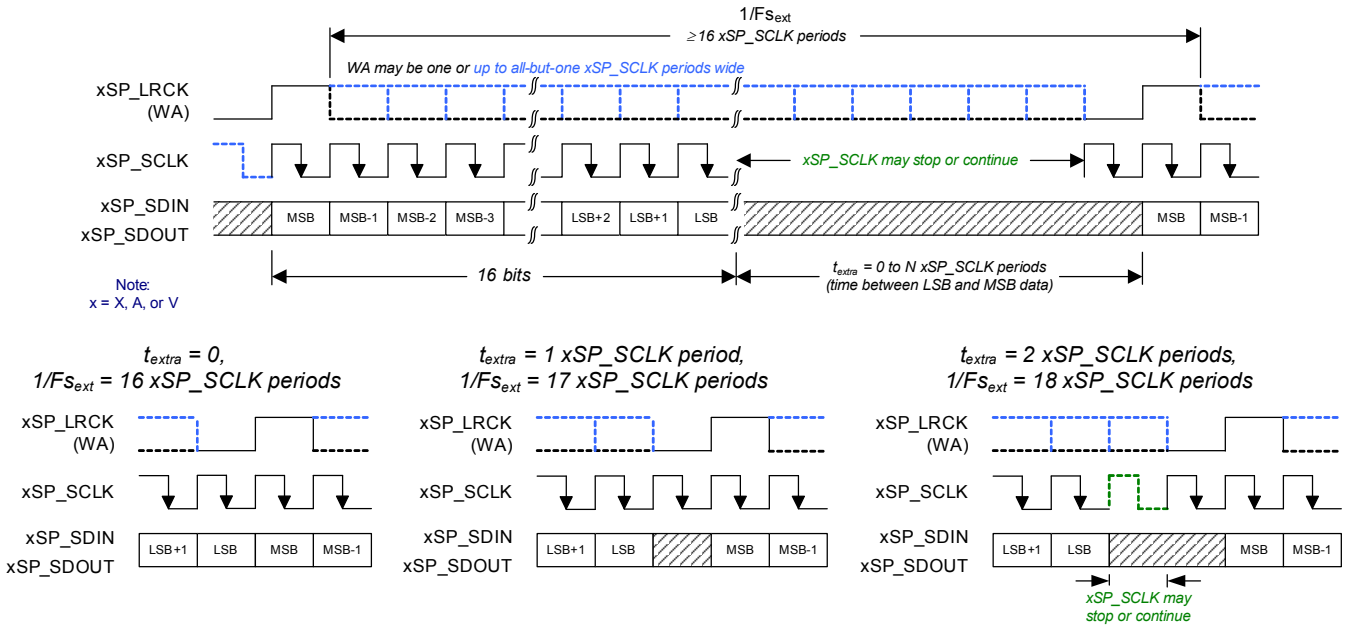


Figure 29. PCM Format—Mode 1

Mode 2:

- WA may be one xSP_SCLK period wide
- 1st data bit follows WA
- Last data bit may be aligned to WA

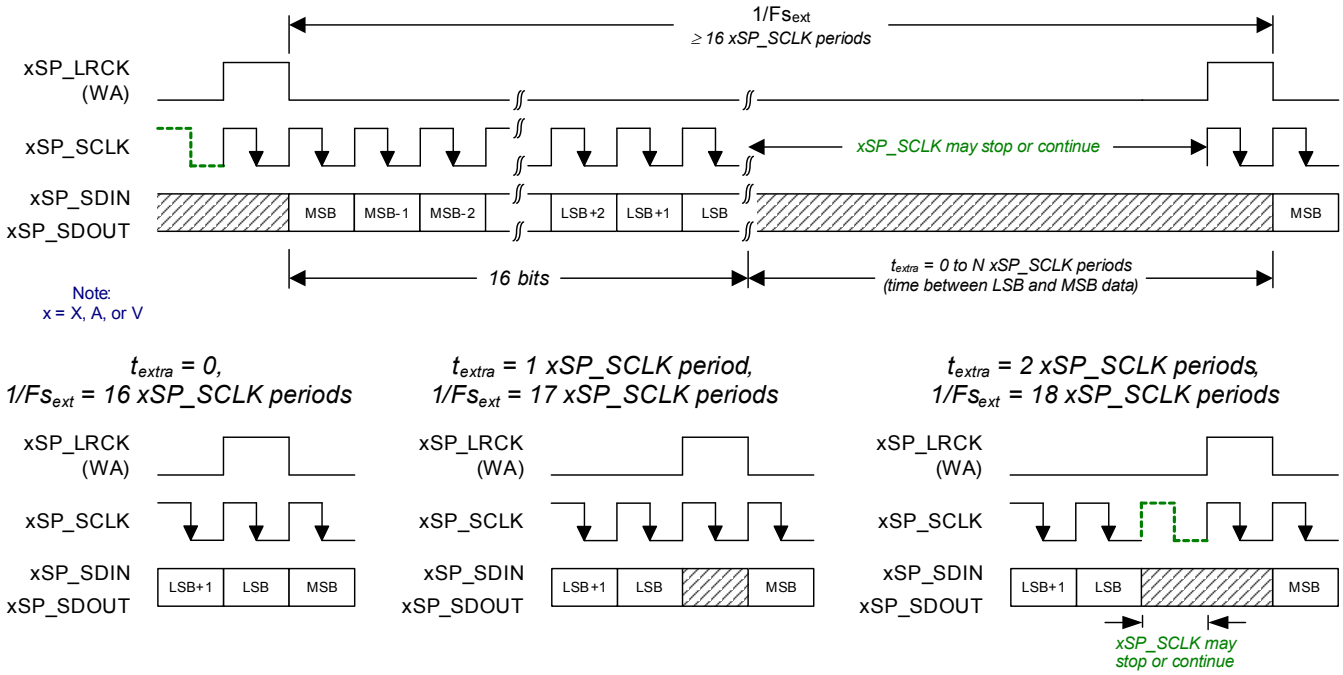


Figure 30. PCM Format—Mode 2

4.8.7 Mono/Stereo

Stereo/mono conversion is required whenever the number of channels for a serial port interface format does not match the number of channels of the ASRC that connects the serial port to the digital mixer.

When the mono PCM format is configured on a port that has a stereo input ASRC, the mono input data is automatically fanned-out by the CS42L73 to both ASRC channels. The XSP is the only serial port where this configuration is possible.

When the stereo I²S format is configured on a port that has a mono input ASRC, one of the input channels is selected by the user to be sent to the ASRC. The VSP is the only serial port where this configuration is possible. The channel selection register bit is named V_SDIN_LOC.

When serial port that supports the stereo I²S format, naturally, a stereo ASRC will feed that port. If that port also supports the mono PCM format, only one of the ASRC's output channels will be transmitted when PCM format is selected. In this case, the digital mixer must be configured to output a mono-mix of its output to both stereo ASRC inputs that are destined for the serial port in question (for more information, including programming instructions, refer to section "Mono and Stereo Paths" on page 63). The XSP and VSP are the only serial ports where this configuration is possible.

4.8.8 Data Bit Depths

The CS42L73's Serial Ports can transmit and receive up to 24 bits of audio data per sample. The number of bits varies depending on the interface format selected and the clocking used.

4.8.8.1 I²S Format Bit Depths

The data word length of the I²S interface format (refer to section "I²S Format" on page 55) is ambiguous. Fortunately, the I²S format also left justified, having a MSB-to-LSB bit ordering, which negates the need for a word length control register. The following text describes how different bit depths are handled with the I²S format.

The CS42L73 will always transmit 24-bit-deep data if at least 24 serial clocks are present per channel sample. If less than 24 serial clocks are present per channel sample, it will output as many bits as there are clocks. If there are more than 24 serial clocks per channel sample, it will output zeros for the additional clock cycles after the 24th bit. The receiving device is expected to load the data in MSB-to-LSB order until its word depth is reached, whereupon it must discard any remaining LSBs from the interface.

The CS42L73 will always attempt to receive 24 bits of data, regardless of the sourcing device's data-bit-depth. If there are less than 24 serial clock cycles per channel sample, it will load the MSBs of its internal 24-bit-wide word with the data associated with all the serial clocks and then augment this data by filling in the LSBs with zeros. If there are more than 24 serial clock cycles per channel sample, all the received data after the 24th bit is discarded.

For instance, if the source data is 16 bits long and the serial clock toggles for 20 cycles per channel, the 16 MSBs of the 24-bit internal data word will be loaded with the 16 bits of source data, whatever follows on the xSP_SDIN input for the remaining 4 cycles will be loaded into the next 4 bits, and then the 4 LSBs will be filled with zeros.

4.8.8.2 PCM Format Bit Depths

For the PCM interface format (refer to section "PCM Format" on page 55), the data bit depth is always 16 bits per sample. Given this unambiguous word length, the following simpler process is used to handle the fact that less than 24 bits are used.

The CS42L73 places the 16 MSBs of its internal 24-bit-wide word into the shorter transmitted (xSP_SDOUT) word and, if, before the next sample word sync pulse, there are additional serial clocks after the 16th transmitted bit, the data associated with the additional serial clocks is to be discarded by the receiving device.

The CS42L73 loads the 16-bit received (xSP_SDIN) word into the MSBs of its internal 24-bit-wide word and then augments the received data with zeros to fill the 8 LSBs of the internal 24-bit word.

Referenced Control	Register Location
MCLKSEL	"Master Clock Source Selection" on page 87
PDN_VSP	"Power Down VSP" on page 83
PDN_ASP_SDOUT	"Power Down ASP SDOUT Path" on page 83
PDN_ASP_SDIN	"Power Down ASP SDIN Path" on page 83
PDN_XSP_SDOUT	"Power Down XSP SDOUT Path" on page 83
PDN_XSP_SDIN	"Power Down XSP SDIN Path" on page 83
3ST_XSP	"Tristate XSP Interface" on page 88
XSPDIF	"XSP Digital Interface Format" on page 88
X_PCM_MODE[1:0]	"XSP PCM Interface Mode" on page 88
XPCM_BIT_ORDER	"XSP PCM Format Bit Order" on page 88
X_SCK=MCK[1:0]	"XSP SCLK Source Equals MCLK" on page 88
X_M/S	"XSP Master/Slave Mode" on page 89
X_MMCC[5:0]	"XSP Master Mode Clock Control Dividers" on page 89
3ST_ASP	"Tristate ASP Interface" on page 89
A_SCK=MCK[1:0]	"ASP SCLK Source Equals MCLK" on page 90
A_M/S	"ASP Master/Slave Mode" on page 90
A_MMCC[5:0]	"ASP Master Mode Clock Control Dividers" on page 90
3ST_VSP	"Tristate VSP Interface" on page 91
VSPDIF	"VSP Digital Interface Format" on page 91
V_PCM_MODE[1:0]	"VSP PCM Interface Mode" on page 91
VPCM_BIT_ORDER	"VSP PCM Format Bit Order" on page 91
V_SDIN_LOC	"VSP SDIN Location" on page 92
V_SCK=MCK[1:0]	"VSP SCLK Source Equals MCLK" on page 92
V_M/S	"VSP Master/Slave Mode" on page 92
V_MMCC[5:0]	"VSP Master Mode Clock Control Dividers" on page 92

4.9 Asynchronous Sample Rate Converters (ASRCs)

The CS42L73 uses ASRCs to bridge potentially different sample rates at the serial ports and within the Digital Processing core. Two stereo ASRCs are used for the XSP and ASP paths, one mono ASRC is used for the VSP input path, and three stereo ASRCs are used for the XSP, ASP, and VSP output paths. The Digital Processing side (as opposed to the serial port side) of the ASRCs connect to the digital mixer (refer to section "Digital Mixer" on page 41). The architecture and operation of the ASRCs is described in this section.

Multirate digital signal processing techniques are used to conceptually up-sample the incoming data to a very high rate and then down-sample to the outgoing rate.

Internal filtering is designed so that a full input audio bandwidth of 20 kHz is preserved if the input sample and output sample rates are greater than or equal to 44.1 kHz. When the output sample rate becomes less than the input sample rate, the input is automatically band limited to avoid aliasing artifacts in the output signal.

Any jitter in the incoming signal has little effect on the dynamic performance of the rate converter and has no influence on the output clock.

A Digital PLL (DPLL) continually measures the heavily low-pass-filtered phase difference and frequency ratio between input and output sample rate clocks. The DPLL, using these measures, adjusts on-the-fly the coefficients of a linear time varying filter. This filter processes a synchronously oversampled version of the input data. The output of this filter is then resampled to the output sample rate.

The input and output sample rate clocks are derived from the external serial port sample clock (xSP_LRCK) and the internal Fs clock respectively in the case of the input serial ports. They are derived in the reverse order in the case of the output serial ports.

The lock time of the ASRCs can be minimized by programming the serial port interface sample rates into the register control words XSPFS[3:0], ASPFS[3:0], and VSPFS[3:0]. If the rates are unknown, program these register control words to "don't know" and incur longer lock times. Proper operation is not assured if the sample rates are mis-programmed.

Refer to section "ASRC Attributes" on page 129 for additional information regarding the ASRCs.

Referenced Control	Register Location
XSPFS[3:0].....	"XSP Sample Rate" on page 93
ASPFS[3:0].....	"ASP Sample Rate" on page 90
VSPFS[3:0].....	"VSP Sample Rate" on page 93

4.10 Input Paths

4.10.1 Input Path Source Selection and Powering

Table 8 describes how the PDN_ADCx and PDN_DMICx controls affect the CS42L73 Input Path. PDN_ADCx has priority over PDN_DMICx.

Table 8. Input Path Source Select and Digital Power States

Control Register States		Selected Input Path x Data Source	Input Path x Digital Power State
PDN_ADCx	PDN_DMICx		
0	X	ADCx	On
1	0	DMICx	
	1	Don't Care	Off

4.10.2 Digital Microphone (DMIC) Interface

The DMIC Interface can be used to collect Pulse Density Modulation (PDM) audio data from the integrated ADCs of one or two digital microphones. The following sections outline how the interface may be used.

4.10.2.1 DMIC Interface Description

The DMIC Interface consists of a serial-data shift clock output (DMIC_SCLK) and a serial data input (DMIC_SD). The "Typical Connection Diagram" on page 17 shows how to connect two digital microphones (Left and Right) to the CS42L73. Note how the clock is fanned out to both digital microphones and both digital microphone's data outputs share a single signal line to the CS42L73. To share a line, the digital microphones tristate their output during one phase of the clock (high or low part of cycle, depending on how they are configured via their L/R input). Alternating between one digital microphone outputting a bit of data and then the other microphone outputting a bit of data, the digital microphones time domain multiplex on the signal data line. Data line contention is avoided by entering the high-impedance tristate faster than removing it.

If only one digital microphone is to be used, the connections to the remaining digital microphone are unchanged from those used for two digital microphones.

The DMIC_SD signal is weakly pulled (up to power or down to ground as per table "Digital Pin/Ball I/O Configurations" on page 16) by its CS42L73 input. When the DMIC Interface is active, this pulling is not strong enough to affect the multiplexed data line significantly while it is in tristate between data slots. When the interface is disabled and the data line is not driven, the weak pulling will ensure the CS42L73 input avoids the power-consuming mid-rail voltage.

4.10.2.2 DMIC Interface Signaling

The signaling on the DMIC Interface is illustrated in following figure. Notice how the left channel (i.e., A or DATA1 Channel) data from the "left" microphone is sampled on the rising edge of the clock and the right channel (i.e., B or DATA2 channel) data from the "right" microphone is sampled on the falling edge.

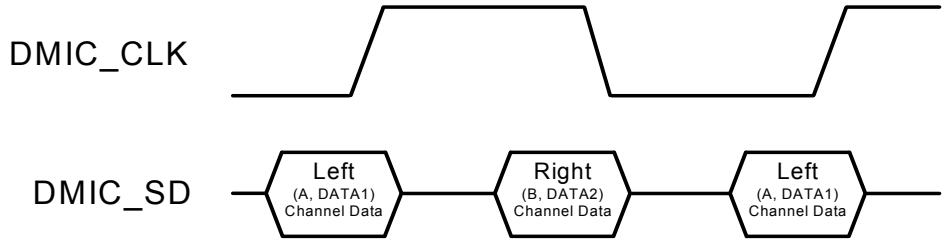


Figure 31. Digital Mic Interface Signaling

4.10.2.3 DMIC Interface Powering

The DMIC Interface is powered up or down (via the register controls PDN_ADCx and PDN_DMICx) according to the logic shown in Table 9.

Table 9. Digital Mic Interface Power States

Control Register States				Digital Mic Interface Power State
PDN_ADCA	PDN_DMICA	PDN_ADCB	PDN_DMICB	
1	0	X	X	On
X	X	1	0	
Otherwise				Off

Note: When the DMIC Interface is off, the DMIC_SCLK pin is set to inactive low.

4.10.2.4 DMIC Interface Clock Generation

Table 10 outlines the supported DMIC Interface Serial Clock (DMIC_SCLK) nominal frequencies and how they are derived from the internal Master Clock (MCLK).

Table 10. Digital Microphone Interface Clock Generation

MCLK Rate (MHz)	Divide Ratio	DMIC_SCLK Rate (MHz)	DMIC_SCLK_DIV Programming
5.6448	2	2.8224	0
	4	1.4112	1
6.0000	2	3.0000	0
	4	1.5000	1
6.1440	2	3.0720	0
	4	1.5360	1
6.5000	2	3.2500	0
	4	1.6250	1
6.4000	2	3.2000	0
	4	1.6000	1

4.11 Digital Mixer

The digital mixer facilitates the mixing and routing of the CODEC's inputs to its outputs. Figure 32. Digital Mixer Diagram on page 62 illustrates the architecture and connectivity of the digital mixer.

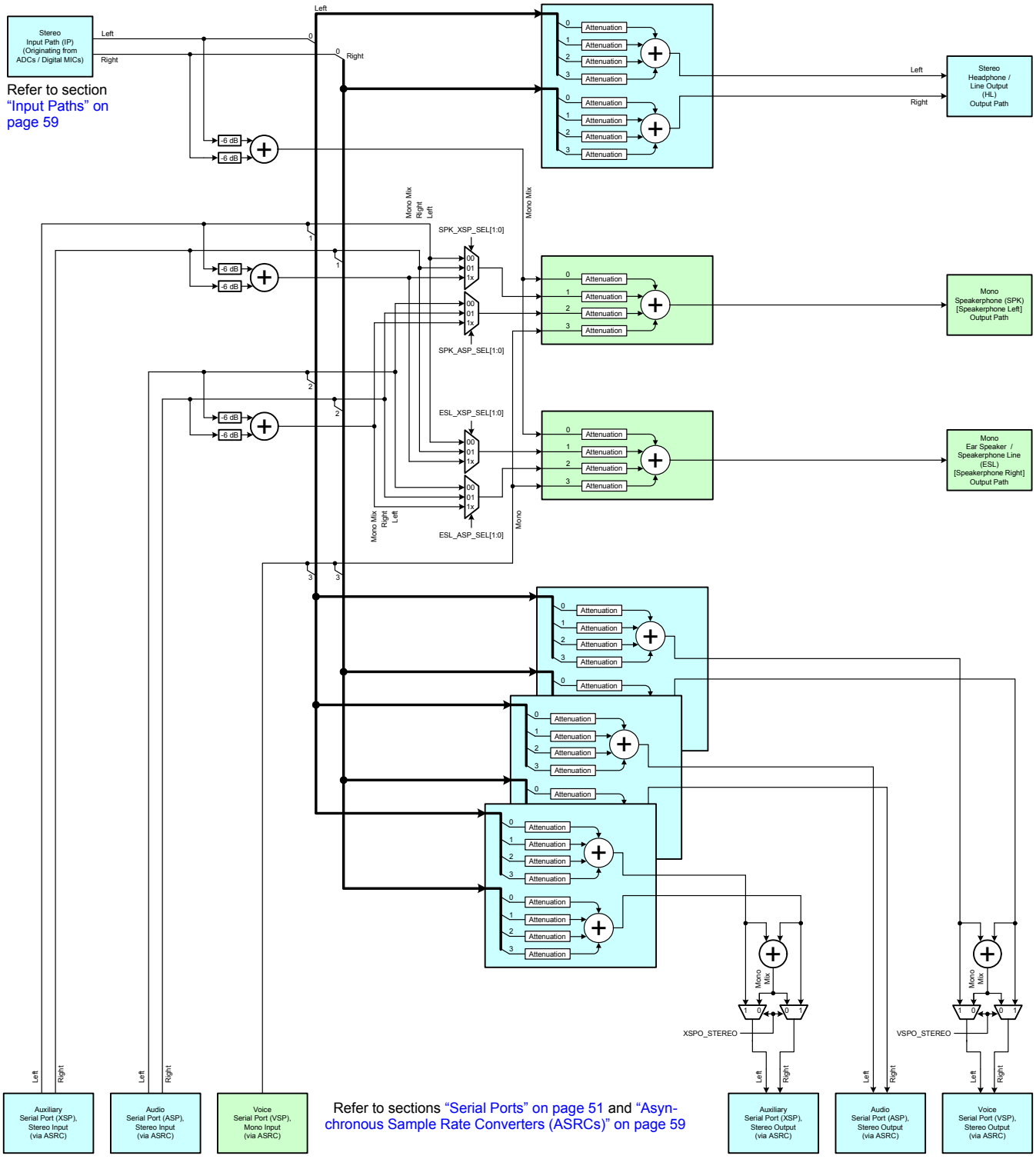


Figure 32. Digital Mixer Diagram

4.11.1 Mono and Stereo Paths

Notice how Figure 32 distinguishes between stereo and mono channels; there are buses for the stereo inputs and the digital mixer's inputs, outputs, and programmable attenuation mixers are color coded (green for mono, blue for stereo).

The figure also illustrates how the outputs destined, via their respective ASRCs, for the XSP and VSP can be configured for normal stereo channeling or to send a mono mix to both stereo channels (as per register bits XSPO_STEREO and VSPO_STEREO). For details on when to use these controls, refer to section "Mono/Stereo" on page 57).

The mixers that fed the green mono analog outputs have flexible ASP and XSP input source selectors (refer to register controls ESL_ASP_SEL[1:0]. ESL_XSP_SEL[1:0], SPK_ASP_SEL[1:0], and SPK_XSP_SEL[1:0]). These selectors are used to either pick one of the stereo inputs or a mono mix of them. One use of these selectors would be to configure stereo play of the ASP input to the Speakerphone (SPK) and Speakerphone Line Outputs (SPKLO). The left channel of the ASP would be routed to the SPK and the right ASP channel would be routed to the SPKLO.

4.11.2 Mixer Input Attenuation Adjustment

Each time a mixer's input attenuation is adjusted, including the setting or resetting the mute condition (via register controls "Stereo *_A[5:0]" and "Mono *_A[5:0]"), a soft ramp can selectively (via register control bit MXR_SFTR_EN) be used to smooth the transition, ensuring no inharmonious artifacts are introduced. The only exception to the selectivity of soft ramping occurs when an ASRC that feeds the digital mixer loses lock. In this situation, to prevent unpredictable data from reaching an device output, the ASRC freezes its last output value sent to the mixer and the mixer soft ramps the affected inputs to mute.

Soft-ramping logarithmically traverses the digital mixer's -90 to 0 dB attenuation range according to the register control MXR_STEP[2:0]. The inaudible steps from/to mute (-∞ dB) to/from -90 dB occur in a linear (vs. logarithmic) magnitude manner. Table 11 lists mixer soft ramping rates for the nominal and extreme internal sample rates (Fs) and all MXR_STEP[2:0] configurations.

Table 11. Digital Mixer Soft Ramp Rates

Fs Rate (kHz)	MXR_STEP[1:0] Setting	Step Size (dB)	Step Period (# Fs period/step)	Soft Ramp Rate (dB/s)
44.100	000	1/8	1	5,512.5
	001	1/4	1	11,025.0
	010	1/2	1	22,050.0
	011	1	1	44,100.0
	100	1/8	4	1,378.1
	101	1/8	2	2,756.3
48.000	000	1/8	1	6,000.0
	001	1/4	1	12,000.0
	010	1/2	1	24,000.0
	011	1	1	48,000.0
	100	1/8	4	1,500.0
	101	1/8	2	3,000.0
50.781	000	1/8	1	6,347.6
	001	1/4	1	12,695.3
	010	1/2	1	25,390.5
	011	1	1	50,781.0
	100	1/8	4	1,586.9
	101	1/8	2	3,173.8

4.11.3 Powered-Down Mixer Inputs

If an input to the digital mixer is powered down (refer to register controls "Power Control 1 (Address 06h)" on page 82 and "Power Control 2 (Address 07h)" on page 83), that input must be muted. The CS42L73 does not automatically mute mixer inputs that are powered down. If a mixer input is not to be used and is not muted upstream, set the input's attenuation to mute.

To minimize audio disturbances, it is recommended that the mute on the mixer input (that is to be powered down) be applied (at the mixer or upstream) using a soft ramp and that the power-down only occur after the attenuation has ramped fully to mute.

4.11.4 Avoiding Mixer Clipping

Digital mixers are essentially adders. As such, when more than one input is fed into a mixer the potential for overflow exists, depending on the bit word length of the inputs and the mixer and depending on the input value range. For example, if two full-range, signed 4-bit channels were mixed to a signed 4-bit result, whenever the sum of the two inputs falls outside the -8 to +7 range, the hypothetical mixer would overflow causing undesired output signal distortion (wrapping).

No mixers within CS42L73's digital mixer are susceptible to overflow because they all have a sufficient number of accumulator bits. If any mixer's result exceeds the bit width of the signal data path, the result is forced either to the full-scale maximum or the minimum value, which ensures the signal is clipped vs. being distorted (by the wrapping effect of truncating the accumulator result to fit into the data path width).

Attention is required to ensure clipping does not occur within the digital mixer. Of course, if the digital mixer is fed a signal that was clipped elsewhere, its output reflects that external clipping.

The three mixers in Figure 32 that provide mono versions of input stereo channels (the Input Path, XSP, and ASP inputs) are impervious to clipping by design. They have -6 dB of attenuation applied to their inputs (see "Mixer Attenuation Values" on page 65). Mathematically this amounts to InputA/2 + InputB/2, which illustrates that, given the input and mixer output bit widths are the same, the result can never clip.

Refer to the mixers on the lower-right side of Figure 32 that are used to provide mono versions of XSP and VSP output channels. They rely on the input attenuation settings of the stereo mixers that feed them to avoid clipping. If the XSP or VSP output is configured as mono, the user must program the sourcing stereo mixer's attenuators to provide mixer outputs that are at least 6 dB down from full scale. This will prevent mixer-caused clipping of the signals that are sent to the XSP and VSP.

All the other mixers are susceptible to clipping. For these mixers, the recommended minimum premixer attenuation level settings (refer to "Mixer Attenuation Values" on page 65) to avoid mixer clipping are provided in Table 12.

Table 12. Digital Mixer Nonclipping Attenuation Settings

Number of Active Channels into Mixer	Max Signal Strength Allowed per Input	Minimum Attenuation (dB) Setting Allowed per Input
1	1	0
2	1/2	6
3	1/3	10
4	1/4	12

For this table, full-scale inputs are assumed (no preattenuation) and that there is no relative volume adjustment between inputs. If any inputs are at less than full scale, less attenuation can be set while still avoiding mixer clipping. If there is to be a relative volume adjustment between the inputs, less attenuation can be set for one or more inputs so long as the other input(s) are attenuated sufficiently to avoid clipping (e.g., with three full-scale inputs, one input could be attenuated by 6 dB, if the other two are attenuated by 12 dB).

4.11.5 Mixer Attenuation Values

The digital mixer contains fixed attenuation blocks and programmable attenuation blocks. The attenuation values associated with these blocks are as described in Figure 32 or in the related control register descriptions, except for one caveat. The caveat is the result of the binary math of the mixer circuit and design intent. For all settings other than 0 dB, the actual attenuation on the mixer input is a little more than the rounded-to-integer number listed in the register description. These small offsets increase with larger amounts of attenuation. At the largest attenuation setting, -62 dB, the applied attenuation is actually -62.216 dB.

The benefits of the offsets are twofold and relate to how premixer attenuation is applied (refer to the "Avoiding Mixer Clipping" section on page 64). First, for commonly used -6n dB (n = {1, 2, etc.}) attenuation settings, the offset rounds the attenuation to exactly the desired $1/2^n$ factor (e.g., 20Log(1/2) = 6.021 dB, not 6.000 dB). Secondly, for attenuation settings other than -6n dB, the always positive offset provides slightly more attenuation, yielding some margin to ensure that mixer clipping is avoided.

Referenced Control	Register Location
XSPO_STEREO	"XSP Mixer Output Stereo" on page 117
VSPO_STEREO	"VSP Mixer Output Stereo" on page 117
MXR_SFTR_EN	"Mixer Soft-Ramp Enable" on page 117
MXR_STEP[2:0]	"Mixer Soft-Ramp Step Size/Period" on page 117
"Stereo *_A[5:0]"	"Stereo Mixer Input Attenuation" on page 119
ESL_ASP_SEL[1:0]	"Ear Speaker/Speakerphone Line Output (ESL) Mixer, ASP Select" on page 120
ESL_XSP_SEL[1:0]	"ESL Mixer, Auxiliary Serial Port (XSP) Select" on page 120
SPK_ASP_SEL[1:0]	"Speakerphone (SPK) Mixer, ASP Select" on page 120
SPK_XSP_SEL[1:0]	"Speakerphone (SPK) Mixer, XSP Select" on page 120
"Mono *_A[5:0]"	"Mono Mixer Input Attenuation" on page 121

4.12 Recommended Operating Procedures

The following sections describe the recommended power-up and power-down sequences for typical use cases. Implement these to minimize audible artifacts and to provide the best-possible user experience.

4.12.1 Initial Power-Up Sequence

The initial power-up sequence must be executed whenever power is applied to the CS42L73 from a powered-down state, or if there is a known or suspected disturbance on the power supply that brings it below the "Recommended Operating Conditions" on page 19.

1. Hold $\overline{\text{RESET}}$ low (active) until all the power supply rails have risen to greater than or equal to the minimum recommended operating voltages.
   - Ensure the ramping-up of each of the supplies is smooth (no down-slope regions) and does not take longer than the specified time ($t_{pwr\text{-}rud}$).
   - The last power supply rail to reach its operating voltage must do so within the specified time ($t_{pwr\text{-}rs}$) from when the first power rail reaches its operating voltage. Exception: the VP supply may be applied or removed independently of $\overline{\text{RESET}}$ and the other power rails (except for the VA supply, see (Note 4)).
2. Continue to hold $\overline{\text{RESET}}$ low for at least the specified hold time ($t_{rh(PWR\text{-}RH)}$) after power supplies reach their operating voltage.
3. Bring $\overline{\text{RESET}}$ high.
4. Wait the specified minimum time (t_{ris}) following $\overline{\text{RESET}}$ going high before using the control port.

Refer to the specifications on page 36, page 40, and Figure 10 on page 36 to find the durations referenced in this power-up sequence.

Note: A valid MCLK signal is not required to be present to communicate with the control port, however changes made to the control port will not take effect until a valid MCLK signal is present. An MCLK

signal may be applied any time during the power-up sequence. If an MCLK signal is present when RESET is brought high, it is recommended that the rising edge of RESET be synchronized to the falling edge of MCLK. After RESET is brought high, the MCLK signal must not have any glitched pulses. A glitched pulse is any pulse that is shorter than the period defined by the minimum/maximum MCLK signal duty cycle specification and the nominal frequency of the MCLK; see the specifications on page 36.

4.12.2 Power-Up Sequence (xSP to HP/LO)

This sequence powers up the CS42L73 and sets basic mixing paths to achieve a playback path to the headphones or lineout. Other output path settings can be substituted for HP/LO. Execute this sequence when playback is desired after either the initial power-up sequence or the power-down sequence.

1. Start with the sequence specified in "Initial Power-Up Sequence" on page 65. If power is already applied, the CS42L73 is to be awakened from a powered down state (refer to section "Power-Down Sequence (xSP to HP/LO)" on page 67) using the following procedure. In either case, the device is in a PDN, PDN_HP/PDN_LO, PDN_xSPSDIN = 1b condition at this point.
2. Activate the MCLK signal feeding one of the MCLKx pins. Configure the internal MCLK according to which pin the clock is applied to. Refer to Section 4.2 "Internal Master Clock Generation" on page 42 for the required configuration. Enable the internal MCLK signal by clearing MCLKDIS.
   Register Controls: **MCLKSEL**, **MCLKDIV**, and **MCLKDIS**
3. To minimize pops on the headphone or line amplifier, the respective analog output must first be muted. Apply the mute immediately by ensuring Analog Soft Ramping (ANLGOSFT) is disabled before changing the HP/LO settings.
   Register Controls: **ANLGOSFT** and then
   Register Controls: **HPxAMUTE/LOxAMUTE**
4. Now that the headphone or line amplifiers are muted, start the power-up of the core and HP/LO DAC.
   Register Controls: **PDN** and **PDN_HP/PDN_LO**
5. If the serial port (xSP) is to be operated in slave mode, activate the external xSP clock signals (xSP_SCLK and xSP_LRCK).
6. Configure the serial port.
   Register Controls: Refer to the xSP control and master mode clocking control registers.
7. Power up the xSP input path.
   Register Controls: **PDN_xSPSDIN**
8. Configure digital volume/muting for the ramping desired for audio startup:
   - Analog soft ramping. Set the associated enable bit now that the analog mutes have had time to be applied.
     Register Controls: **ANLGOSFT, mixer volumes**
   - Digital soft ramping. Ensure the digital mixer and/or HP/LO DAC digital volume is muted and digital soft-ramping is configured/enabled.
     Register Controls: **DIGSFT**, **HLxDMUTE, mixer volumes**
   - No soft ramping. Configure the digital and analog soft-ramp controls accordingly and set the digital mixer volume to the desired level.
     Register Controls: **ANLGOSFT, DIGSFT, mixer volumes**
9. Set analog volumes, according to whether soft ramping is used:
   - Soft ramping (digital or analog). Set the desired analog volumes on the HP/LO output.
   - No soft-ramping. Set the analog volume to maximum attenuation.

   Register Controls: HPx_AVOL/LOx_AVOL
10. Set the desired digital volume on the HP/LO output.
    Register Controls: **HLx_DVOL**
11. Wait for the headphone/line amplifier to finish powering up. For most configurations, the used ASRC should lock during this time (refer to section "Lock Time" on page 130) as indicated by the status bit

in "Audio ASRC Data In Lock" on page 124.

12. Start transmission of audio data to device.
13. Ramp up audio output.
    - Unmute the analog volume for the headphone or line amplifiers.
      Register Controls: **HPxAMUTE/LOxAMUTE**
    - If digital soft-ramping is used, unmute the mixer path (setting mixer volume) and/or DAC digital volume.
      Register Controls: mixer volume and/or **HLxDMUTE**
    - If no soft ramping is used, ramp up the analog and mixer volume to the desired level with however many steps (control port writes) desired. This method (vs. using CS42L73's soft-ramp features) allows for potentially faster but more zipper-noise like volume ramp-ups or for ultraslow ramp ups with equal-to (vs. analog soft-ramping) or coarser/noisier (vs. digital soft-ramping) steps.

4.12.3 Power-Down Sequence (xSP to HP/LO)

The power-down sequence must be used when the playback path is no longer needed and low power consumption is desired and/or before calling the final power-down sequence.

1. To minimize pops on the headphone or line amplifier, according to the soft-ramping configuration:
    - Analog soft ramping. Mute the analog outputs.
      Register Controls: **HPxAMUTE/LOxAMUTE**
    - Digital soft ramping. Mute the mixer path and/or DAC digital volume.
      Register Controls: mixer volume and/or **HLxDMUTE**
      If either digital or analog soft ramping is being used, wait until the soft ramping to mute is completed (refer to sections "Analog Output Soft Ramp" on page 96, "Digital Soft-Ramp" on page 96, and "Mixer Soft-Ramp Step Size/Period" on page 117 for ramp rate values that can be used to calculate the ramp-to-mute time).
    - No soft-ramping. Ramp the analog and/or digital volume down to the minimum level (maximum attenuation) with however many steps (control port writes) as is desired.
      Register Controls: **HPx_AVOL/LOx_AVOL**, "Stereo Mixer Input Attenuation (Addresses 35h through 54h)" on page 118, and/or **HLx_DVOL**
2. Power down the device.
   Register Controls: **PDN**, **PDN_HP/PDN_LO**, and **PDN_xSPSDIN**
3. Wait to allow the CS42L73's circuits to finish powering down. The amount of time to wait depends on which output path is being powered down.
   HPOUT: 30 ms
   EAR SPKOUT or LINEOUT: 50 ms
   SPKOUT or SPKLINEOUT: 150 ms
4. Deactivate external xSP input signals.
5. Deactivate the MCLK signal first by using the MCLKDIS (if possible) and then by removing the external source.

Note: The PDN and PDN_xx bits do not take effect if the MCLK signal is removed first.

6. If the device is to be completely powered down by removing the power supply rails, follow the sequence specified in "Final Power-Down Sequence" on page 68. Otherwise, optionally bring $\overline{\text{RESET}}$ low to achieve the lowest quiescent current. Note, by setting $\overline{\text{RESET}}$ low, the Control Port register values will return to their default states.

4.12.4 Recommended Sequence for Modification of the MCLK Signal

The CS42L73 requires the MCLK signal to be stable in frequency and uninterrupted whenever any sub-blocks (PDN_xx) are powered up. When it is known there is going to be a change to the MCLK frequency or that it will be stopping/starting, the following procedure should be executed.

1. The CS42L73 must be put into a powered down state using the procedure in section "Power-Down Sequence (xSP to HP/LO)" on page 67.
2. The MCLK signal may then be modified or disabled at its external source (when applicable), and/or changes to the related CS42L73 control registers (see register controls list below) can be made. Use the procedure in section "Power-Up Sequence (xSP to HP/LO)" on page 66 to bring the CS42L73 out of the powered down state.
   Register Controls: **MCLKSEL**, **MCLKDIV**, and **MCLKDIS**

4.12.5 Microphone Enabling/Switching Sequence

When the microphone inputs are enabled or disabled, temporary disturbances will occur on them. In addition, switching the PGA Mux will cause an audible discontinuity disturbance. To avoid the transmission of these disturbances, the following procedure must be used.

1. Mute the ADC output (Input Path Digital) with the soft mute enabled if it is not already muted, and wait until the ADC is fully muted (mute soft-ramp rate is defined in register description "Digital Soft-Ramp" on page 96). Note for initializing the Microphone soft-ramp enable of mute is not necessary.
   Register Controls: **IPxMUTE** and **DIGSFT**
2. Enable and/or disable the MIC bias outputs as desired and wait until all MIC inputs have stabilized (about 20 ms for C_{INM} = 1 µF; refer to "Typical Connection Diagram" on page 17).
   Register Controls: **PDN_MICx_BIAS**
3. If desired, switch the input to the ADC (MICx/LINEx).
   Register Controls: **PGAxMUX**
4. Soft-release ADC mute.

4.12.6 Final Power-Down Sequence

The final power-down sequence must be executed before removing the power for the CS42L73.

1. If not already completed, follow the sequence specified in "Power-Down Sequence (xSP to HP/LO)" on page 67 and the disable steps of "Microphone Enabling/Switching Sequence" on page 68. If other audio paths are active in the CS42L73, use a similar approach to avoid pops and properly shutdown each sub-block of the device.
2. Power down the CS42L73 by setting register bit PDN = 1. If step 1 is not followed a wait of 50 ms is recommended before proceeding.
3. To minimize pops and clicks when the power supplies are pulled to ground, it is recommended that the DISCHG_FILT bit is set before the supplies are pulled to ground. This will discharge the recommended 2.2 µF FILT+ capacitor within approximately 10 ms.
   Register Controls: **DISCHG_FILT**
4. Set $\overline{\text{RESET}}$ low (active).
5. Wait the specified setup time ($t_{rs(RL\text{-}PWR)}$) before lowering the power supply rails to less than the minimum recommended operating voltages (specified in spec. table "Recommended Operating Conditions" on page 19).
6. Continue to hold $\overline{\text{RESET}}$ low at least until all the power supplies have ramped down to ground.
   – Ensure the ramping-down of each of the supplies is smooth (no up-slope regions) and does not take longer than the specified time ($t_{pwr\text{-}rud}$).
   – The last power supply rail to reach ground must do so within the specified time ($t_{pwr\text{-}rs}$) from when the first power rail reaches ground. Exception: the VP supply may be applied or removed independently of $\overline{\text{RESET}}$ and the other power rails (except for the VA supply, see (Note 4)).

Refer to Section on page 36, and Figure 10 on page 36 to find the durations referenced in this power-down sequence.

4.13 Using MIC2_SDET as Headphone Plug Detect

Although the CS42L73 does not have a dedicated headphone plug detect pin, the MIC2_SDET pin may be used to perform a similar function. However, doing so requires that MIC2_SDET not be used as a microphone button short detect.

To use the MIC2_SDET pin as a headphone detect pin, connect the headphone jack pins to the CS42L73 as shown in Figure 33.

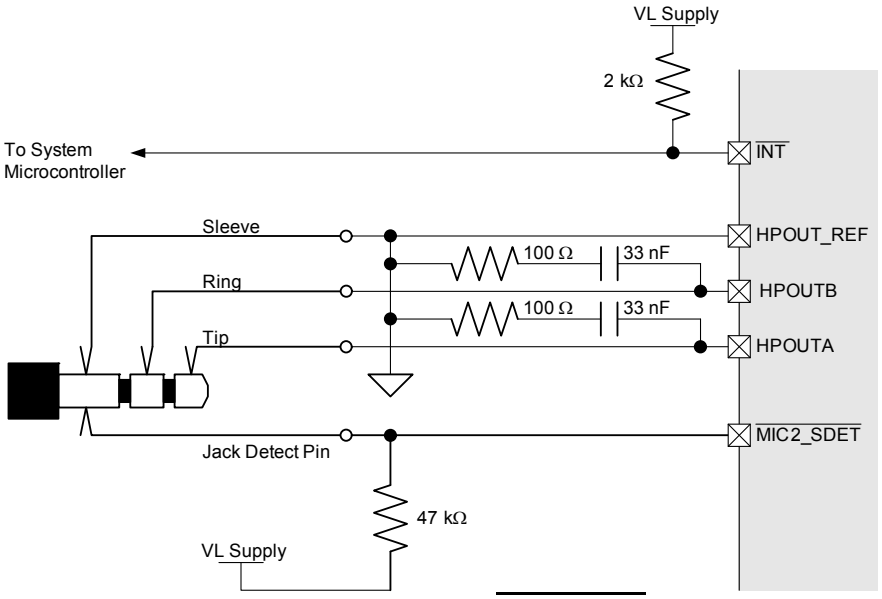


Figure 33. Connection Diagram for Using MIC2_SDET as Headphone Detect

Next, set register 0x5E bit 6 = 1. If no state change other than MIC2_SDET is required to trigger the INT pin, then the value of register 0x5E may be set to 0x40. This unmasks the MIC2_SDET status bit (register 0x60 bit 6) so that the INT pin will be driven low or pulled high based on the MIC2_SDET status bit.

With the system connected and registers configured as described above, the CS42L73 will drive the INT pin low when a high-to-low transition on MIC2_SDET is detected (indicating headphone plug insertion). The INT pin will also be asserted when a low-to-high transition on MIC2_SDET is detected (indicating headphone plug removal). The INT pin will remain low unless register 0x60 is read; reading register 0x60 sets the INT pin high. The MIC2_SDET state (shorted or not shorted) can be read via register 0x60 bit 6 at any time.

Figure 34 summarizes the behavior of the INT pin when register 0x5E = 0x40.

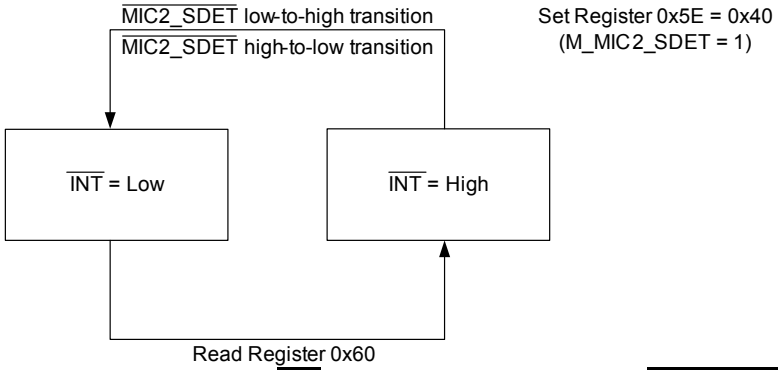


Figure 34. Flow Diagram Showing the INT Pin State in Response to MIC2_SDET State Changes

4.14 Headphone Plug Detect and Mic Short Detect

To implement “headphone plug detect,” a suitable jack and system GPIO are required. Figure 35 shows two common implementations of headphone plug using additional pins within the jack. Jack detect pin type B (refer to Figure 35) is preferred, because type A requires additional filtering to remove signal from the HPOUTA pin when the headset is disconnected.

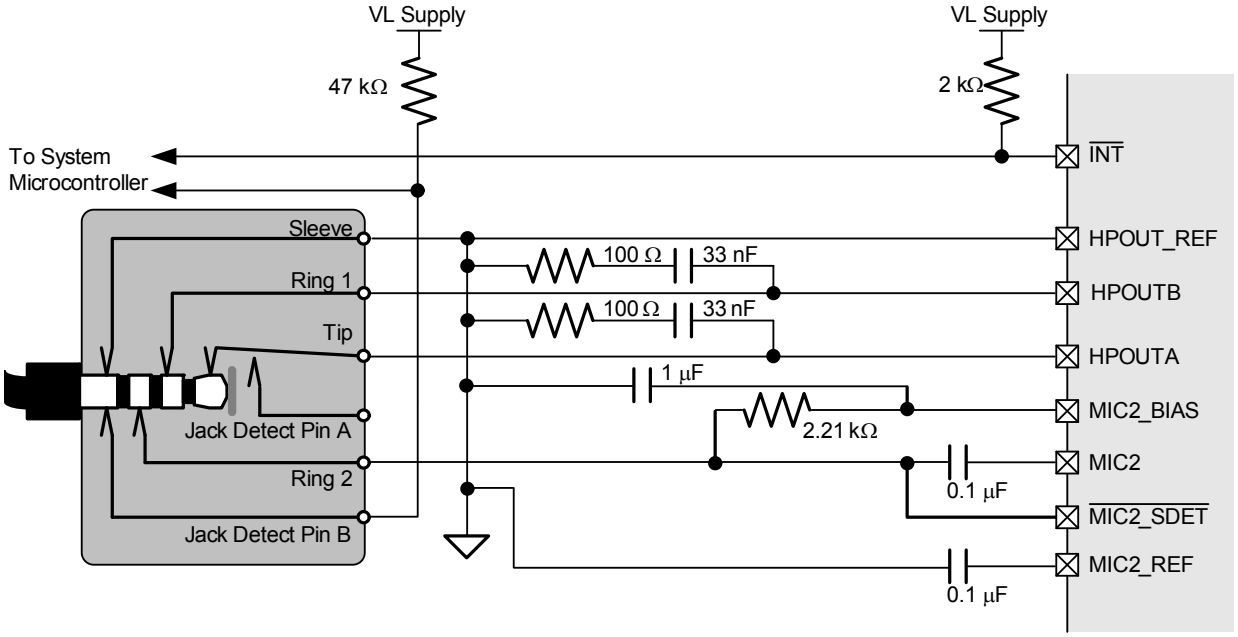


Figure 35. Connection Diagram for Headphone Detect with Additional Short Detect

Note that Figure 35 shows one possible configuration of TRRS (Tip, Ring 1, Ring 2, Sleeve) signaling regarding ring 2 and sleeve. Some headsets in the marketplace use an alternate pinout and assign the mic signal to sleeve and ground to ring 2. The decision of which headset type to support must be made in hardware, as the CS42L73 does not support detection of or automatic reconfiguring of the pins for alternate headset pinout assignments.

Microphone short detect is accomplished using the internal detect feature of the CS42L73. Connect the short detect pin as shown in Figure 35. Next, set register 0x5E bit 6 = 1. If no other state change other than MIC2_SDET is required to trigger the INT pin, the value of register 0x5E may be set to 0x40. This unmasks the MIC2_SDET status bit (register 0x60 bit 6) so that the INT pin will be driven low or pulled high based on the MIC2_SDET status bit.

With the system connected and registers configured as described above, the CS42L73 will drive the INT pin low when a high-to-low transition on MIC2_SDET is detected (indicating the mic short button has been pressed). The INT pin will also be driven low when a low-to-high transition on MIC2_SDET is detected (indicating the button has been released). The INT pin will remain low unless register 0x60 is read; reading register 0x60 sets the INT pin high. The MIC2_SDET state (shorted or not shorted) can be read via register 0x60 bit 6 at any time.

The flow diagram in Figure 34 summarizes the behavior of the INT pin when Register 0x5E = 0x40.

4.15 Interrupts

The CS42L73 includes an open-drain, active-low interrupt output. The registers “Interrupt Mask Register 1 (Address 5Eh)” on page 122 and “Interrupt Mask Register 2 (Address 5Fh)” on page 122 must be used to unmask any interrupt status bits (registers “Interrupt Status Register 1 (Address 60h)” on page 122 and “In-

terrupt Status Register 2 (Address 61h)" on page 123) that are desired to cause an interrupt. The interrupt pin is either rising-edge or rising-and-falling-edge sensitive to any unmasked interrupt status change event. It will be set low when any of the unmasked status bits change state in the sensitive direction(s) and it will remain low until the status register(s) with the interrupt causing bit(s) is (are) read.

Most status bits are "sticky": If the raw signal feeding the status register bit becomes high, the status register bit remains high, regardless of the raw signal's state, at least until the next status register read is completed. Status bits are implemented as sticky to ensure that transient events are not missed. Reads of the status register facilitate the clearing of the status bits when the raw signals are no longer high.

With little effort, the present state of a sticky status signals can optionally be determined. Any read indicating a low level is assured to be the present state. If a high is read, reading the status register again in quick succession will promptly provide the present, non-sticky state of the status signal.

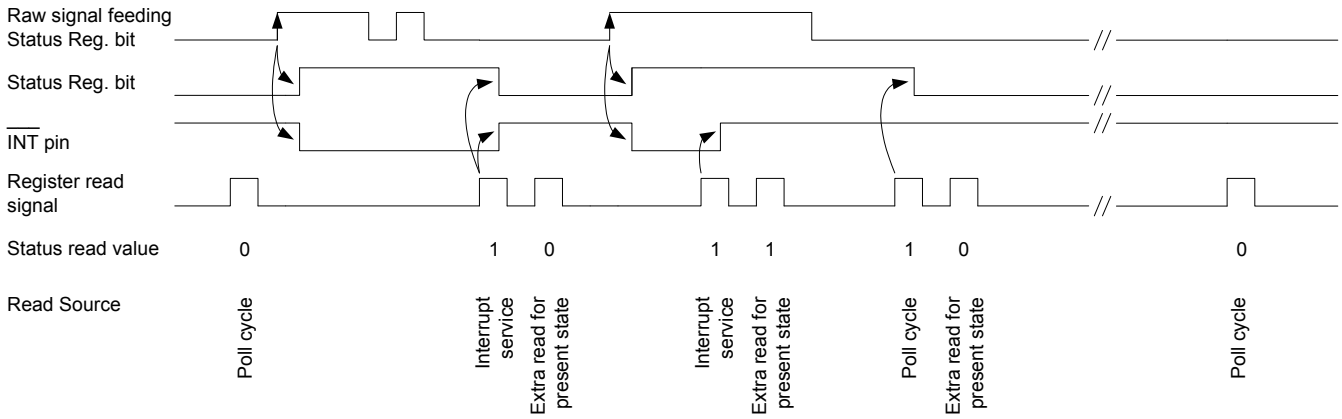


Figure 36. Example of Rising-Edge Sensitive, Sticky, Interrupt Status Bit Behavior

4.16 Control Port Operation

The control port is used to access the registers allowing the CODEC to be configured for the desired operational modes and formats. The operation of the control port may be completely asynchronous with respect to the audio sample rates. However, to avoid potential interference problems, the control port pins must remain static if no operation is required.

The control port operates using an I²C interface with the CODEC acting as a slave device. Device communication must not begin until the reset and power-up timing requirements specified in tables "Switching Specifications—Power, Reset, and Master Clocks" on page 36 and "Switching Specifications—Control Port" on page 40 are satisfied.

Note: The MCLK signal is not required for I²C communication with the CS42L73. However, an MCLK signal is required to be present for the programmed registers to take effect; this is because the state machines affected by register settings cannot be operated without an MCLK signal.

4.16.1 I²C Control

SDA is a bidirectional data line. Data is clocked into and out of the CS42L73 by the clock, SCL. The signal timings for read and write cycles are shown in Figures 37, 38, and 39. A Start condition is defined as a falling transition of SDA while the clock is high. A Stop condition is defined as a rising transition of SDA while the clock is high. All other transitions of SDA occur while the clock is low.

The first byte sent to the CS42L73 after a Start condition consists of a 7-bit chip address field and a R/$\overline{W}$ bit (high for a read, low for a write) in the LSB. To communicate with the CS42L73, the chip address field, must match 1001010b.

If the operation is a write, the next byte is the Memory Address Pointer (MAP); the 7 LSBs of the MAP byte select the address of the register to be read or written to next. The MSB of the MAP byte, INCR, selects whether autoincrementing is to be used (INCR = 1), allowing successive reads or writes of consecutive registers.

Each byte is separated by an acknowledge bit. The ACK bit is output from the CS42L73 after each input byte is read and is input to the CS42L73 from the microcontroller after each transmitted byte.

If the operation is a write, the bytes following the MAP byte will be written to the CS42L73 register addresses pointed to by the last received MAP address, plus however many autoincrements have occurred. Figure 37 illustrates a write pattern with autoincrementing.

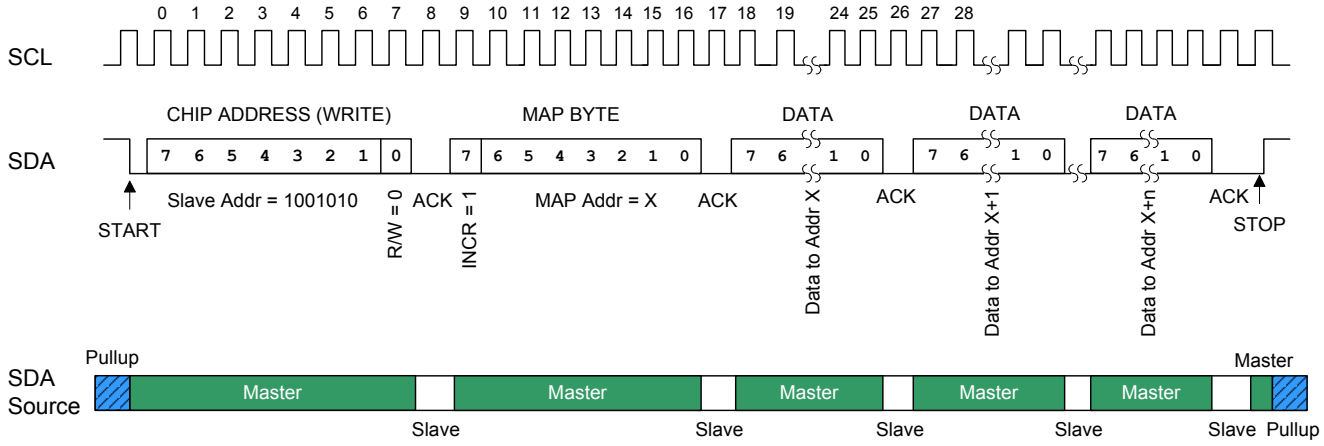


Figure 37. Control Port Timing, I²C Writes with Autoincrement

If the operation is a read, the contents of the register pointed to by the last received MAP address, plus however many autoincrements have occurred, will be output in the next byte. Figure 38 illustrates a read pattern following the write pattern in Figure 37. Notice how the read addresses are based on the MAP byte from Figure 37.

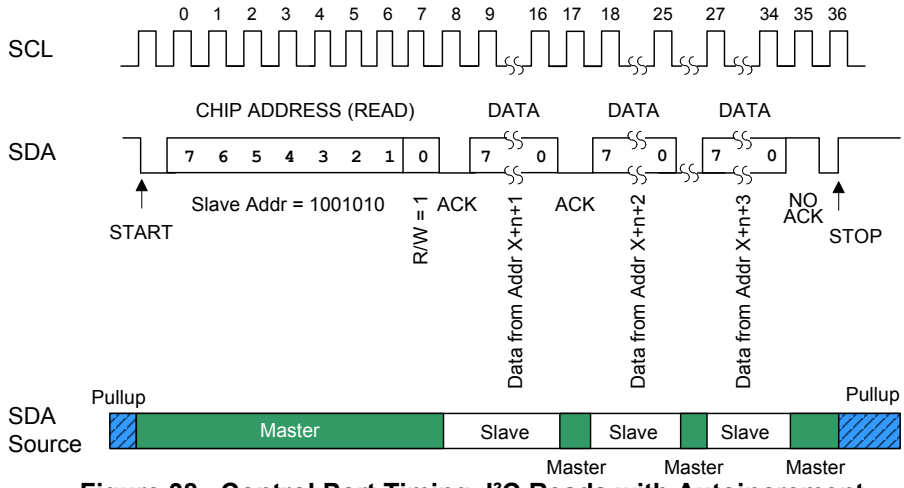


Figure 38. Control Port Timing, I²C Reads with Autoincrement

If a read address different from that which is based on the last received MAP address is desired, an aborted write operation can be used as a preamble that sets the desired read address. This preamble technique is illustrated in Figure 39. In the figure, a write operation is aborted (after the acknowledge for the MAP byte) by sending a stop condition.

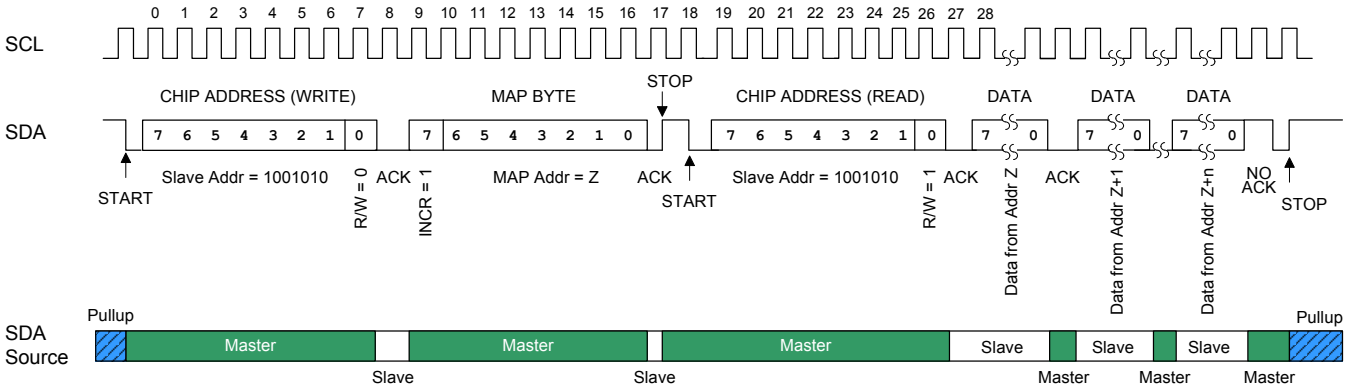


Figure 39. Control Port Timing, I²C Reads with Preamble and Autoincrement

The following pseudocode illustrates an aborted write operation followed by a single read operation. For multiple read operations, autoincrement would be set on (as is shown in Figure 39).

Send start condition.
Send 10010100 (chip address and write operation).
Receive acknowledge bit.
Send MAP byte, autoincrement off.
Receive acknowledge bit.
Send stop condition, aborting write.
Send start condition.
Send 10010101 (chip address and read operation).
Receive acknowledge bit.
Receive byte, contents of selected register.
Send acknowledge bit.
Send stop condition.

4.17 Fast Start Mode

Using fast start mode can reduce the transition time from a low power state to producing audio in the default power-up sequence ("Power-Up Sequence (xSP to HP/LO)" on page 66) to meet stricter requirements. See Table 13 for typical power up times for normal mode and fast start mode. See "Startup Times" on page 133 for setup conditions.

Table 13. Start Up Times

Output Path	Normal Mode	Fast Start Mode	Unit
HPOUT/LINEOUT	70	30	ms
EAR SPKOUT	46	35	ms
SPKOUT/SPKLINEOUT	140	45	ms

Also, a system may want the CS42L73 to be in a low power state but still needs the $\overline{\text{MIC2\_SDET}}$ microphone button short detection to function as a system wake feature. In this case, reducing the MCLK frequency will help to lower the power consumption without affecting audio performance since all of the ADCs and DACs are powered down. To support this power state, fast start mode needs to be enabled to properly detect microphone button presses with a slow MCLK frequency.

To use fast start mode, a set of registers must be written in a certain sequence. To enable fast start mode, perform the following sequence of register writes:

1. Register 00h = 99h
2. Register 7Eh = 81h
3. Register 7Fh = 01h
4. Register 00h = 00h

To disable fast start mode, perform the following sequence of register writes:

1. Register 00h = 99h
2. Register 7Eh = 81h
3. Register 7Fh = 00h
4. Register 00h = 00h

To use fast start mode to reduce the power up time, write the enable sequence prior to step 4 in the default power up sequence ("Power-Up Sequence (xSP to HP/LO)" on page 66). After the power up sequence is completed, write the disable sequence.

To use fast start mode for microphone button short detection with a slow MCLK frequency, write the enable sequence while PDN=1, then clear the PDN and PDN_MIC2_BIAS bits. Make sure all ADCs and DACs are powered down.

The following paragraphs describe behavior when using fast start mode.

The speakerphone output, speakerphone line output, and ear speakerphone output paths create an audible pop during power up if fast start mode is enabled. To avoid hearing this pop on the speaker line output path, mute the external speaker amplifier connected to the CS42L73 during the period when the pop occurs, and then unmute the external amplifier afterwards. Figure 40 shows the pop behavior for fast start mode and the recommended mute control for the external amplifier.

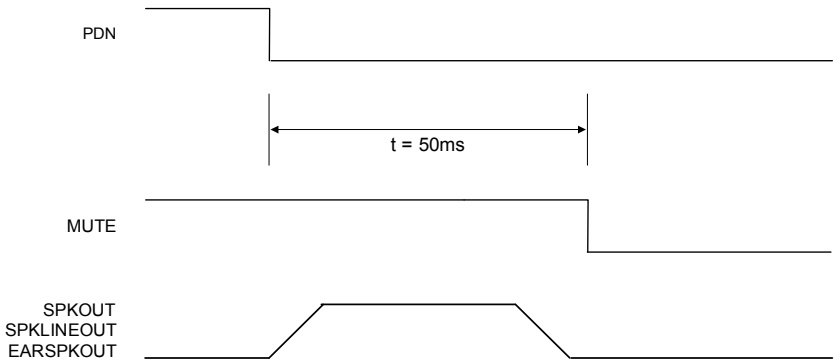


Figure 40. Fast Start Pop

Also, in fast start mode, bias voltages have slower ramp-up times than normal, which may affect the audio quality (reduced volume and increased clipping) while the device powers up. This behavior is seen only on the speakerphone output, speakerphone line output, and ear speakerphone output paths. Figure 41 and Table 14 show the approximate time periods when the output audio may be affected.

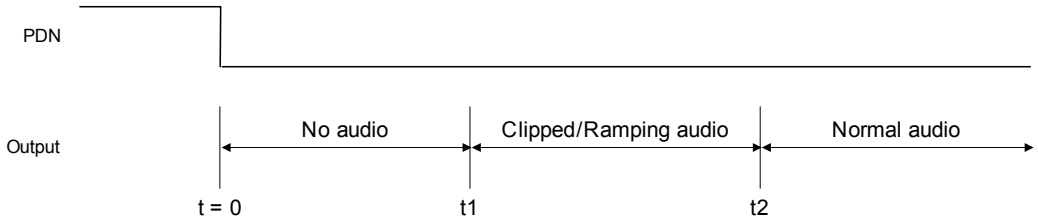


Figure 41. Start Up Transition Diagram

Table 14. Start Up Transition Values

Output Path	t1		t2		Unit
	Normal Mode	Fast Start Mode	Normal Mode	Fast Start Mode	
HPOUT/LINEOUT	65	25	70	30	ms
EAR SPKOUT	40	35	46	180	ms
SPKLINEOUT	140	45	500	500	ms
SPKOUT	140	45	700	700	ms

4.18 Headphone High-Impedance Mode

During normal operation, the headphone output pins are driven by the CS42L73 with low impedance drivers to support low impedance loads. While the headphone amplifier is powered down, the headphone output pins are clamped to ground to prevent undesired transients. Systems that support headset jack–type detection with an external circuit may require the headphone output pins to be in a high-impedance state to properly detect the connected headset type. To accommodate this case, the headphone output pins can be placed into a high-impedance state while the headphone amplifier is powered down. Once the headset type detection has completed, the headphone output pins should be returned to the normal-impedance mode.

To enable high-impedance mode, perform the following sequence of register writes:

1. Power down the headphone amplifier by following Section 4.12.3 through Step 3.
2. Register 00h = 99h
3. Register 7Eh = 96h
4. Register 7Fh = 95h
5. Register 00h = 00h

To disable high-impedance mode, perform the following sequence of register writes:

1. Register 00h = 99h
2. Register 7Eh = 96h
3. Register 7Fh = 94h
4. Register 00h = 00h

5. REGISTER QUICK REFERENCE

(Default values are shown below the bit names)

I²C Address: 1001010[R/W]—10010100 = 0x94(Write); 10010101 = 0x95(Read)									
Adr.	**Function**	**7**	**6**	**5**	**4**	**3**	**2**	**1**	**0**
00h p 81	Fast Mode Enable.	FM_EN7 0	FM_EN6 0	FM_EN5 0	FM_EN4 0	FM_EN3 0	FM_EN2 0	FM_EN1 0	FM_EN0 0
01h p 81	Device ID A and B *(Read Only)*.	DEVIDA3 0	DEVIDA2 1	DEVIDA1 0	DEVIDA0 0	DEVIDB3 0	DEVIDB2 0	DEVIDB1 1	DEVIDB0 0
02h p 81	Device ID C and D *(Read Only)*.	DEVIDC3 1	DEVIDC2 0	DEVIDC1 1	DEVIDC0 0	DEVIDD3 0	DEVIDD2 1	DEVIDD1 1	DEVIDD0 1
03h p 81	Device ID E *(Read Only)*.	DEVIDE3 0	DEVIDE2 0	DEVIDE1 1	DEVIDE0 1	Reserved 0	Reserved 0	Reserved 0	Reserved 0
04h -	Reserved.	Reserved 0	Reserved 0	Reserved 0	Reserved 0	Reserved x	Reserved x	Reserved x	Reserved x
05h p 81	Rev ID *(Read Only)*.	AREVID3 x	AREVID2 x	AREVID1 x	AREVID0 x	MTLREVID3 x	MTLREVID2 x	MTLREVID1 x	MTLREVID0 x
06h p 82	Power Ctl 1.	PDN_ADCB 1	PDN_DMICB 1	PDN_ADCA 1	PDN_DMICA 1	Reserved 0	Reserved 0	DISCHG_FILT 0	PDN 1
07h p 83	Power Ctl 2.	PDN_MIC2_BIAS 1	PDN_MIC1_BIAS 1	Reserved 0	PDN_VSP 1	PDN_ASPSDOUT 1	PDN_ASPSDIN 1	PDN_XSPSDOUT 1	PDN_XSPSDIN 1
08h p 84	Power Ctl 3, Thermal Overload Threshold.	THMOVLD_THLD1 0	THMOVLD_THLD0 0	PDN_THMS 1	PDN_SPKLO 1	PDN_EAR 1	PDN_SPK 1	PDN_LO 1	PDN_HP 1
09h p 85	Charge Pump Freq. and Class H Control.	CHGFREQ3 0	CHGFREQ2 1	CHGFREQ1 0	CHGFREQ0 1	Reserved 0	ADPTPWR2 0	ADPTPWR1 0	ADPTPWR0 0
0Ah p 86	Output Load, Mic Bias, and MIC2 Short Detect Config.	Reserved 0	VP_MIN 1	SPK_LITE_LOAD 0	MIC_BIAS_CTRL 1	SDET_AMUTE 0	Reserved 0	Reserved 1	Reserved 1
0Bh p 87	Digital Mic and Master Clock Control.	DMIC_SCLK_DIV 0	Reserved 0	Reserved 0	MCLKSEL 0	MCLKDIV2 0	MCLKDIV1 0	MCLKDIV0 0	MCLKDIS 0
0Ch p 88	XSP Control.	3ST_XSP 0	XSPDIF 0	X_PCM_MODE1 0	X_PCM_MODE0 0	X_PCM_BIT_ORDER 0	Reserved 0	X_SCK=MCK1 0	X_SCK=MCK0 0
0Dh p 89	XSP Master Mode Clocking Control.	X_M/$\overline{S}$ 0	Reserved 0	X_MMCC5 0	X_MMCC4 1	X_MMCC3 0	X_MMCC2 1	X_MMCC1 0	X_MMCC0 1
0Eh p 89	ASP Control.	3ST_ASP 0	Reserved 0	ASPFS3 0	ASPFS2 0	ASPFS1 0	ASPFS0 0	A_SCK=MCK1 0	A_SCK=MCK0 0
0Fh p 90	ASP Master Mode Clocking Control.	A_M/$\overline{S}$ 0	Reserved 0	A_MMCC5 0	A_MMCC4 1	A_MMCC3 0	A_MMCC2 1	A_MMCC1 0	A_MMCC0 1
10h p 91	VSP Control.	3ST_VSP 0	VSPDIF 0	V_PCM_MODE1 0	V_PCM_MODE0 0	V_PCM_BIT_ORDER 0	V_SDIN_LOC 0	V_SCK=MCK1 0	V_SCK=MCK0 0
11h p 92	VSP Master Mode Clocking Control.	V_M/$\overline{S}$ 0	Reserved 0	V_MMCC5 0	V_MMCC4 1	V_MMCC3 0	V_MMCC2 1	V_MMCC1 0	V_MMCC0 1
12h p 93	VSP and XSP Sample Rate.	VSPFS3 0	VSPFS2 0	VSPFS1 0	VSPFS0 0	XSPFS3 0	XSPFS2 0	XSPFS1 0	XSPFS0 0
13h p 94	Misc. Input and Output Path Control.	D_SWAP_MONO_CTL1 0	D_SWAP_MONO_CTL0 0	IPB=A 0	PGAB=A 0	PGASFT 0	ANLGZC 1	DIGSFT 1	ANLGOSFT 0
14h p 97	ADC/IP Control.	PGABMUX 0	BOOSTB 0	INV_ADCB 0	IPBMUTE 0	PGAAMUX 0	BOOSTA 0	INV_ADCA 0	IPAMUTE 0

I²C Address: 1001010[R/W]—10010100 = 0x94(Write); 10010101 = 0x95(Read)									
Adr.	Function	7	6	5	4	3	2	1	0
15h p 98	Mic 1 [A] Pre-Amp, PGAA Vol.	Reserved 0	MIC_ PREAMPA 0	PGAAVOL5 0	PGAAVOL4 0	PGAAVOL3 0	PGAAVOL2 0	PGAAVOL1 0	PGAAVOL0 0
16h p 98	Mic 2 [B] Pre-Amp, PGAA Vol.	Reserved 0	MIC_ PREAMPB 0	PGABVOL5 0	PGABVOL4 0	PGABVOL3 0	PGABVOL2 0	PGABVOL1 0	PGABVOL0 0
17h p 99	Input Path A Digital Volume.	IPADVOL7 0	IPADVOL6 0	IPADVOL5 0	IPADVOL4 0	IPADVOL3 0	IPADVOL2 0	IPADVOL1 0	IPADVOL0 0
18h p 99	Input Path B Digital Volume.	IPBDVOL7 0	IPBDVOL6 0	IPBDVOL5 0	IPBDVOL4 0	IPBDVOL3 0	IPBDVOL2 0	IPBDVOL1 0	IPBDVOL0 0
19h p 100	Playback Digital Control.	SES_ PLYBCKB=A 0	HL_ PLYBCKB=A 0	LIMSRDIS 0	Reserved 0	ESLDMUTE 0	SPKDMUTE 0	HLBDMUTE 0	HLADMUTE 0
1Ah p 101	Headphone/Line A Out Digital Vol.	HLADVOL7 0	HLADVOL6 0	HLADVOL5 0	HLADVOL4 0	HLADVOL3 0	HLADVOL2 0	HLADVOL1 0	HLADVOL0 0
1Bh p 101	Headphone/Line B Out Digital Vol.	HLBDVOL7 0	HLBDVOL6 0	HLBDVOL5 0	HLBDVOL4 0	HLBDVOL3 0	HLBDVOL2 0	HLBDVOL1 0	HLBDVOL0 0
1Ch p 102	Speakerphone Out [A] Digital Vol .	SPKDVOL7 0	SPKDVOL6 0	SPKDVOL5 0	SPKDVOL4 0	SPKDVOL3 0	SPKDVOL2 0	SPKDVOL1 0	SPKDVOL0 0
1Dh p 102	Ear/Speaker-phone-Line Out [B] Digital Vol.	ESLDVOL7 0	ESLDVOL6 0	ESLDVOL5 0	ESLDVOL4 0	ESLDVOL3 0	ESLDVOL2 0	ESLDVOL1 0	ESLDVOL0 0
1Eh p 103	Headphone A Analog Volume.	HPAAMUTE 0	HPAAVOL6 0	HPAAVOL5 0	HPAAVOL4 0	HPAAVOL3 0	HPAAVOL2 0	HPAAVOL1 0	HPAAVOL0 0
1Fh p 103	Headphone B Analog Volume.	HPBAMUTE 0	HPBAVOL6 0	HPBAVOL5 0	HPBAVOL4 0	HPBAVOL3 0	HPBAVOL2 0	HPBAVOL1 0	HPBAVOL0 0
20h p 104	Line Out A Analog Volume.	LOAAMUTE 0	LOAAVOL6 0	LOAAVOL5 0	LOAAVOL4 0	LOAAVOL3 0	LOAAVOL2 0	LOAAVOL1 0	LOAAVOL0 0
21h p 104	Line Out B Analog Volume.	LOBAMUTE 0	LOBAVOL6 0	LOBAVOL5 0	LOBAVOL4 0	LOBAVOL3 0	LOBAVOL2 0	LOBAVOL1 0	LOBAVOL0 0
22h p 105	Stereo Input Path Adv. Vol.	STRINV7 0	STRINV6 0	STRINV5 0	STRINV4 0	STRINV3 0	STRINV2 0	STRINV1 0	STRINV0 0
23h p 105	Auxiliary Serial Port Input Advisory Vol.	XSPINV7 0	XSPINV6 0	XSPINV5 0	XSPINV4 0	XSPINV3 0	XSPINV2 0	XSPINV1 0	XSPINV0 0
24h p 106	Audio Serial Port Input Advisory Vol.	ASPINV7 0	ASPINV6 0	ASPINV5 0	ASPINV4 0	ASPINV3 0	ASPINV2 0	ASPINV1 0	ASPINV0 0
25h p 106	Voice Serial Port Input Advisory Vol.	VSPINV7 0	VSPINV6 0	VSPINV5 0	VSPINV4 0	VSPINV3 0	VSPINV2 0	VSPINV1 0	VSPINV0 0
26h p 107	Limiter Attack Rate Headphone/Line.	Reserved 0	Reserved 0	LIMARATEHL5 0	LIMARATEHL4 0	LIMARATEHL3 0	LIMARATEHL2 0	LIMARATEHL1 0	LIMARATEHL0 0
27h p 107	Limiter Ctl, Rel. Rate Headphone/Line.	LIMITHL 0	LIMIT_ALLHL 1	LIMRRATEHL5 1	LIMRRATEHL4 1	LIMRRATEHL3 1	LIMRRATEHL2 1	LIMRRATEHL1 1	LIMRRATEHL0 1
28h p 108	Limiter Thresholds Headphone/Line.	LMAXHL2 0	LMAXHL1 0	LMAXHL0 0	CUSHHL2 0	CUSHHL1 0	CUSHHL0 0	Reserved 0	Reserved 0
29h p 108	Limiter Attack Rate Speakerphone [A].	Reserved 0	Reserved 0	LIMARATESPK5 0	LIMARATESPK4 0	LIMARATESPK3 0	LIMARATESPK2 0	LIMARATESPK1 0	LIMARATESPK0 0
2Ah p 109	Limiter Ctl, Release Rate Speakerph. [A].	LIMITSPK 0	LIMIT_ALLSPK 0	LIMRRATESPK5 1	LIMRRATESPK4 1	LIMRRATESPK3 1	LIMRRATESPK2 1	LIMRRATESPK1 1	LIMRRATESPK0 1

I²C Address: 1001010[R/W]—10010100 = 0x94(Write); 10010101 = 0x95(Read)									
Adr.	**Function**	**7**	**6**	**5**	**4**	**3**	**2**	**1**	**0**
2Bh p 110	Limiter Thresholds Speakerphone [A].	LMAXSPK2 0	LMAXSPK1 0	LMAXSPK0 0	CUSHSPK2 0	CUSHSPK1 0	CUSHSPK0 0	Reserved 0	Reserved 0
2Ch p 111	Limiter Attack Rate Ear/Speakerph.-Line [B].	Reserved 0	Reserved 0	LIMARATEESL5 0	LIMARATEESL4 0	LIMARATEESL3 0	LIMARATEESL2 0	LIMARATEESL1 0	LIMARATEESL0 0
2Dh p 111	Limiter Ctl, Release Rate Ear/Speaker-phone-Line [B].	LIMITESL 0	Reserved 0	LIMRRATEESL5 1	LIMRRATEESL4 1	LIMRRATEESL3 1	LIMRRATEESL2 1	LIMRRATEESL1 1	LIMRRATEESL0 1
2Eh p 112	Limiter Thresholds Ear/Speakerph.-Line [B].	LMAXESL2 0	LMAXESL1 0	LMAXESL0 0	CUSHESL2 0	CUSHESL1 0	CUSHESL0 0	Reserved 0	Reserved 0
2Fh p 113	ALC Enable, Attack Rate AB.	ALCB 0	ALCA 0	ALCARATEAB5 0	ALCARATEAB4 0	ALCARATEAB3 0	ALCARATEAB2 0	ALCARATEAB1 0	ALCARATEAB0 0
30h p 113	ALC Release Rate AB.	Reserved 0	Reserved 0	ALCRRATEAB5 1	ALCRRATEAB4 1	ALCRRATEAB3 1	ALCRRATEAB2 1	ALCRRATEAB1 1	ALCRRATEAB0 1
31h p 114	ALC Thresholds AB.	ALCMAXAB2 0	ALCMAXAB1 0	ALCMAXAB0 0	ALCMINAB2 0	ALCMINAB1 0	ALCMINAB0 0	Reserved 0	Reserved 0
32h p 115	Noise Gate Ctl AB.	NGB 0	NGA 0	NG_BOOSTAB 0	THRESHAB2 0	THRESHAB1 0	THRESHAB0 0	NGDELAYAB1 0	NGDELAYAB0 0
33h p 116	ALC and Noise Gate Misc Ctl.	ALC_AB 0	NG_AB 0	ALCBSRDIS 0	ALCBZCDIS 0	ALCASRDIS 0	ALCAZCDIS 0	Reserved 0	Reserved 0
34h p 117	Mixer Control.	Reserved 0	Reserved 0	VSPO_STEREO 0	XSPO_STEREO 1	MXR_SFTR_EN 1	MXR_STEP2 0	MXR_STEP1 0	MXR_STEP0 0
35h p 118	HP/LO Left Mixer: Input Path Left Atten.	Reserved 0	Reserved 0	HLA_IPA_A5 1	HLA_IPA_A4 1	HLA_IPA_A3 1	HLA_IPA_A2 1	HLA_IPA_A1 1	HLA_IPA_A0 1
36h p 118	HP/LO Right Mixer: Input Path Rt. Atten.	Reserved 0	Reserved 0	HLB_IPB_A5 1	HLB_IPB_A4 1	HLB_IPB_A3 1	HLB_IPB_A2 1	HLB_IPB_A1 1	HLB_IPB_A0 1
37h p 118	HP/LO Left Mixer: XSP Left Attenuation.	Reserved 0	Reserved 0	HLA_XSPA_A5 1	HLA_XSPA_A4 1	HLA_XSPA_A3 1	HLA_XSPA_A2 1	HLA_XSPA_A1 1	HLA_XSPA_A0 1
38h p 118	HP/LO Right Mixer: XSP Rt. Attenuation.	Reserved 0	Reserved 0	HLB_XSPB_A5 1	HLB_XSPB_A4 1	HLB_XSPB_A3 1	HLB_XSPB_A2 1	HLB_XSPB_A1 1	HLB_XSPB_A0 1
39h p 118	HP/LO Left Mixer: ASP Left Attenuation.	Reserved 0	Reserved 0	HLA_ASPA_A5 1	HLA_ASPA_A4 1	HLA_ASPA_A3 1	HLA_ASPA_A2 1	HLA_ASPA_A1 1	HLA_ASPA_A0 1
3Ah p 118	HP/LO Right Mixer: ASP Rt. Attenuation.	Reserved 0	Reserved 0	HLB_ASPB_A5 1	HLB_ASPB_A4 1	HLB_ASPB_A3 1	HLB_ASPB_A2 1	HLB_ASPB_A1 1	HLB_ASPB_A0 1
3Bh p 118	HP/LO Left Mixer: VSP Mono Atten.	Reserved 0	Reserved 0	HLA_VSPM_A5 1	HLA_VSPM_A4 1	HLA_VSPM_A3 1	HLA_VSPM_A2 1	HLA_VSPM_A1 1	HLA_VSPM_A0 1
3Ch p 118	HP/LO Right Mixer: VSP Mono Atten.	Reserved 0	Reserved 0	HLB_VSPM_A5 1	HLB_VSPM_A4 1	HLB_VSPM_A3 1	HLB_VSPM_A2 1	HLB_VSPM_A1 1	HLB_VSPM_A0 1
3Dh p 118	XSP Left Mixer: Input Path Left Attenuation.	Reserved 0	Reserved 0	XSPA_IPA_A5 1	XSPA_IPA_A4 1	XSPA_IPA_A3 1	XSPA_IPA_A2 1	XSPA_IPA_A1 1	XSPA_IPA_A0 1
3Eh p 118	XSP Rt. Mixer: Input Path Right Attenuation.	Reserved 0	Reserved 0	XSPB_IPB_A5 1	XSPB_IPB_A4 1	XSPB_IPB_A3 1	XSPB_IPB_A2 1	XSPB_IPB_A1 1	XSPB_IPB_A0 1
3Fh p 118	XSP Left Mixer: XSP Left Attenuation.	Reserved 0	Reserved 0	XSPA_XSPA_A5 1	XSPA_XSPA_A4 1	XSPA_XSPA€_A3 1	XSPA_XSPA_A2 1	XSPA_XSPA_A1 1	XSPA_XSPA_A0 1

I²C Address: 1001010[R/W]—10010100 = 0x94(Write); 10010101 = 0x95(Read)									
Adr.	Function	7	6	5	4	3	2	1	0
40h p 118	XSP Rt. Mixer: XSP Right Attenuation.	Reserved 0	Reserved 0	XSPB_XSPB_A5 1	XSPB_XSPB_A4 1	XSPB_XSPB_A3 1	XSPB_XSPB_A2 1	XSPB_XSPB_A1 1	XSPB_XSPB_A0 1
41h p 118	XSP Left Mixer: ASP Left Attenuation.	Reserved 0	Reserved 0	XSPA_ASPA_A5 1	XSPA_ASPA_A4 1	XSPA_ASPA_A3 1	XSPA_ASPA_A2 1	XSPA_ASPA_A1 1	XSPA_ASPA_A0 1
42h p 118	XSP Rt. Mixer: ASP Right Attenuation.	Reserved 0	Reserved 0	XSPB_ASPB_A5 1	XSPB_ASPB_A4 1	XSPB_ASPB_A3 1	XSPB_ASPB_A2 1	XSPB_ASPB_A1 1	XSPB_ASPB_A0 1
43h p 118	XSP Left Mixer: VSP Mono Attenuation.	Reserved 0	Reserved 0	XSPA_VSPM_A5 1	XSPA_VSPM_A4 1	XSPA_VSPM_A3 1	XSPA_VSPM_A2 1	XSPA_VSPM_A1 1	XSPA_VSPM_A0 1
44h p 118	XSP Rt. Mixer: VSP Mono Attenuation.	Reserved 0	Reserved 0	XSPB_VSPM_A5 1	XSPB_VSPM_A4 1	XSPB_VSPM_A3 1	XSPB_VSPM_A2 1	XSPB_VSPM_A1 1	XSPB_VSPM_A0 1
45h p 118	ASP Left Mixer: Input Path Left Attenuation.	Reserved 0	Reserved 0	ASPA_IPA_A5 1	ASPA_IPA_A4 1	ASPA_IPA_A3 1	ASPA_IPA_A2 1	ASPA_IPA_A1 1	ASPA_IPA_A0 1
46h p 118	ASP Rt. Mixer: Input Path Right Attenuation.	Reserved 0	Reserved 0	ASPB_IPB_A5 1	ASPB_IPB_A4 1	ASPB_IPB_A3 1	ASPB_IPB_A2 1	ASPB_IPB_A1 1	ASPB_IPB_A0 1
47h p 118	ASP Left Mixer: XSP Left Attenuation.	Reserved 0	Reserved 0	ASPA_XSPA_A5 1	ASPA_XSPA_A4 1	ASPA_XSPA_A3 1	ASPA_XSPA_A2 1	ASPA_XSPA_A1 1	ASPA_XSPA_A0 1
48h p 118	ASP Rt. Mixer: XSP Right Attenuation.	Reserved 0	Reserved 0	ASPB_XSPB_A5 1	ASPB_XSPB_A4 1	ASPB_XSPB_A3 1	ASPB_XSPB_A2 1	ASPB_XSPB_A1 1	ASPB_XSPB_A0 1
49h p 118	ASP Left Mixer: ASP Left Attenuation.	Reserved 0	Reserved 0	ASPA_ASPA_A5 1	ASPA_ASPA_A4 1	ASPA_ASPA_A3 1	ASPA_ASPA_A2 1	ASPA_ASPA_A1 1	ASPA_ASPA_A0 1
4Ah p 118	ASP Rt. Mixer: ASP Right Attenuation.	Reserved 0	Reserved 0	ASPB_ASPB_A5 1	ASPB_ASPB_A4 1	ASPB_ASPB_A3 1	ASPB_ASPB_A2 1	ASPB_ASPB_A1 1	ASPB_ASPB_A0 1
4Bh p 118	ASP Left Mixer: VSP Mono Attenuation.	Reserved 0	Reserved 0	ASPA_VSPM_A5 1	ASPA_VSPM_A4 1	ASPA_VSPM_A3 1	ASPA_VSPM_A2 1	ASPA_VSPM_A1 1	ASPA_VSPM_A0 1
4Ch p 118	ASP Rt. Mixer: VSP Mono Attenuation.	Reserved 0	Reserved 0	ASPB_VSPM_A5 1	ASPB_VSPM_A4 1	ASPB_VSPM_A3 1	ASPB_VSPM_A2 1	ASPB_VSPM_A1 1	ASPB_VSPM_A0 1
4Dh p 118	VSP Left Mixer: Input Path Left Attenuation.	Reserved 0	Reserved 0	VSPA_IPA_A5 1	VSPA_IPA_A4 1	VSPA_IPA_A3 1	VSPA_IPA_A2 1	VSPA_IPA_A1 1	VSPA_IPA_A0 1
4Eh p 118	VSP Rt. Mixer: Input Path Right Attenuation.	Reserved 0	Reserved 0	VSPB_IPB_A5 1	VSPB_IPB_A4 1	VSPB_IPB_A3 1	VSPB_IPB_A2 1	VSPB_IPB_A1 1	VSPB_IPB_A0 1
4Fh p 118	VSP Left Mixer: XSP Left Attenuation.	Reserved 0	Reserved 0	VSPA_XSPA_A5 1	VSPA_XSPA_A4 1	VSPA_XSPA_A3 1	VSPA_XSPA_A2 1	VSPA_XSPA_A1 1	VSPA_XSPA_A0 1
50h p 118	VSP Rt. Mixer: XSP Right Attenuation.	Reserved 0	Reserved 0	VSPB_XSPB_A5 1	VSPB_XSPB_A4 1	VSPB_XSPB_A3 1	VSPB_XSPB_A2 1	VSPB_XSPB_A1 1	VSPB_XSPB_A0 1
51h p 118	VSP Left Mixer: ASP Left Attenuation.	Reserved 0	Reserved 0	VSPA_ASPA_A5 1	VSPA_ASPA_A4 1	VSPA_ASPA_A3 1	VSPA_ASPA_A2 1	VSPA_ASPA_A1 1	VSPA_ASPA_A0 1
52h p 118	VSP Rt. Mixer: ASP Right Attenuation.	Reserved 0	Reserved 0	VSPB_ASPB_A5 1	VSPB_ASPB_A4 1	VSPB_ASPB_A3 1	VSPB_ASPB_A2 1	VSPB_ASPB_A1 1	VSPB_ASPB_A0 1
53h p 118	VSP Left Mixer: VSP Mono Attenuation.	Reserved 0	Reserved 0	VSPA_VSPM_A5 1	VSPA_VSPM_A4 1	VSPA_VSPM_A3 1	VSPA_VSPM_A2 1	VSPA_VSPM_A1 1	VSPA_VSPM_A0 1

I²C Address: 1001010[R/W]—10010100 = 0x94(Write); 10010101 = 0x95(Read)									
Adr.	Function	7	6	5	4	3	2	1	0
54h p 118	VSP Rt. Mixer: VSP Mono Attenuation.	Reserved 0	Reserved 0	VSPB_VSPM_A5 1	VSPB_VSPM_A4 1	VSPB_VSPM_A3 1	VSPB_VSPM_A2 1	VSPB_VSPM_A1 1	VSPB_VSPM_A0 1
55h p 120	Mono Mixer Controls.	SPK_ASP_SEL1 1	SPK_ASP_SEL0 0	SPK_XSP_SEL1 1	SPK_XSP_SEL0 0	ESL_ASP_SEL1 1	ESL_ASP_SEL0 0	ESL_XSP_SEL1 1	ESL_XSP_SEL0 0
56h p 121	SPK Mono Mixer: In. Path Mono Atten.	Reserved 0	Reserved 0	SPKM_IPM_A5 1	SPKM_IPM_A4 1	SPKM_IPM_A3 1	SPKM_IPM_A2 1	SPKM_IPM_A1 1	SPKM_IPM_A0 1
57h p 121	SPK Mono Mixer: XSP Mono/L/R Att.	Reserved 0	Reserved 0	SPKM_XSP_A5 1	SPKM_XSP_A4 1	SPKM_XSP_A3 1	SPKM_XSP_A2 1	SPKM_XSP_A1 1	SPKM_XSP_A0 1
58h p 121	SPK Mono Mixer: ASP Mono/L/R Att.	Reserved 0	Reserved 0	SPKM_ASP_A5 1	SPKM_ASP_A4 1	SPKM_ASP_A3 1	SPKM_ASP_A2 1	SPKM_ASP_A1 1	SPKM_ASP_A0 1
59h p 121	SPK Mono Mixer: VSP Mono Atten.	Reserved 0	Reserved 0	SPKM_VSPM_A5 1	SPKM_VSPM_A4 1	SPKM_VSPM_A3 1	SPKM_VSPM_A2 1	SPKM_VSPM_A1 1	SPKM_VSPM_A0 1
5Ah p 121	Ear/SpLO Mono Mixer: In. Path Mono Atten.	Reserved 0	Reserved 0	ESLM_IPM_A5 1	ESLM_IPM_A4 1	ESLM_IPM_A3 1	ESLM_IPM_A2 1	ESLM_IPM_A1 1	ESLM_IPM_A0 1
5Bh p 121	Ear/SpLO Mono Mixer: XSP Mono/L/R Att.	Reserved 0	Reserved 0	ESLM_XSP_A5 1	ESLM_XSP_A4 1	ESLM_XSP_A3 1	ESLM_XSP_A2 1	ESLM_XSP_A1 1	ESLM_XSP_A0 1
5Ch p 121	Ear/SpLO Mono Mixer: ASP Mono/L/R Att.	Reserved 0	Reserved 0	ESLM_ASP_A5 1	ESLM_ASP_A4 1	ESLM_ASP_A3 1	ESLM_ASP_A2 1	ESLM_ASP_A1 1	ESLM_ASP_A0 1
5Dh p 121	Ear/SpLO Mono Mixer: VSP Mono Atten.	Reserved 0	Reserved 0	ESLM_VSPM_A5 1	ESLM_VSPM_A4 1	ESLM_VSPM_A3 1	ESLM_VSPM_A2 1	ESLM_VSPM_A1 1	ESLM_VSPM_A0 1
5Eh p 122	Interrupt Mask 1.	Reserved 0	M_MIC2_SDET 0	Reserved 0	M_THMOVLD 0	M_DIGMIXOVFL 0	Reserved 0	M_IPBOVFL 0	M_IPAOVFL 0
5Fh p 122	Interrupt Mask 2.	Reserved 0	Reserved 0	M_VASRC_DOLK 0	M_VASRC_DILK 0	M_AASRC_DOLK 0	M_AASRC_DILK 0	M_XASRC_DOLK 0	M_XASRC_DILK 0
60h p 122	Interrupt Status 1 *(Read Only)*.	Reserved 0	MIC2_SDET 0	Reserved 0	THMOVLD 0	DIGMIXOVFL 0	Reserved 0	IPBOVFL 0	IPAOVFL 0
61h p 123	Interrupt Status 2 *(Read Only)*.	Reserved 0	Reserved 0	VASRC_DOLK 0	VASRC_DILK 0	AASRC_DOLK 0	AASRC_DILK 0	XASRC_DOLK 0	XASRC_DILK 0
7Eh p 125	Fast Mode 1.	FM15 0	FM14 0	FM13 0	FM12 0	FM11 0	FM10 0	TEST9 0	FM8 0
7Fh p 125	Fast Mode 2.	FM7 0	FM6 0	FM5 0	FM4 0	FM3 0	FM2 0	FM1 0	FM0 0

6. REGISTER DESCRIPTION

Registers are read/write except for chip ID, revision, and status registers, which are read only. The following bit definition tables show bit assignments. The default state of each bit after a power-up sequence or reset is indicated for each bit description via row shading. Reserved registers must maintain their default state.

I²C Address: 1001010[R/W]

6.1 Fast Mode Enable (Address 00h)

7	6	5	4	3	2	1	0
FM_EN7	FM_EN6	FM_EN5	FM_EN4	FM_EN3	FM_EN2	FM_EN1	FM_EN0

6.1.1 Test Bits

See “Fast Start Mode” on page 73.

6.2 Device ID A and B (Address 01h), C and D (Address 02h), and E (Address 03h) *(Read Only)*

7	6	5	4	3	2	1	0
DEVIDA3	DEVIDA2	DEVIDA1	DEVIDA0	DEVIDB3	DEVIDB2	DEVIDB1	DEVIDB0
7	6	5	4	3	2	1	0
DEVIDC3	DEVIDC2	DEVIDC1	DEVIDC0	DEVIDD3	DEVIDD2	DEVIDD1	DEVIDD0
7	6	5	4	3	2	1	0
DEVIDE3	DEVIDE2	DEVIDE1	DEVIDE0	Reserved	Reserved	Reserved	Reserved

6.2.1 Device I.D. (Read Only)

Device I.D. code for the CS42L73.

DEVIDA[3:0]	DEVIDB[3:0]	DEVIDC[3:0]	DEVIDD[3:0]	DEVIDE[3:0]	Part Number
4h	2h	Ah (= L in CS42L73)	7h	3h	CS42L73

6.3 Revision ID (Address 05h) *(Read Only)*

7	6	5	4	3	2	1	0
AREVID3	AREVID2	AREVID1	AREVID0	MTLREVID3	MTLREVID2	MTLREVID1	MTLREVID0

6.3.1 Alpha Revision (Read Only)

CS42L73 alpha revision level.

AREVID[3:0]	Alpha Revision Level
Ah	A
...	...
Fh	F

6.3.2 Metal Revision (Read Only)

CS42L73 numeric revision level.

MTLREVID[3:0]	Metal Revision Level
0h	0
...	...
Fh	F

Note: The Alpha and Metal revision ID form the complete device revision ID. Example: A0, A1, B0, etc.

6.4 Power Control 1 (Address 06h)

7	6	5	4	3	2	1	0
PDN_ADCB	PDN_DMICB	PDN_ADCA	PDN_DMICA	Reserved	Reserved	DISCHG_FILT	PDN

6.4.1 Power Down ADC x

Configures the power state of ADC channel x. All the analog front-end circuitry (PreAmp, PGA, etc.) associated with that channel is powered up or down according to this register bit.

Coupled with the PDN_DMICx controls, these bits also select between the ADC and digital mic inputs and determine the power state of the Input Path digital processing circuitry. Refer to section "Input Paths" on page 59 for more details.

PDN_ADCx	ADC Status
0	Powered Up
1	Powered Down

6.4.2 Power Down Digital Mic x

Coupled with the PDN_ADCx controls, this control selects between the ADC and digital mic inputs and determines the power state of the digital mic interface and the Input Path digital processing circuitry. Refer to sections "Input Paths" on page 59 and "DMIC Interface Powering" on page 60 for more details.

PDN_DMICx	Digital Mic Interface Status
0	Power State as per table "Digital Mic Interface Power States" on page 60
1	

6.4.3 Discharge Filt+ Capacitor

Configures the state of the internal clamp on the FILT+ pin.

DISCHG_FILT	FILT+ Status
0	FILT+ is not clamped to ground
1	FILT+ is clamped to ground

Note: This must only be set if PDN = 1b. Discharge time with an external 2.2-µF capacitor on FILT+ is ~10 ms.

6.4.4 Power Down Device

Configures the power state of the entire CS42L73.

PDN	Device Status
0	Powered Up, as per "Power Control 1 (Address 06h)" on page 82, "Power Control 2 (Address 07h)" on page 83, and "Power Control 3 and Thermal Overload Threshold Control (Address 08h)" on page 84
1	Powered Down

Notes:

- After powering up the device (PDN: 1b → 0b), all sub-blocks will cease to ignore their individual power controls (i.e. will be powered according to their power control programming).

6.5 Power Control 2 (Address 07h)

7	6	5	4	3	2	1	0
PDN_MIC2_BIAS	PDN_MIC1_BIAS	Reserved	PDN_VSP	PDN_ASP_SDOUT	PDN_ASP_SDIN	PDN_XSP_SDOUT	PDN_XSP_SDIN

6.5.1 Power Down MICx Bias

Configures the power state of the mic bias output.

PDN_MICx_BIAS	Mic Bias Status
0	Powered Up
1	Powered Down

6.5.2 Power Down VSP

Configures the power state of the VSP.

PDN_VSP	Voice Serial Port Status
0	Powered Up
1	Powered Down
Application:	Refer to section "Power Management" on page 51.

6.5.3 Power Down ASP SDOUT Path

Configures the power state of the ASP SDOUT path.

PDN_ASP_SDOUT	Audio Serial Port SDOUT Status
0	Powered Up
1	Powered Down
Application:	Refer to section "Power Management" on page 51.

6.5.4 Power Down ASP SDIN Path

Configures the power state of the ASP SDIN path.

PDN_ASP_SDIN	Audio Serial Port SDIN Status
0	Powered Up
1	Powered Down
Application:	Refer to section "Power Management" on page 51.

6.5.5 Power Down XSP SDOUT Path

Configures the power state of the XSP SDOUT path.

PDN_XSP_SDOUT	Auxiliary Serial Port SDOUT Status
0	Powered Up
1	Powered Down
Application:	Refer to section "Power Management" on page 51.

6.5.6 Power Down XSP SDIN Path

Configures the power state of the XSP SDIN path.

PDN_XSP_SDIN	Auxiliary Serial Port SDIN Status
0	Powered Up
1	Powered Down
Application:	Refer to section "Power Management" on page 51.

6.6 Power Control 3 and Thermal Overload Threshold Control (Address 08h)

7	6	5	4	3	2	1	0
THMOVLD_THLD1	THMOVLD_THLD0	PDN_THMS	PDN_SPKLO	PDN_EAR	PDN_SPK	PDN_LO	PDN_HP

6.6.1 Thermal Overload Threshold Settings

Configures the threshold temperature level for the Thermal Overload Interrupt Status bit.

THMOVLD_THLD[1:0]	Nominal Threshold Level (°C)
00	Refer to table "Thermal Overload Detect Characteristics" on page 25
01 to 11	
Application:	"Thermal Overload Notification" on page 42

6.6.2 Power Down Thermal Sense

Configures the power state of Thermal Sense circuit.

PDN_THMS	Thermal Sense Status
0	Powered Up
1	Powered Down
Application:	"Thermal Overload Notification" on page 42

6.6.3 Power Down Speakerphone Line Output

Configures the Speakerphone Line Output Driver power state. If the Speakerphone Line Output Driver or Ear Speaker Driver is powered up, the DAC that drives them is powered up; otherwise, it is powered down.

PDN_SPKLO	Speakerphone Line Output Driver Status
0	Powered Up
1	Powered Down

6.6.4 Power Down Ear Speaker

Configures the Ear Speaker Driver power state. If the Speakerphone Line Output Driver or Ear Speaker Driver is powered up, the DAC that drives them is powered up; otherwise, it is powered down.

PDN_EAR	Ear Speaker Driver Status
0	Powered Up
1	Powered Down

6.6.5 Power Down Speakerphone

Configures the power state of the Speakerphone DAC and Driver.

PDN_SPK	Speakerphone DAC and Driver Status
0	Powered Up
1	Powered Down

6.6.6 Power Down Line Output

Configures the Output Driver power state. If the Line Output Driver or Headphone Driver is powered up, the DAC that drives them will be powered up; otherwise, it is powered down.

PDN_LO	Line Output Driver Status
0	Powered Up
1	Powered Down

6.6.7 Power Down Headphone

Configures the Headphone Driver power state. If the Line Output Driver or Headphone Driver is powered up, the DAC that drives them will be powered up; otherwise, it is powered down.

PDN_HP	Headphone Driver Status
0	Powered Up
1	Powered Down

6.7 Charge Pump Frequency and Class H Configuration (Address 09h)

7	6	5	4	3	2	1	0
CHGFREQ3	CHGFREQ2	CHGFREQ1	CHGFREQ0	Reserved	ADPTPWR2	ADPTPWR1	ADPTPWR0

6.7.1 Charge Pump Frequency

Sets the charge pump frequency on FLYN and FLYP.

CHGFREQ[3:0]	N
0000	0
...	
0101	5
...	
1111	15
Formula:	**Frequency** = f_{MCLK} / [4x(N+2)]

Note: The output THD+N performance improves at higher frequencies; power consumption increases at higher frequencies.

6.7.2 Adaptive Power Adjustment

Configures how the power to the Headphone and Line Output amplifiers adapts to the output signal level.

ADPTPWR[2:0]	Power Supply
000	Adapted to volume setting; Voltage level is determined by the sum of the relevant volume settings
001	Fixed. Headphone and Line1&2 Amp supply = ±VCP
010	Fixed. Headphone and Line1&2 Amp supply = ±VCP/2
011	Fixed. Headphone and Line1&2 Amp supply = ±VCP/3
100	Reserved
101	Reserved
110	Reserved
111	Adapted to Signal; Voltage level is dynamically determined by the output signal

6.8 Output Load, Mic Bias, and MIC2 Short Detect Configuration (Address 0Ah)

7	6	5	4	3	2	1	0
Reserved	VP_MIN	SPK_LITE_LOAD	MIC_BIAS_CTRL	SDET_AMUTE	Reserved	Reserved	Reserved

6.8.1 VP Supply Minimum Voltage Setting

Configures the mic bias generation circuitry to accept the VP supply with the specified minimum value.

VP_MIN	VP Supply Minimum Voltage
0	Lower Voltage
1	Higher Voltage

Notes:
- Refer to "Recommended Operating Conditions" on page 19 for definitions of the lower and higher minimum voltages.
- Note 3 on page 19 explains how to use the VP_MIN control.
- See "Mic BIAS Characteristics" on page 26 details how selecting the lower minimum voltage mode reduces the mic biases PSRR.
- If a mic path is active, it is recommended that the path be muted before changing the VP_MIN setting to avoid audible artifacts.

6.8.2 Speakerphone Light Load Mode Enable

Configures the Speakerphone Driver to minimize power consumption. When the CS42L73 Speakerphone output is used as a line driver to a light load, such as an external amplifier, quiescent power consumption is reduced by setting this control.

SPK_LITE_LOAD	Light Load Mode	Speakerphone Loading	Applicable R_L Range
0	Disabled	Heavy	$R_L \leq R_{L(max)}$ with SPK_LITE_LOAD = 0 b
1	Enabled	Light	$R_L \geq R_{L(min)}$ with SPK_LITE_LOAD = 1 b

Note: R_L is defined in spec. table "Serial Port-to-Mono Speakerphone Output Characteristics" on page 31.

6.8.3 Mic Bias Output Control

Sets the mode for the MIC1_BIAS and MIC2_BIAS device outputs.

MIC_BIAS_CTRL	Mic 1 and 2 Bias Status (When Powered Up)
0	Output Voltage as per "Mic BIAS Characteristics" on page 26.
1	

Note: If either PDN or PDN_MICx_BIAS are set to powered down, the MICx_BIAS output will be Hi-Z, regardless of the MIC_BIAS_CTRL setting.

6.8.4 Short Detect Automatic Mute Control

Configures the reaction to MIC2 Short Detect events.

SDET_AMUTE	Reaction To MIC2 Short Detect Events
0	No reaction
1	MIC2 fed Input Path(s) is (are) automatically muted

6.9 Digital Mic and Master Clock Control (Address 0Bh)

7	6	5	4	3	2	1	0
DMIC_SCLK_DIV	Reserved	Reserved	MCLKSEL	MCLKDIV2	MCLKDIV1	MCLKDIV0	MCLKDIS

6.9.1 Digital Mic Shift Clock Divide Ratio

Sets the divide ratio between the internal Master Clock (MCLK) and the digital mic interface shift clock output.

DMIC_SCLK_DIV	DMIC_SCLK Divide Ratio (from MCLK)
0	/2
1	/4

Note: Refer to section "Digital Microphone (DMIC) Interface" on page 60 for a listing of the supported digital mic Interface shift clock rates and their associated programming settings.

6.9.2 Master Clock Source Selection

Selects the clock source for internal converters and core Master Clock (internal MCLK).

MCLKSEL	Internal MCLK source
0	MCLK1
1	MCLK2

6.9.3 Master Clock Divide Ratio

Selects the divide ratio between the selected MCLK source and the internal MCLK.

MCLKDIV[2:0]	MCLK Divide Ratio (from MCLK1 or MCLK2 Input)
000	Divide by 1
001	Reserved
010	Divide by 2
011	Divide by 3
100	Divide by 4
101	Divide by 6
110 to 111	Reserved

Note: Refer to section "Internal Master Clock Generation" on page 42 for a listing of the supported MCLK rates and their associated programming settings.

6.9.4 Master Clock Disable

Configures the state of the internal MCLK signal prior to its fanout to all internal circuitry.

MCLKDIS	MCLK signal into CODEC
0	On
1	Off; Disables the clock tree to save power when the CODEC is powered down.

6.10 XSP Control (Address 0Ch)

7	6	5	4	3	2	1	0
3ST_XSP	XSPDIF	X_PCM_MODE1	X_PCM_MODE0	X_PCM_BIT_ ORDER	Reserved	X_SCK=MCK1	X_SCK=MCK0

6.10.1 Tristate XSP Interface

Determines the state of the XSP drivers.

3ST_XSP	XSP State	
	Slave Mode	Master Mode
0	Serial port clocks are inputs and SDOUT is output	Serial port clocks and SDOUT are outputs
1	Serial port clocks are inputs and SDOUT is HI-Z	Serial port clocks and SDOUT are HI-Z
Application:	Refer to section "High-impedance Mode" on page 52.	

Note: Slave/Master Mode is determined by the XSP Master/Slave Mode bit described on page 89.

6.10.2 XSP Digital Interface Format

Configures the XSP digital interface format.

XSPDIF	XSP Interface Format
0	I²S
1	PCM (must also set X_PCM_MODE[1:0] and X_PCM_BIT_ORDER)
Application:	Refer to section "Formats" on page 54.

6.10.3 XSP PCM Interface Mode

Applicable only if XSPDIF = 1b (PCM Format). Configures the XSP PCM interface mode.

X_PCM_MODE[1:0]	XSP PCM Interface Mode
00	Mode 0
01	Mode 1
10	Mode 2
11	Reserved
Application:	Refer to section "PCM Format" on page 55.

6.10.4 XSP PCM Format Bit Order

Applicable only if XSPDIF = 1b (PCM Format). Configures the order in which the bits are transmitted on XSP_SDOUT and received on XSP_SDIN.

X_PCM_BIT_ORDER	XSP_SDOUT/XSP_SDIN Bit Order
0	MSB to LSB
1	LSB to MSB
Application:	Refer to section "PCM Format" on page 55.

6.10.5 XSP SCLK Source Equals MCLK

Applicable only if XSPDIF = 0b (I²S Format) and X_M/$\overline{S}$ = 1b (Master Mode). Configures the XSP_SCLK signal source and speed.

X_SCK=MCK[1:0]	Output XSP_SCLK Sourcing Mode
00	SCLK ≠ MCLK (SCLK = ~64•Fs) Mode
01	Reserved
10	SCLK = MCLK Mode
11	SCLK = Pre-MCLK Mode
Application:	Refer to section "SCLK = MCLK Modes" on page 53.

6.11 XSP Master Mode Clocking Control (Address 0Dh)

7	6	5	4	3	2	1	0
X_M/S	Reserved	X_MMCC5	X_MMCC4	X_MMCC3	X_MMCC2	X_MMCC1	X_MMCC0

Refer to XSP Master Mode Clocking relevant control bits "XSP SCLK Source Equals MCLK" on page 88.

6.11.1 XSP Master/Slave Mode

Applicable only if XSPDIF = 0b (I²S Format). Configures the XSP clock source (direction).

X_M/$\overline{S}$	Serial Port Clocks
0	Slave (Input)
1	Master (Output)
Application:	Refer to section "Master and Slave Timing" on page 52.

6.11.2 XSP Master Mode Clock Control Dividers

Applicable only if XSPDIF = 0b (I²S Format). Provides the appropriate divide ratios for all supported serial port master mode clock timings.

X_MMCC[5:0]	Master Mode Clock Control Settings
01 0101	Refer to section "Serial Port Sample Rates and Master Mode Settings" on page 53
Others	

6.12 ASP Control (Address 0Eh)

7	6	5	4	3	2	1	0
3ST_ASP	Reserved	ASPFS3	ASPFS2	ASPFS1	ASPFS0	A_SCK=MCK1	A_SCK=MCK0

6.12.1 Tristate ASP Interface

Determines the state of the ASP drivers.

3ST_ASP	ASP State	
	Slave Mode	Master Mode
0	Serial port clocks are inputs and SDOUT is output	Serial port clocks and SDOUT are outputs
1	Serial port clocks are inputs and SDOUT is HI-Z	Serial port clocks and SDOUT are HI-Z
Application:	Refer to section "High-impedance Mode" on page 52.	

Note: Slave/Master Mode is determined by the ASP Master/Slave Mode bit described on page 90.

6.12.2 ASP Sample Rate

Identifies the ASP audio sample rate.

ASPFS[3:0]	Audio Sample Rate for ASP
0000	Don't know
0001	8.00 kHz
0010	11.025 kHz
0011	12.000 kHz
0100	16.000 kHz
0101	22.050 kHz
0110	24.000 kHz
0111	32.000 kHz
1000	44.100 kHz
1001	48.000 kHz
1010 to 1111	Reserved
Application:	Refer to section "Asynchronous Sample Rate Converters (ASRCs)" on page 59.

6.12.3 ASP SCLK Source Equals MCLK

Applicable only if A_M/$\overline{S}$ = 1b (Master Mode). Configures the ASP_SCLK signal source and speed.

A_SCK=MCK[1:0]	Output ASP_SCLK Sourcing Mode
00	SCLK ≠ MCLK (SCLK = ~64•Fs) Mode
01	Reserved
10	SCLK = MCLK Mode
11	SCLK = Pre-MCLK Mode
Application:	Refer to section "SCLK = MCLK Modes" on page 53.

6.13 ASP Master Mode Clocking Control (Address 0Fh)

7	6	5	4	3	2	1	0
A_M/S	Reserved	A_MMCC5	A_MMCC4	A_MMCC3	A_MMCC2	A_MMCC1	A_MMCC0

Also refer to ASP Master Mode Clocking relevant control bits "ASP SCLK Source Equals MCLK" on page 90.

6.13.1 ASP Master/Slave Mode

Configures the ASP clock source (direction).

A_M/$\overline{S}$	Serial Port Clocks
0	Slave (Input)
1	Master (Output)
Application:	Refer to section "Master and Slave Timing" on page 52.

6.13.2 ASP Master Mode Clock Control Dividers

Provides the appropriate divide ratios for all supported serial port master mode clock timings.

A_MMCC[5:0]	Master Mode Clock Control Settings
01 0101	Refer to section "Serial Port Sample Rates and Master Mode Settings" on page 53
Others	

6.14 VSP Control (Address 10h)

7	6	5	4	3	2	1	0
3ST_VSP	VSPDIF	V_PCM_MODE1	V_PCM_MODE0	V_PCM_BIT_ ORDER	V_SDIN_LOC	V_SCK=MCK1	V_SCK=MCK0

6.14.1 Tristate VSP Interface

Determines the state of the VSP drivers.

3ST_VSP	VSP State	
	Slave Mode	Master Mode
0	Serial port clocks are inputs and SDOUT is output	Serial port clocks and SDOUT are outputs
1	Serial port clocks are inputs and SDOUT is HI-Z	Serial port clocks and SDOUT are HI-Z
Application:	Refer to section "High-impedance Mode" on page 52.	

Note: Slave/Master Mode is determined by the register control bit VSP Master/Slave Mode described on page 92.

6.14.2 VSP Digital Interface Format

Configures the XSP digital interface format.

VSPDIF	VSP Interface Format
0	I²S
1	PCM (must also set V_PCM_MODE[1:0] and V_PCM_BIT_ORDER)
Application:	Refer to section "Formats" on page 54.

6.14.3 VSP PCM Interface Mode

Applicable only if VSPDIF = 1b (PCM Format). Configures the VSP PCM interface mode.

V_PCM_MODE[1:0]	VSP PCM Interface Mode
00	Mode 0
01	Mode 1
10	Mode 2
11	Reserved
Application:	Refer to section "PCM Format" on page 55.

6.14.4 VSP PCM Format Bit Order

Applicable only if VSPDIF = 1b (PCM Format). Configures the order in which bits are transmitted on VSP_SDOUT and received on VSP_SDIN.

V_PCM_BIT_ORDER	VSP_SDOUT/VSP_SDIN Bit Order
0	MSB to LSB
1	LSB to MSB
Application:	Refer to section "PCM Format" on page 55.

6.14.5 VSP SDIN Location

Applicable only if VSPDIF = 0b (I²S Format). Indicates if the received mono data is in the left or right portion of the frame.

VSDIN_LOC	Position
0	Left
1	Right
Application:	Refer to section "Mono/Stereo" on page 57.

6.14.6 VSP SCLK Source Equals MCLK

Applicable only if VSPDIF = 0b (I²S Format) and V_M/$\overline{S}$ = 1b (Master Mode). Configures the VSP_SCLK signal source and speed.

V_SCK=MCK[1:0]	Output VSP_SCLK Sourcing Mode
00	SCLK ≠ MCLK (SCLK = ~64•Fs) Mode
01	Reserved
10	SCLK = MCLK Mode
11	SCLK = Pre-MCLK Mode
Application:	Refer to section "SCLK = MCLK Modes" on page 53.

6.15 VSP Master Mode Clocking Control (Address 11h)

7	6	5	4	3	2	1	0
V_M/S	Reserved	V_MMCC5	V_MMCC4	V_MMCC3	V_MMCC2	V_MMCC1	V_MMCC0

Refer to VSP Master Mode Clocking relevant control bits "VSP SCLK Source Equals MCLK" on page 92.

6.15.1 VSP Master/Slave Mode

Applicable only if VSPDIF = 0b (I²S Format). Configures the VSP clock source (direction).

V_M/$\overline{S}$	Serial Port Clocks
0	Slave (Input)
1	Master (Output)
Application:	Refer to section "Master and Slave Timing" on page 52.

6.15.2 VSP Master Mode Clock Control Dividers

Applicable only if VSPDIF = 0b (I²S Format). Provides the appropriate divide ratios for all supported serial port master mode clock timings.

V_MMCC[5:0]	Master Mode Clock Control Settings
01 0101	Refer to section "Serial Port Sample Rates and Master Mode Settings" on page 53
Others	

6.16 VSP and XSP Sample Rate (Address 12h)

7	6	5	4	3	2	1	0
VSPFS3	VSPFS2	VSPFS1	VSPFS0	XSPFS3	XSPFS2	XSPFS1	XSPFS0

6.16.1 VSP Sample Rate

Identifies the VSP audio sample rate.

VSPFS[3:0]	Audio Sample Rate for VSP
0000	Don't know
0001	8.00 kHz
0010	11.025 kHz
0011	12.000 kHz
0100	16.000 kHz
0101	22.050 kHz
0110	24.000 kHz
0111	32.000 kHz
1000	44.100 kHz
1001	48.000 kHz
1010 to 1111	Reserved
Application:	Refer to section "Asynchronous Sample Rate Converters (ASRCs)" on page 59.

6.16.2 XSP Sample Rate

Identifies the XSP audio sample rate.

XSPFS[3:0]	Audio Sample Rate for XSP
0000	Don't know
0001	8.00 kHz
0010	11.025 kHz
0011	12.000 kHz
0100	16.000 kHz
0101	22.050 kHz
0110	24.000 kHz
0111	32.000 kHz
1000	44.100 kHz
1001	48.000 kHz
1010 to 1111	Reserved
Application:	Refer to section "Asynchronous Sample Rate Converters (ASRCs)" on page 59.

6.17 Miscellaneous Input and Output Path Control (Address 13h)

7	6	5	4	3	2	1	0
D_SWAP_MONO_CTL1	D_SWAP_MONO_CTL0	IPB=A	PGAB=A	PGASFT	ANLGZC	DIGSFT	ANLGOSFT

6.17.1 Digital Swap/Mono

Configures transformations on the Input Path A and B channel inputs to the digital mixer. Note that for any of the transformed cases (‘01’, ‘10’, or ‘11’), both ADC/DMIC A and B must be powered up.

D_SWAP_MONO_CTL[1:0]	Transform	Digital Mixer Stereo Input Sources	
		Input A	Input B
00	Not transformed	Input Path A	Input Path B
01	Mono Fanout of Input Path A	Input Path A	Input Path A
10	Mono Fanout of Input Path B	Input Path B	Input Path B
11	Swap of A and B	Input Path B	Input Path A

6.17.2 Input Path Channel B=A

Configures independent or ganged volume control of the mic/line input path. If ganging is enabled, channel B’s volume will be equal to channel A’s, regardless of channel B’s programming (see affected volume controls listed below).

IPB=A	Single Volume Control (Ganging)	Affected Volume Controls
0	Disabled; Independent channel Input Path volume control.	IPBMUTE and BOOSTB (“ADC/Input Path Control (Address 14h)” on page 97) IPBDVOL[7:0] (“Input Path x Digital Volume Control: Channel A (Address 17h) and B (Address 18h)” on page 99)
1	Enabled; Ganged channel input Path volume control. Channel A’s Input Path volume control controls both A and B channels’ volume.	

6.17.3 PREAMP and PGA Channel B=A

Configures independent or ganged volume control of the Preamp and PGA. If ganging is enabled, channel B’s volume will be equal to channel A’s, regardless of channel B’s programming (see affected analog volume controls listed below).

PGAB=A	Single Volume Control (Ganging)	Affected Analog Volume Controls
0	Disabled; Independent channel PGA and Preamp volume control.	PREAMPB[1:0] and PGABVOL[5:0] (“Mic PreAmp and PGA Volume Control: Channel A (Mic 1, Address 15h) and Channel B (Mic 2, Address 16h)” on page 98)
1	Enabled; Ganged channel PGA and Preamp volume control. Channel A’s PGA volume control controls both A and B channels’ PGA volume.	

6.17.4 PGA Soft-Ramp

Configures an incremental volume ramp from the current level to the new level at the specified rate. If PGA Soft-Ramping is enabled (PGASFT = 1b), the effect of changes to the PGA analog volume controls (listed below) are applied progressively (see exceptions below); if disabled, changes are applied all at once.

PGASFT	Volume Changes	Affected Analog Volume Control
0	Abruptly take effect without a soft-ramp	PGAxVOL[5:0] ("PGAx Volume" on page 98)
1	Occur with a soft-ramp	

Ramp Rate:	If (ALC is disabled)
	// [PGA Volume setting is changed via PGAxVOL]
	If (ANLGZC = 0b)
	0.5 dB every 32 Fs cycles
	Else // [ANLGZC = 1b]
	0.5 dB per ANLGZC event
	Else // (ALC is enabled)
	If ((ALC attack soft-ramping is disabled) and (Considering attack ramp-rate))
	If (ANLGZC = 0b)
	Abrupt volume change
	Else // (ANLGZC = 1b)
	Abrupt volume change at next zero cross event
	Else // [ALC attack soft-ramping is enabled] or [Considering release ramp-rate]]
	If (ANLGZC = 0b)
	If (ALC_Rate_Setting < 2))
	0.5 dB every 32 Fs cycles
	Else // (ALC_Rate_Setting > 1)
	0.5 x Fs / (16 * ALC_Rate_Setting + 1)
	Else // [(ANLGZC = 1b)
	0.5 x f_{step} / (16 * ALC_Rate_Setting + 1); f_{step} is the freq. of zero cross events (including time-outs)
	Notes:
	- ALC_Rate_Setting is either ALCARATE or ALCRRATE
	- ALC is disabled via ALCx
	- ALC attack soft-ramping can be disabled via ALCxSRDIS	

Notes:
- Refer to section "Analog Zero Cross" on page 95 for a description of the ANLGZC control.
- This register bit also affects the ALC volume attack and release rates (soft-ramped as a function of Fs or abrupt). ALC attack soft-ramping can be disabled, regardless of this register control bit, via control "ALCx Soft-Ramp Disable" on page 116.

6.17.5 Analog Zero Cross

Configures when the signal-level changes occur for the analog volume controls. If Analog Zero Cross is enabled (ANLGZC = 1b), the effect of changes to the affected analog volume controls (listed below) are delayed to occur quietly at zero crossings; if disabled, changes will not be aligned to zero crosses.

ANLGZC	Volume Changes	Affected Analog Volume Controls
0	Do not occur on a zero crossing	PGAxVOL[5:0] ("PGAx Volume" on page 98) HPxAMUTE ("Headphone x Analog Mute" on page 103) HPxAVOL[6:0] ("Headphone x Analog Volume Control" on page 103) LOxAMUTE ("Line Output x Analog Mute" on page 104) LOxAVOL[6:0] ("Line Output x Analog Volume Control" on page 104)
1	Occur on a zero crossing	

Notes:
- If the signal does not encounter a zero crossing, the requested volume change will occur after a timeout period of 1024 sample periods (approximately 21.3 ms at 48 kHz sample rate).
- The size of the "Volume Change" per zero cross depends on whether soft-ramping is used (refer to sections "PGA Soft-Ramp" on page 95 and "Analog Output Soft Ramp" on

page 96). If soft-ramping is disabled, a single volume change will occur according to the volume control change. If soft ramping is enabled, the volume change is the soft-ramp step size. With zero cross and soft-ramping enabled, with each zero cross, the volume will step until it eventually matches the volume control.

6.17.6 Digital Soft-Ramp

Configures an incremental volume ramp from the present level to the new level, at the specified rate. If Digital Soft-Ramping is enabled (DIGSFT = 1b), the effect of changes to the affected digital volume controls (listed below) is applied progressively over time (see exceptions noted below); if disabled, changes are applied abruptly, all at once.

DIGSFT	Volume Changes	Affected Digital Volume Controls
0	Abruptly take effect without a soft-ramp	IPxMUTE ("Input Path x Digital Mute" on page 97) IPxDVOL[7:0] ("Input Path x Digital Volume Control" on page 99) HLxDMUTE ("Headphone/Line Output (HL) x Digital Mute" on page 101) HLxDVOL[7:0] ("Headphone/Line Output (HL) x Digital Volume Control" on page 101) ESLDMUTE ("Ear Speaker/Speakerphone Line Output Digital Mute" on page 100) ESLDVOL[7:0] ("Ear Speaker/Speakerphone Line Output (ESL) [B] Digital Volume Control" on page 102) SPKDMUTE ("Ear Speaker/Speakerphone Line Output Digital Mute" on page 100) SPKDVOL[7:0] ("Speakerphone Out [A] Digital Volume Control" on page 102)
1	Occur with a soft-ramp	
Soft-Ramp Rate:	1/8 dB every Fs cycle	

Notes:
- This register bit also sets the noise gate mute/unmute volume ramp rate.
- This register bit does not affect the digital mixer's soft ramping. Register "Mixer Soft-Ramp Enable" on page 117 configures the digital mixer's soft ramping.
- This register bit also affects the ALC and Limiter digital volume attack and release rates (soft-ramped at programmed rates or abrupt). The ALC and Limiter Attack soft-ramping can be disabled, regardless of this register control bit, via the override controls "ALCx Soft-Ramp Disable" on page 116 and "Limiter Soft-Ramp Disable" on page 100.

6.17.7 Analog Output Soft Ramp

Configures an incremental volume ramp from the present level to the new level, at the specified rate. If Analog Output Soft-Ramping is enabled (ANLGOSFT = 1b), the effect of changes to the affected analog volume controls (listed below) is applied progressively over time; if disabled, changes will be applied abruptly, all at once.

ANLGOSFT	Volume Changes	Affected Analog Output Volume Controls
0	Abruptly take effect without a soft-ramp	HPxAMUTE ("Headphone x Analog Mute" on page 103) HPxAVOL[6:0] ("Headphone x Analog Volume Control" on page 103) LOxAMUTE ("Line Output x Analog Mute" on page 104) LOxAVOL[6:0] ("Line Output x Analog Volume Control" on page 104)
1	Occur with a soft ramp	
Ramp Rate:	ANLGZC = 0b: 1.0 dB (-50 dB to +12 dB) or 2 dB (-76 dB to -50 dB) every 32 Fs cycles ANLGZC = 1b: 1.0 dB per ANLGZC event	

Notes:
- Refer to section "Analog Zero Cross" on page 95 for a description of the ANLGZC control.

6.18 ADC/Input Path Control (Address 14h)

7	6	5	4	3	2	1	0
PGABMUX	BOOSTB	INV_ADCB	IPBMUTE	PGAAMUX	BOOSTA	INV_ADCA	IPAMUTE

6.18.1 PGA x Input Select

Selects the specified analog input signal into channel x's PGA.

PGAxMUX	Selected Input to PGAA/PGAB
0	LINEINA/LINEINB
1	MIC1/MIC2

Note: For pseudodifferential inputs, the CODEC automatically chooses the respective pseudoground (LINEIN_REF or MIC1_REF, LINEIN_REF or MIC2_REF) for each input selection.

6.18.2 Boost x

Configures a +20 dB digital boost on channel x.

BOOSTx	+20 dB Digital Boost
0	No boost applied
1	+20 dB boost applied

6.18.3 Invert ADCx Signal Polarity

Configures the polarity of the ADC channel x signal.

INV_ADCx	ADCx Signal Polarity
0	Not Inverted
1	Inverted

6.18.4 Input Path x Digital Mute

Configures a digital mute on the volume control for Input Path channel x, overriding the Input Path digital volume setting (IPxDVOL) and the associated ALC volume control.

IPxMUTE	Input Path Mute
0	Not muted
1	Muted

6.19 Mic PreAmp and PGA Volume Control: Channel A (Mic 1, Address 15h) and Channel B (Mic 2, Address 16h)

7	6	5	4	3	2	1	0
Reserved	MIC_PREAMPx	PGAxVOL5	PGAxVOL4	PGAxVOL3	PGAxVOL2	PGAxVOL1	PGAxVOL0

6.19.1 Mic PREAMP x Volume

Sets the gain of the mic preamp on channel x.

MIC_PREAMPx	Volume
0	+10 dB
1	+20 dB

6.19.2 PGAx Volume

Normally, this control sets the attenuation/gain of the PGA on channel x. When the ALC is engaged, it sets the maximum volume.

PGAxVOL[5:0]	Volume
01 1111	12 dB
...	...
01 1000	12 dB
...	...
00 0001	+0.5 dB
00 0000	0 dB
11 1111	-0.5 dB
...	...
11 1010	-3.0 dB (Target setting for 600 mVrms analog input amplitude)
...	...
11 0100	-6.0 dB
...	...
10 0000	-6.0 dB
Step Size:	0.5 dB

Note: The step size may deviate slightly from 0.5 dB. Refer to Figures 42-47 on page 126.

6.20 Input Path x Digital Volume Control: Channel A (Address 17h) and B (Address 18h)

7	6	5	4	3	2	1	0
IPxDVOL7	IPxDVOL6	IPxDVOL5	IPxDVOL4	IPxDVOL3	IPxDVOL2	IPxDVOL1	IPxDVOL0

6.20.1 Input Path x Digital Volume Control

Normally, this control sets the volume of the Input Path signal on channel x. When the ALC is engaged, it sets the maximum volume. Input Path digital mutes (IPxMUTE) override this register control.

IPxDVOL[7:0]	**Volume**
0111 1111	+12 dB
...	...
0000 1100	+12 dB
...	...
0000 0000	0 dB
1111 1111	-1.0 dB
1111 1110	-2.0 dB
...	...
1010 0000	-96.0 dB
...	...
1000 0000	-96.0 dB
Step Size:	1.0 dB

6.21 Playback Digital Control (Address 19h)

7	6	5	4	3	2	1	0
SES_ PLYBCKB=A	HL_PLYBCKB=A	LIMSRDIS	Reserved	ESLDMUTE	SPKDMUTE	HLBDMUTE	HLADMUTE

6.21.1 Speakerphone [A], Ear Speaker/Speakerphone Line Output [B] (SES) Playback Channels B=A

Configures independent or ganged volume control of the stereo playback channels. If ganging is enabled, channel B's volume will be equal to channel A's, regardless of channel B's programming (see affected volume controls listed below).

SES_ PLYBCKB=A	Single Volume Control (Ganging)	Affected Volume Controls
0	Disabled; Independent channel volume control.	ESLDMUTE ("Ear Speaker/Speakerphone Line Output Digital Mute" on page 100) ESLDVOL[7:0] ("Ear Speaker/Speakerphone Line Output (ESL) [B] Digital Volume Control" on page 102)
1	Enabled; Ganged channel volume control. Channel A's volume control controls both A and B channels' volume.	

6.21.2 Headphone/Line Output (HL) Playback Channels B=A

Configures independent or ganged volume control of the stereo playback channels. If ganging is enabled, channel B's volume will be equal to channel A's, regardless of channel B's programming (see affected volume controls listed below).

HL_ PLYBCKB=A	Single Volume Control (Ganging)	Affected Volume Controls
0	Disabled; Independent channel volume control.	HLBDMUTE ("Headphone/Line Output (HL) x Digital Mute" on page 101) HLBDVOL[7:0] ("Headphone/Line Output (HL) x Digital Volume Control" on page 101) HPBAMUTE and HPBAVOL[6:0] ("Headphone Analog Volume Control: Channel A (Address 1Eh) and B (Address 1Fh)" on page 103) LOBAMUTE and LOBAVOL[6:0] ("Line Output Analog Volume Control: Channel A (Address 20h) and B (Address 21h)" on page 104)
1	Enabled; Ganged channel volume control. Channel A's volume control controls both A and B channels' volume.	

6.21.3 Limiter Soft-Ramp Disable

Configures an override of the Limiter Attack soft-ramp setting.

LIMSRDIS	Limiter Soft-Ramp Disable
0	OFF; Limiter Attack Rate is dictated by the DIGSFT ("Digital Soft-Ramp" on page 96) setting
1	ON; Limiter Attack volume changes take effect in one step, regardless of the DIGSFT setting

6.21.4 Ear Speaker/Speakerphone Line Output Digital Mute

Configures a digital mute on the volume control for ear speaker, overriding the Ear Speaker/Speakerphone Line Output digital volume setting (ESLDVOL) and the associated Limiter volume control.

ESLDMUTE	Ear Speaker/Speakerphone Line Output Digital Mute
0	Not muted
1	Muted

6.21.5 Speakerphone Digital Mute

Configures a digital mute on the volume control for speakerphone, overriding the Speakerphone digital volume setting (SPKDVOL) and the associated Limiter volume control.

SPKDMUTE	Speakerphone Digital Mute
0	Not muted
1	Muted

6.21.6 Headphone/Line Output (HL) x Digital Mute

Configures a digital mute on the volume control for the Headphone/Line Output channel x, overriding the Headphone/Line Output digital volume setting (HLxDVOL) and the associated Limiter volume control.

HLxDMUTE	Headphone/Line Output Digital Mute
0	Not muted
1	Muted

6.22 Headphone/Line Output (HL) x Digital Volume Control: Channel A (Address 1Ah) and B (Address 1Bh)

7	6	5	4	3	2	1	0
HLxDVOL7	HLxDVOL6	HLxDVOL5	HLxDVOL4	HLxDVOL3	HLxDVOL2	HLxDVOL1	HLxDVOL0

6.22.1 Headphone/Line Output (HL) x Digital Volume Control

Normally, this control sets the volume of the channel x signal out of the Headphone/Line Output DAC. If the Limiter is engaged, it sets the maximum volume. Headphone/Line Output digital mutes (HLxDMUTE) override this register control.

HLxDVOL[7:0]	Headphone/Line Output Volume
0001 1000	+12.0 dB
…	…
0000 0000	0 dB
1111 1111	-0.5 dB
1111 1110	-1.0 dB
…	…
0011 0100	-102 dB
…	…
0001 1001	-102 dB
Step Size:	0.5 dB

6.23 Speakerphone Out [A] Digital Volume Control (Address 1Ch)

7	6	5	4	3	2	1	0
SPKDVOL7	SPKDVOL6	SPKDVOL5	SPKDVOL4	SPKDVOL3	SPKDVOL2	SPKDVOL1	SPKDVOL0

6.23.1 Speakerphone Out [A] Digital Volume Control

Normally, this control sets the volume of the signal out of the Speakerphone DAC. If the Limiter is engaged, it sets the maximum volume. Speakerphone digital mutes (SPKDMUTE) override this control.

SPKDVOL[7:0]	Speakerphone Volume
0001 1000	+12.0 dB
…	…
0000 0000	0 dB
1111 1111	-0.5 dB
1111 1110	-1.0 dB
…	…
0011 0100	-102 dB
…	…
0001 1001	-102 dB
Step Size:	0.5 dB

6.24 Ear Speaker/Speakerphone Line Output (ESL) [B] Digital Volume Control (Address 1Dh)

7	6	5	4	3	2	1	0
ESLDVOL7	ESLDVOL6	ESLDVOL5	ESLDVOL4	ESLDVOL3	ESLDVOL2	ESLDVOL1	ESLDVOL0

6.24.1 Ear Speaker/Speakerphone Line Output (ESL) [B] Digital Volume Control

Normally, this control sets the volume of the signal out of the Ear Speaker/Speakerphone Line Output DAC. If the Limiter is engaged, it sets the maximum volume. Ear Speaker/Speakerphone Line Output digital mutes (ESLDMUTE) override this register control.

ESLDVOL[7:0]	Ear Speaker/Speakerphone Line Output Volume
0001 1000	+12.0 dB
…	…
0000 0000	0 dB
1111 1111	-0.5 dB
1111 1110	-1.0 dB
…	…
0011 0100	-102 dB
…	…
0001 1001	-102 dB
Step Size:	0.5 dB

6.25 Headphone Analog Volume Control: Channel A (Address 1Eh) and B (Address 1Fh)

7	6	5	4	3	2	1	0
HPxAMUTE	HPxAVOL6	HPxAVOL5	HPxAVOL4	HPxAVOL3	HPxAVOL2	HPxAVOL1	HPxAVOL0

6.25.1 Headphone x Analog Mute

Configures an analog mute on the channel x Headphone (HP) amplifier.

HPxAMUTE	Headphone Amp Mute
0	Not muted
1	Muted

6.25.2 Headphone x Analog Volume Control

Sets the volume of the signal out of the channel x Headphone (HP) amplifier.

HPxAVOL[6:0]	Headphone Volume
0111111	+12 dB
...	...
0001100	+12 dB
...	...
0000001	+1.0 dB
0000000	0 dB
1111111	-1.0 dB
...	...
1001111	-49.0 dB
1001110	-50.0 dB
1001101	-52.0 dB
1001100	-54.0 dB
...	...
1000010	-74.0 dB
1000001	-76.0 dB
1000000	Reserved
Step Size:	1.0 dB (-50 dB to +12 dB) or 2 dB (-76 dB to -50 dB)

Note: The step size may deviate slightly from 1.0 dB (-50 dB to +12 dB) or 2 dB (-76 dB to -50 dB). Refer to Figures 60 through 63 on page 132.

6.26 Line Output Analog Volume Control: Channel A (Address 20h) and B (Address 21h)

7	6	5	4	3	2	1	0
LOxAMUTE	LOxAVOL6	LOxAVOL5	LOxAVOL4	LOxAVOL3	LOxAVOL2	LOxAVOL1	LOxAVOL0

6.26.1 Line Output x Analog Mute

Configures an analog mute on the channel x Line Output (LO) amplifier.

LOxAMUTE	Line Output Amp Mute
0	Not muted
1	Muted

6.26.2 Line Output x Analog Volume Control

Sets the volume of the signal out of the channel x Line Output (LO) amplifier.

LOxAVOL[6:0]	Line Output Volume
0111111	+12 dB
...	...
0001100	+12 dB
...	...
0000001	+1.0 dB
0000000	0 dB
1111111	-1.0 dB
...	...
1001111	-49.0 dB
1001110	-50.0 dB
1001101	-52.0 dB
1001100	-54.0 dB
...	...
1000010	-74.0 dB
1000001	-76.0 dB
1000000	Reserved
Step Size:	1.0 dB (-50 dB to +12 dB) or 2 dB (-76 dB to -50 dB)

Note: The step size may deviate slightly from 1.0 dB (-50 dB to +12 dB) or 2 dB (-76 dB to -50 dB). Refer to Figures 64 through 67 on page 133.

6.27 Stereo Input Path Advisory Volume (Address 22h)

7	6	5	4	3	2	1	0
STRINV7	STRINV6	STRINV5	STRINV4	STRINV3	STRINV2	STRINV1	STRINV0

Register is applicable only if ADPTPWR = 000b (Class-H power adapted to volume setting) (refer to register control description “Adaptive Power Adjustment” on page 85).

6.27.1 Stereo Input Path Advisory Volume

Defines the maximum analog input volume level used by the Class H controller to determine the appropriate supply for the Headphone and Line Output amplifiers.

STRINV[7:0]	Defined Input Volume
0001 1000	Reserved
…	…
0000 0001	Reserved
0000 0000	0 dB
1111 1111	-0.5 dB
1111 1110	-1.0 dB
…	…
0011 0100	-102 dB
…	…
0001 1001	-102 dB
Step Size:	0.5 dB

6.28 XSP Input Advisory Volume (Address 23h)

7	6	5	4	3	2	1	0
XSPINV7	XSPINV6	XSPINV5	XSPINV4	XSPINV3	XSPINV2	XSPINV1	XSPINV0

Register is applicable only if ADPTPWR = 000b (Class-H power adapted to volume setting) (refer to register control description “Adaptive Power Adjustment” on page 85.

6.28.1 XSP Input Advisory Volume

Defines the maximum analog input volume level used by the Class H controller to determine the appropriate supply for the Headphone and Line Output amplifiers.

XSPINV[7:0]	Defined Input Volume
0001 1000	Reserved
…	…
0000 0001	Reserved
0000 0000	0 dB
1111 1111	-0.5 dB
1111 1110	-1.0 dB
…	…
0011 0100	-102 dB
…	…
0001 1001	-102 dB
Step Size:	0.5 dB

6.29 ASP Input Advisory Volume (Address 24h)

7	6	5	4	3	2	1	0
ASPINV7	ASPINV6	ASPINV5	ASPINV4	ASPINV3	ASPINV2	ASPINV1	ASPINV0

Register is applicable only if ADPTPWR = 000b (Class-H power adapted to volume setting) (refer to register control description “Adaptive Power Adjustment” on page 85).

6.29.1 ASP Input Advisory Volume

Defines the maximum analog input volume level used by the Class H controller to determine the appropriate supply for the Headphone and Line Output amplifiers.

ASPINV[7:0]	Defined Input Volume
0001 1000	Reserved
…	…
0000 0001	Reserved
0000 0000	0 dB
1111 1111	-0.5 dB
1111 1110	-1.0 dB
…	…
0011 0100	-102 dB
…	…
0001 1001	-102 dB
Step Size:	0.5 dB

6.30 VSP Input Advisory Volume (Address 25h)

7	6	5	4	3	2	1	0
VSPINV7	VSPINV6	VSPINV5	VSPINV4	VSPINV3	VSPINV2	VSPINV1	VSPINV0

Register is applicable only if ADPTPWR = 000b (Class-H power adapted to volume setting) (refer to register control description “Adaptive Power Adjustment” on page 85.

6.30.1 VSP Input Advisory Volume

Defines the maximum analog input volume level used by the Class H controller to determine the appropriate supply for the Headphone and Line Output amplifiers.

VSPINV[7:0]	Defined Input Volume
0001 1000	Reserved
…	…
0000 0001	Reserved
0000 0000	0 dB
1111 1111	-0.5 dB
1111 1110	-1.0 dB
…	…
0011 0100	-102 dB
…	…
0001 1001	-102 dB
Step Size:	0.5 dB

6.31 Limiter Attack Rate Headphone/Line Output (HL) (Address 26h)

7	6	5	4	3	2	1	0
Reserved	Reserved	LIMARATEHL5	LIMARATEHL4	LIMARATEHL3	LIMARATEHL2	LIMARATEHL1	LIMARATEHL0

6.31.1 Limiter Attack Rate HL

If limiter attack volume changes are configured to occur with a soft-ramp, this register sets the soft-ramp rate by specifying the soft-ramp step period size.

LIMARATEHL[5:0]	Step Period Size
00 0000	1 (Fastest Attack)
…	…
11 1111	1008 (Slowest Attack)
Formula:	**Step Period Size** = MAXIMUM_OF(1, 16 x LIMARATEHL)

6.32 Limiter Control, Release Rate Headphone/Line Output (HL) (Address 27h)

7	6	5	4	3	2	1	0
LIMITHL	LIMIT_ALLHL	LIMRRATEHL5	LIMRRATEHL4	LIMRRATEHL3	LIMRRATEHL2	LIMRRATEHL1	LIMRRATEHL0

6.32.1 Peak Detect and Limiter HL

Configures the peak detect and limiter circuitry.

LIMITHL	Limiter Status
0	Disabled
1	Enabled

6.32.2 Peak Signal Limit All Channels HL

Sets how channels are attenuated when the limiter is enabled.

LIMIT_ALLHL	Limiter action:
0	Apply the necessary attenuation on a specific channel only if the signal amplitudes on *that* specific channel rise above LMAXHL. Remove attenuation on a specific channel only if the signal amplitude on that channel falls below CUSHHL.
1	Apply the necessary attenuation on BOTH channels when the signal amplitude on any ONE channel exceeds LMAXHL. Remove attenuation on BOTH channels only if the signal amplitude on BOTH channels falls below CUSHHL.

6.32.3 Limiter Release Rate HL

If limiter release volume changes are configured to occur with a soft-ramp, this register sets the soft-ramp rate by specifying the soft-ramp step period size.

LIMRRATEHL[5:0]	Step Period Size
00 0000	1 (Fastest Release)
…	…
11 1111	1008 (Slowest Release)
Formula:	**Step Period Size** = MAXIMUM_OF(1, 16 x LIMRRATESHL)

6.33 Limiter Min/Max Thresholds Headphone/Line Output (HL) (Address 28h)

7	6	5	4	3	2	1	0
LMAXHL2	LMAXHL1	LMAXHL0	CUSHHL2	CUSHHL1	CUSHHL0	Reserved	Reserved

6.33.1 Limiter Maximum Threshold HL

Sets the maximum level, below full scale, above which, the Limiter will attack (increase attenuation) until the output signal's level is below this threshold.

LMAXHL[2:0]	Threshold Setting
000	0 dB
001	-3 dB
010	-6 dB
011	-9 dB
100	-12 dB
101	-18 dB
110	-24 dB
111	-30 dB

6.33.2 Limiter Cushion Threshold HL

Sets the minimum level, below full scale, below which, the Limiter will release (remove attenuation) until the output signal's level is above this threshold.

CUSHHL[2:0]	Threshold Setting
000	0 dB
001	-3 dB
010	-6 dB
011	-9 dB
100	-12 dB
101	-18 dB
110	-24 dB
111	-30 dB

Note: This setting is usually set slightly below the LMAXHL threshold.

6.34 Limiter Attack Rate Speakerphone [A] (Address 29h)

7	6	5	4	3	2	1	0
Reserved	Reserved	LIMARATESPK5	LIMARATESPK4	LIMARATESPK3	LIMARATESPK2	LIMARATESPK1	LIMARATESPK0

6.34.1 Limiter Attack Rate Speakerphone [A]

If limiter attack volume changes are configured to occur with a soft ramp, this register sets the soft-ramp rate by specifying the soft-ramp step period size.

LIMARATESPK[5:0]	Step Period Size
00 0000	1 (Fastest Attack)
…	…
11 1111	1008 (Slowest Attack)
Formula:	**Step Period Size** = MAXIMUM_OF(1, 16 x LIMARATESPK)

6.35 Limiter Control, Release Rate Speakerphone [A] (Address 2Ah)

7	6	5	4	3	2	1	0
LIMITSPK	LIMIT_ALLSPK	LIMRRATESPK5	LIMRRATESPK4	LIMRRATESPK3	LIMRRATESPK2	LIMRRATESPK1	LIMRRATESPK0

6.35.1 Peak Detect and Limiter Speakerphone [A]

Configures the peak detect and limiter circuitry.

LIMITSPK	Limiter Status
0	Disabled
1	Enabled

6.35.2 Peak Signal Limit All Channels Speakerphone

Sets how stereo Speakerphone channels (SPK [A] and ESL [B]) are attenuated when the limiter is enabled.

LIMIT_ALLSPK	Limiter action:
0	Apply the necessary attenuation on a specific channel only if the signal amplitudes on *that* specific channel rise above LMAXSPK [for A channel]/LMAXESL [for B channel]. Remove attenuation on a specific channel only if the signal amplitude on *that* specific channel falls below CUSHSPK [for A channel]/CUSHESL [for B channel]. Speakerphone [A channel] Limiter attack and release rates are as per LIMARATESPK and LIMRRATESPK. Ear Speaker/Speakerphone Line Out [B channel] Limiter attack and release rates are as per LIMARATEESL and LIMRRATEESL.
1	Apply necessary attenuation on BOTH channels if signal amplitude on any ONE channel exceeds LMAXSPK. Remove attenuation on BOTH channels only if the signal amplitude on BOTH channels fall below CUSHSPK. Both channels Limiter attack and release rates are as per LIMARATESPK and LIMRRATESPK.

6.35.3 Limiter Release Rate Speakerphone [A]

If limiter release volume changes are configured to occur with a soft ramp, this register sets the soft-ramp rate by specifying the soft-ramp step period size.

LIMRRATESPK[5:0]	Step Period Size
00 0000	1 (Fastest Release)
…	…
11 1111	1008 (Slowest Release)
Formula:	**Step Period Size** = MAXIMUM_OF(1, 16 x LIMRRATESPK)

6.36 Limiter Min/Max Thresholds Speakerphone [A] (Address 2Bh)

7	6	5	4	3	2	1	0
LMAXSPK2	LMAXSPK1	LMAXSPK0	CUSHSPK2	CUSHSPK1	CUSHSPK0	Reserved	Reserved

6.36.1 Limiter Maximum Threshold Speakerphone [A]

Sets the maximum level, below full scale, above which, the Limiter will attack (increase attenuation) until the output signal's level is below this threshold.

LMAXSPK[2:0]	Threshold Setting
000	0 dB
001	-3 dB
010	-6 dB
011	-9 dB
100	-12 dB
101	-18 dB
110	-24 dB
111	-30 dB

6.36.2 Limiter Cushion Threshold Speakerphone [A]

Sets the minimum level, below full scale, below which, the Limiter will release (remove attenuation) until the output signal's level is above this threshold.

CUSHSPK[2:0]	Threshold Setting
000	0 dB
001	-3 dB
010	-6 dB
011	-9 dB
100	-12 dB
101	-18 dB
110	-24 dB
111	-30 dB

Note: This setting is usually set slightly below the LMAXSPK threshold.

6.37 Limiter Attack Rate Ear Speaker/Speakerphone Line Output (ESL) [B] (Address 2Ch)

7	6	5	4	3	2	1	0
Reserved	Reserved	LIMARATEESL5	LIMARATEESL4	LIMARATEESL3	LIMARATEESL2	LIMARATEESL1	LIMARATEESL0

6.37.1 Limiter Attack Rate ESL [B]

If limiter attack volume changes are configured to occur with a soft-ramp, this register sets the soft-ramp rate by specifying the soft-ramp step period size.

LIMARATEESL[5:0]	Step Period Size
00 0000	1 (Fastest Attack)
…	…
11 1111	1008 (Slowest Attack)
Formula:	**Step Period Size** = MAXIMUM_OF(1, 16 x LIMARATEESL)

6.38 Limiter Control, Release Rate Ear Speaker/Speakerphone Line Output (ESL) [B] (Address 2Dh)

7	6	5	4	3	2	1	0
LIMITESL	Reserved	LIMRRATEESL5	LIMRRATEESL4	LIMRRATEESL3	LIMRRATEESL2	LIMRRATEESL1	LIMRRATEESL0

6.38.1 Peak Detect and Limiter ESL [B]

Configures the peak detect and limiter circuitry.

LIMITESL	Limiter Status
0	Disabled
1	Enabled

6.38.2 Limiter Release Rate ESL [B]

If limiter release volume changes are configured to occur with a soft-ramp, this register sets the soft-ramp rate by specifying the soft-ramp step period size.

LIMRRATEESL[5:0]	Step Period Size
00 0000	1 (Fastest Release)
…	…
11 1111	1008 (Slowest Release)
Formula:	**Step Period Size** = MAXIMUM_OF(1, 16 x LIMRRATEESL)

6.39 Limiter Min/Max Thresholds Ear Speaker/Speakerphone Line Output (ESL) [B] (Address 2Eh)

7	6	5	4	3	2	1	0
LMAXESL2	LMAXESL1	LMAXESL0	CUSHESL2	CUSHESL1	CUSHESL0	Reserved	Reserved

6.39.1 Limiter Maximum Threshold ESL [B]

Sets the maximum level, below full scale, above which, the Limiter will attack (increase attenuation) until the output signal's level is below this threshold.

LMAXESL[2:0]	Threshold Setting
000	0 dB
001	-3 dB
010	-6 dB
011	-9 dB
100	-12 dB
101	-18 dB
110	-24 dB
111	-30 dB

6.39.2 Limiter Cushion Threshold ESL [B]

Sets the minimum level, below full scale, below which, the Limiter will release (remove attenuation) until the output signal's level is above this threshold.

CUSHESL[2:0]	Threshold Setting
000	0 dB
001	-3 dB
010	-6 dB
011	-9 dB
100	-12 dB
101	-18 dB
110	-24 dB
111	-30 dB

Note: This setting is usually set slightly below the LMAXESL threshold.

6.40 ALC Enable and Attack Rate AB (Address 2Fh)

7	6	5	4	3	2	1	0
ALCB	ALCA	ALCARATEAB5	ALCARATEAB4	ALCARATEAB3	ALCARATEAB2	ALCARATEAB1	ALCARATEAB0

6.40.1 ALC for Channels A and B (ALCx)

Enables ALC independently for channels A and B if ALC_AB = 0b (refer to "ALC Ganging of Channels A and B" on page 116). If enabled, and if the particular channel's signal amplitude exceeds the maximum threshold setting or falls below the minimum threshold setting, ALC is applied to only that channel. If ganged behavior is desired, set ALC_AB from 0b to 1b and then set ALCA and ALCB from 0b to 1b simultaneously.

ALCx	ALC Status
0	Disabled ALC on channel x
1	Enabled ALC on channel x

6.40.2 ALC Attack Rate for Channels A and B

Sets the rate at which the ALC applies analog and/or digital attenuation from levels above the ALCMAXAB[2:0] threshold ("ALC Maximum Threshold for Channels A and B" on page 114).

ALCARATEAB[5:0]	Attack Time
00 0000	Fastest Attack
…	…
11 1111	Slowest Attack

Note: The ALC attack rate is user selectable, but is also a function of the sampling frequency, Fs, the PGASFT ("PGA Soft-Ramp" on page 95), ANLGZC ("Analog Zero Cross" on page 95), and DIGSFT ("Digital Soft-Ramp" on page 96) settings unless either of the respective disable bits (ALCxSRDIS—refer to "ALCx Soft-Ramp Disable" on page 116 or ALCxZCDIS—refer to "ALCx Zero Cross Disable" on page 116) is enabled.

6.41 ALC Release Rate AB (Address 30h)

7	6	5	4	3	2	1	0
Reserved	Reserved	ALCRRATEAB5	ALCRRATEAB4	ALCRRATEAB3	ALCRRATEAB2	ALCRRATEAB1	ALCRRATEAB0

6.41.1 ALC Release Rate for Channels A and B

Sets the rate at which the ALC releases the analog and/or digital attenuation from levels below the ALCMINAB[2:0] threshold ("ALC Minimum Threshold for Channels A and B" on page 114) and returns the signal level to the PGAxVOL[5:0] ("PGAx Volume" on page 98) and ADCxDVOL[7:0] ("Input Path x Digital Volume Control" on page 99) setting.

ALCRRATEAB[5:0]	Release Time
00 0000	Fastest Release
…	…
11 1111	Slowest Release

Notes:
- The ALC release rate is user selectable, but is also a function of the sampling frequency, Fs, the PGASFT ("PGA Soft-Ramp" on page 95), ANLGZC ("Analog Zero Cross" on page 95), and DIGSFT ("Digital Soft-Ramp" on page 96) settings.
- The Release Rate setting must always be slower than the Attack Rate.

6.42 ALC Threshold AB (Address 31h)

7	6	5	4	3	2	1	0
ALCMAXAB2	ALCMAXAB1	ALCMAXAB0	ALCMINAB2	ALCMINAB1	ALCMINAB0	Reserved	Reserved

6.42.1 ALC Maximum Threshold for Channels A and B

Sets the maximum level, below full scale, at which to limit and attenuate the input signal at the attack rate (ALCARATEAB—"ALC Attack Rate for Channels A and B" on page 113).

ALCMAXAB[2:0]	Threshold Setting
000	0 dB
001	-3 dB
010	-6 dB
...	...
111	-30 dB

6.42.2 ALC Minimum Threshold for Channels A and B

Sets the minimum level at which to disengage the ALC's attenuation or amplify the input signal at the release rate (ALCRRATEAB—"ALC Release Rate for Channels A and B" on page 113) until levels lie between the ALCMAXAB and ALCMINAB thresholds.

ALCMINAB[2:0]	Threshold Setting
000	0 dB
001	-3 dB
010	-6 dB
...	...
111	-30 dB

Note: This setting is usually set slightly below the ALCMAXAB threshold.

6.43 Noise Gate Control AB (Address 32h)

7	6	5	4	3	2	1	0
NGB	NGA	NG_BOOSTAB	THRESHAB2	THRESHAB1	THRESHAB0	NGDELAYAB1	NGDELAYAB0

6.43.1 Noise Gate Enable for Channels A and B (NGx)

Enables noise gating independently for channels A and B if NG_AB = 0b. When enabled, if the particular channel's signal amplitude is continuously below the threshold setting (see "Noise gate Threshold and Boost for Channels A and B" on page 115) for longer than the attack delay (debounce) time (see "Noise Gate Delay Timing for Channels A and B" on page 115), noise gate muting is applied to only that channel.

Noise gate muting is removed (released) without debouncing if the signal level exceeds the threshold. This bit has no effect if NG_AB = 1b (refer to "Noise Gate Ganging of Channels A and B" on page 116).

NGx	Noise Gate Status (NG_AB = 0b)
0	Disable noise gating on channel x
1	Enable noise gating on channel x

Note: Noise gate attack and release rates (soft-ramped as a function of Fs or abrupt) are set according to the programming of DIGSFT ("Digital Soft-Ramp" on page 96).

6.43.2 Noise gate Threshold and Boost for Channels A and B

THRESHAB sets threshold level (±2 dB) for the A and B channel noise gates. NG_BOOSTAB configures a +30 dB boost to the threshold setting.

THRESHAB[2:0]	Minimum Setting (NG_BOOSTAB = 0b)	Minimum Setting (NG_BOOSTAB = 1b)
000	-64 dB	-34 dB
001	-66 dB	-36 dB
010	-70 dB	-40 dB
011	-73 dB	-43 dB
100	-76 dB	-46 dB
101	-82 dB	-52 dB
110	Reserved	-58 dB
111	Reserved	-64 dB

Note: The combined threshold and boost setting defines the signal level where the noise gate begins to engage. For low settings, the noise gate may not fully engage until the signal level is a few dB lower.

6.43.3 Noise Gate Delay Timing for Channels A and B

Sets the delay (debounce) time before the noise gate mute attacks.

NGDELAYAB[1:0]	Delay Setting
00	50 ms
01	100 ms
10	150 ms
11	200 ms

6.44 ALC and Noise Gate Misc Control (Address 33h)

7	6	5	4	3	2	1	0
ALC_AB	NG_AB	ALCBSRDIS	ALCBZCDIS	ALCASRDIS	ALCAZCDIS	Reserved	Reserved

6.44.1 ALC Ganging of Channels A and B

Configures whether ALC for channels A and B is independent (see "ALC for Channels A and B (ALCx)" on page 113) or ganged. If ganged, ALC is applied equally to channels A and B and is triggered by

- either channel A or B exceeding the ALC AB maximum threshold

  or
- both channels A and B falling below the ALC AB minimum threshold ("ALC Maximum Threshold for Channels A and B" on page 114 and "ALC Minimum Threshold for Channels A and B" on page 114).

ALC_AB	ALC Status
0	Independent ALC on channels A and B
1	Ganged ALC for channels A and B

Note: If ganged behavior is desired, set ALC_AB from 0b to 1b and then set ALCA and ALCB from 0b to 1b simultaneously.

6.44.2 Noise Gate Ganging of Channels A and B

Configures whether noise gating for channels A and B will be independent (refer to "Noise Gate Enable for Channels A and B (NGx)" on page 115) or ganged.

If ganged, noise gate muting is applied to both channels A and B when the signal amplitudes of both channels A and B are continuously below the noise gate AB minimum threshold (refer to "Noise gate Threshold and Boost for Channels A and B" on page 115) for longer than the attack delay (debounce) time (refer to "Noise Gate Delay Timing for Channels A and B" on page 115).

Noise gate muting will be removed (released) without debouncing when either of the A or B signal levels rise above the threshold.

NG_AB	Noise Gate triggered by:
0	Independent noise gating on channels A and B
1	Ganged noise gating on channels A and B

Note: The noise gate attack and release rates (soft-ramped as a function of Fs or abrupt) are set according to the programming of DIGSFT ("Digital Soft-Ramp" on page 96).

6.44.3 ALCx Soft-Ramp Disable

Configures an override of the ALC Attack soft-ramp settings.

ALCxSRDIS	ALC Soft-Ramp Disable
0	OFF; ALC Attack Rate is dictated by the DIGSFT ("Digital Soft-Ramp" on page 96) and the PGASFT ("PGA Soft-Ramp" on page 95) settings
1	ON; ALC Attack volume changes take effect in one step, regardless of the DIGSFT or PGASFT settings

6.44.4 ALCx Zero Cross Disable

Configures an override of the ALC Attack Analog Zero Cross setting.

ALCxZCDIS	ALC Zero Cross Disable
0	OFF; ALC Attack Rate is dictated by the ANLGZC ("Analog Zero Cross" on page 95) setting
1	ON; ALC Attack volume changes take effect at any time, regardless of the ANLGZC setting

6.45 Mixer Control (Address 34h)

7	6	5	4	3	2	1	0
Reserved	Reserved	VSPO_STEREO	XSPO_STEREO	MXR_SFTR_EN	MXR_STEP2	MXR_STEP1	MXR_STEP0

6.45.1 VSP Mixer Output Stereo

Selects which of the following mixer outputs is sent to the VSP output:
- The stereo VSP mixer outputs
- The output from the mixer that combines the stereo VSP mixer's outputs is sent to the left and right channels of the VSP output interface.

VSPO_STEREO	Mixer(s) Selected
0	Mono
1	Stereo
Application:	Refer to section "Mono and Stereo Paths" on page 63.

6.45.2 XSP Mixer Output Stereo

Selects which mixer output is sent to the XSP output:
- The stereo XSP mixer outputs
- The output from the mixer that combines the stereo XSP mixer's outputs is sent to the left and right channels of the XSP output interface.

XSPO_STEREO	Mixer(s) Selected
0	Mono
1	Stereo
Application:	Refer to section "Mono and Stereo Paths" on page 63.

6.45.3 Mixer Soft-Ramp Enable

Selects whether the digital mixer's mixer inputs will change volume all at once or use an incremental volume ramp (configured by "Mixer Soft-Ramp Step Size/Period" on page 117) to change from the current level to the new level, via "Stereo Mixer Input Attenuation (Addresses 35h through 54h)" on page 118.

MXR_SFTR_EN	Mixer Volume Changes
0	Abruptly take effect without a soft-ramp
1	Occur with a soft-ramp
Application:	Refer to section "Mixer Input Attenuation Adjustment" on page 63.

6.45.4 Mixer Soft-Ramp Step Size/Period

Selects the mixer attenuation change soft-ramp step size and step period.

MXR_STEP[2:0]	Step Size	Step Period (# of Fs periods between each step)
000	1/8 dB	1
001	1/4 dB	1
010	1/2 dB	1
011	1 dB	1
100	1/8 dB	4
101	1/8 dB	2
110	Reserved	
111	Reserved	
Application:	Refer to section "Mixer Input Attenuation Adjustment" on page 63.	

6.46 Stereo Mixer Input Attenuation (Addresses 35h through 54h)

Sets the level of attenuation to be applied to various stereo digital mixer inputs. Each mixer input can be muted or attenuated from 0 to 62 dB in 1 dB steps.

7	6	5	4	3	2	1	0
Reserved	Reserved	*_A5	*_A4	*_A3	*_A2	*_A1	*_A0

Note: * indicates the field name, taken from the mixer and attenuation characteristics and listed below..

Register		Mixer		Attenuator		Field Name
Addr.	Name	ID (Destination)	Output Channel	Source	Source Channel	
35h	Headphone/Line Output Left Mixer: Input Path Left Attenuation	Headphone/Line Out (HL)	Left (A)	Input Path (IP)	Left (A)	HLA_IPA_A
36h	Headphone/Line Output Right Mixer: Input Path Right. Attenuation	Headphone/Line Out (HL)	Right (B)	Input Path (IP)	Right (B)	HLB_IPB_A
37h	Headphone/Line Output Left Mixer: XSP Left Attenuation	Headphone/Line Out (HL)	Left (A)	Auxiliary Serial Port (XSP)	Left (A)	HLA_XSPA_A
38h	Headphone/Line Output Right Mixer: XSP Right Attenuation	Headphone/Line Out (HL)	Right (B)	Auxiliary Serial Port (XSP)	Right (B)	HLB_XSPB_A
39h	Headphone/Line Output Left Mixer: ASP Left Attenuation	Headphone/Line Out (HL)	Left (A)	Audio Serial Port (ASP)	Left (A)	HLA_ASPA_A
3Ah	Headphone/Line Output Right Mixer: ASP Right Attenuation	Headphone/Line Out (HL)	Right (B)	Audio Serial Port (ASP)	Right (B)	HLB_ASPB_A
3Bh	Headphone/Line Output Left Mixer: VSP Mono Attenuation	Headphone/Line Out (HL)	Left (A)	Voice Serial Port (VSP)	Mono (M)	HLA_VSPM_A
3Ch	Headphone/Line Output Right Mixer: VSP Mono Attenuation	Headphone/Line Out (HL)	Right (B)	Voice Serial Port (VSP)	Mono (M)	HLB_VSPM_A
3Dh	XSP Left Mixer: Input Path Left Attenuation	Auxiliary Serial Port (XSP)	Left (A)	Input Path (IP)	Left (A)	XSPA_IPA_A
3Eh	XSP Right Mixer: Input Path Right Attenuation	Auxiliary Serial Port (XSP)	Right (B)	Input Path (IP)	Right (B)	XSPB_IPB_A
3Fh	XSP Left Mixer: XSP Left Attenuation	Auxiliary Serial Port (XSP)	Left (A)	Auxiliary Serial Port (XSP)	Left (A)	XSPA_XSPA_A
40h	XSP Right Mixer: XSP Right Attenuation	Auxiliary Serial Port (XSP)	Right (B)	Auxiliary Serial Port (XSP)	Right (B)	XSPB_XSPB_A
41h	XSP Left Mixer: ASP Left Attenuation	Auxiliary Serial Port (XSP)	Left (A)	Audio Serial Port (ASP)	Left (A)	XSPA_ASPA_A
42h	XSP Right Mixer: ASP Right Attenuation	Auxiliary Serial Port (XSP)	Right (B)	Audio Serial Port (ASP)	Right (B)	XSPB_ASPB_A
43h	XSP Left Mixer: VSP Mono Attenuation	Auxiliary Serial Port (XSP)	Left (A)	Voice Serial Port (VSP)	Mono (M)	XSPA_VSPM_A
44h	XSP Right Mixer: VSP Mono Attenuation	Auxiliary Serial Port (XSP)	Right (B)	Voice Serial Port (VSP)	Mono (M)	XSPB_VSPM_A
45h	ASP Left Mixer: Input Path Left Attenuation	Audio Serial Port (ASP)	Left (A)	Input Path (IP)	Left (A)	ASPA_IPA_A
46h	ASP Right Mixer: Input Path Right Attenuation	Audio Serial Port (ASP)	Right (B)	Input Path (IP)	Right (B)	ASPB_IPB_A
47h	ASP Left Mixer: XSP Left Attenuation	Audio Serial Port (ASP)	Left (A)	Auxiliary Serial Port (XSP)	Left (A)	ASPA_XSPA_A
48h	ASP Right Mixer: XSP Right Attenuation	Audio Serial Port (ASP)	Right (B)	Auxiliary Serial Port (XSP)	Right (B)	ASPB_XSPB_A
49h	ASP Left Mixer: ASP Left Attenuation	Audio Serial Port (ASP)	Left (A)	Audio Serial Port (ASP)	Left (A)	ASPA_ASPA_A
4Ah	ASP Right Mixer: ASP Right Attenuation	Audio Serial Port (ASP)	Right (B)	Audio Serial Port (ASP)	Right (B)	ASPB_ASPB_A

Register		Mixer		Attenuator		Field Name
Addr.	Name	ID (Destination)	Output Channel	Source	Source Channel	
4Bh	ASP Left Mixer: VSP Mono Attenuation	Audio Serial Port (ASP)	Left (A)	Voice Serial Port (VSP)	Mono (M)	ASPA_VSPM_A
4Ch	ASP Right Mixer: VSP Mono Attenuation	Audio Serial Port (ASP)	Right (B)	Voice Serial Port (VSP)	Mono (M)	ASPB_VSPM_A
4Dh	VSP Left Mixer: Input Path Left Attenuation	Voice Serial Port (VSP)	Left (A)	Input Path (IP)	Left (A)	VSPA_IPA_A
4Eh	VSP Right Mixer: Input Path Right Attenuation	Voice Serial Port (VSP)	Right (B)	Input Path (IP)	Right (B)	VSPB_IPB_A
4Fh	VSP Left Mixer: XSP Left Attenuation	Voice Serial Port (VSP)	Left (A)	Auxiliary Serial Port (XSP)	Left (A)	VSPA_XSPA_A
50h	VSP Right Mixer: XSP Right Attenuation	Voice Serial Port (VSP)	Right (B)	Auxiliary Serial Port (XSP)	Right (B)	VSPB_XSPB_A
51h	VSP Left Mixer: ASP Left Attenuation	Voice Serial Port (VSP)	Left (A)	Audio Serial Port (ASP)	Left (A)	VSPA_ASPA_A
52h	VSP Right Mixer: ASP Right Attenuation	Voice Serial Port (VSP)	Right (B)	Audio Serial Port (ASP)	Right (B)	VSPB_ASPB_A
53h	VSP Left Mixer: VSP Mono Attenuation	Voice Serial Port (VSP)	Left (A)	Voice Serial Port (VSP)	Mono (M)	VSPA_VSPM_A
54h	VSP Right Mixer: VSP Mono Attenuation	Voice Serial Port (VSP)	Right (B)	Voice Serial Port (VSP)	Mono (M)	VSPB_VSPM_A

6.46.1 Stereo Mixer Input Attenuation

*_A[5:0]	Volume
000000	0 dB
000001	-1.0 dB
000010	-2.0 dB
...	...
111110	-62.0 dB
111111	Mute
Step Size:	1.0 dB
Application:	Refer to sections “Mixer Input Attenuation Adjustment” on page 63 and “Mixer Attenuation Values” on page 65.

6.47 Mono Mixer Controls (Address 55h)

7	6	5	4	3	2	1	0
SPK_ASP_SEL1	SPK_ASP_SEL0	SPK_XSP_SEL1	SPK_XSP_SEL0	ESL_ASP_SEL1	ESL_ASP_SEL0	ESL_XSP_SEL1	ESL_XSP_SEL0

6.47.1 Speakerphone (SPK) Mixer, ASP Select

Selects the input to ASP input attenuator of the SPK mono mixer.

SPK_ASP_SEL[1:0]	Selected Input
00	ASP Input Left Channel
01	ASP Input Right Channel
10	ASP Input Mono Mix
11	
Application:	Refer to section "Mono and Stereo Paths" on page 63.

6.47.2 Speakerphone (SPK) Mixer, XSP Select

Selects the input to XSP input attenuator of the SPK mono mixer.

SPK_XSP_SEL[1:0]	Selected Input
00	XSP Input Left Channel
01	XSP Input Right Channel
10	XSP Input Mono Mix
11	
Application:	Refer to section "Mono and Stereo Paths" on page 63.

6.47.3 Ear Speaker/Speakerphone Line Output (ESL) Mixer, ASP Select

Selects the input to ASP input attenuator of the ESL mono mixer.

ESL_ASP_SEL[1:0]	Selected Input
00	ASP Input Left Channel
01	ASP Input Right Channel
10	ASP Input Mono Mix
11	
Application:	Refer to section "Mono and Stereo Paths" on page 63.

6.47.4 ESL Mixer, Auxiliary Serial Port (XSP) Select

Selects the input to XSP input attenuator of the ESL mono mixer.

ESL_XSP_SEL[1:0]	Selected Input
00	XSP Input Left Channel
01	XSP Input Right Channel
10	XSP Input Mono Mix
11	
Application:	Refer to section "Mono and Stereo Paths" on page 63.

6.48 Mono Mixer Input Attenuation (Addresses 56h through 5Dh)

Sets the level of attenuation to be applied to various mono digital mixer inputs. Each mixer input can be muted or attenuated from 0 to 62 dB in 1 dB steps.

7	6	5	4	3	2	1	0
Reserved	Reserved	*_A5	*_A4	*_A3	*_A2	*_A1	*_A0

Note: * indicates the field name, taken from the mixer and attenuation characteristics and listed below..

Register		Mixer		Attenuator		Field Name
Addr.	Name	ID (Destination)	Output Channel	Source	Source Channel	
56h	Speakerphone Mono Mixer: Input Path Mono Attenuation	Speakerphone (SPK)	Mono (M)	Input Path (IP)	Mono (M)	SPKM_IPM_A
57h	Speakerphone Mono Mixer: XSP Mono/L/R Attenuation	Speakerphone (SPK)	Mono (M)	Auxiliary Serial Port (XSP)	Mono/Left/ Right	SPKM_XSP_A
58h	Speakerphone Mono Mixer: ASP Mono/L/R Attenuation	Speakerphone (SPK)	Mono (M)	Audio Serial Port (ASP)	Mono/Left/ Right	SPKM_ASP_A
59h	Speakerphone Mono Mixer: VSP Mono Attenuation	Speakerphone (SPK)	Mono (M)	Voice Serial Port (VSP)	Mono (M)	SPKM_VSPM_A
5Ah	Ear Speaker/Speakerphone Line Output Mono Mixer: Input Path Mono Attenuation	Ear Speaker/ Speakerphone Line Output (ESL)	Mono (M)	Input Path (IP)	Mono (M)	ESLM_IPM_A
5Bh	Ear Speaker/Speakerphone Line Output Mono Mixer: XSP Mono/L/R Attenuation	Ear Speaker/ Speakerphone Line Output (ESL)	Mono (M)	Auxiliary Serial Port (XSP)	Mono/Left/ Right	ESLM_XSP_A
5Ch	Ear Speaker/Speakerphone Line Output Mono Mixer: ASP Mono/L/R Attenuation	Ear Speaker/ Speakerphone Line Output (ESL)	Mono (M)	Audio Serial Port (ASP)	Mono/Left/ Right	ESLM_ASP_A
5Dh	Ear Speaker/Speakerphone Line Output Mono Mixer: VSP Mono Attenuation	Ear Speaker/ Speakerphone Line Output (ESL)	Mono (M)	Voice Serial Port (VSP)	Mono (M)	ESLM_VSPM_A

6.48.1 Mono Mixer Input Attenuation

*_A[5:0]	Volume
000000	0 dB
000001	-1.0 dB
000010	-2.0 dB
...	...
111110	-62.0 dB
111111	Mute
Step Size:	1.0 dB
Application:	Refer to sections "Mixer Input Attenuation Adjustment" on page 63 and "Mixer Attenuation Values" on page 65.

6.49 Interrupt Mask Register 1 (Address 5Eh)

The bits of this register serve as a mask for the interrupt sources found in Interrupt Status Register 1. If a mask bit is set, the interrupt is unmasked, meaning that its occurrence affects the INT pin. If a mask bit is cleared, the condition is masked, meaning that its occurrence will not affect the INT pin. The bit positions align with the corresponding bits in Interrupt Status Register 1.

7	6	5	4	3	2	1	0
Reserved	M_MIC2_SDET	Reserved	M_THMOVLD	M_DIGMIXOVFL	Reserved	M_IPBOVFL	M_IPAOVFL

All the mask bits default to 0b.

6.50 Interrupt Mask Register 2 (Address 5Fh)

The bits of this register serve as a mask for the respective interrupt sources found in Interrupt Status Register 2. If a mask bit is set, the interrupt is unmasked, meaning that its occurrence affects the INT pin. If a mask bit is cleared, the condition is masked, meaning that its occurrence does not affect the INT pin. The bit positions align with the corresponding bits in Interrupt Status Register 2.

7	6	5	4	3	2	1	0
Reserved	Reserved	M_VASRC_ DOLK	M_VASRC_DILK	M_AASRC_ DOLK	M_AASRC_DILK	M_XASRC_ DOLK	M_XASRC_DILK

All the mask bits default to 0b.

6.51 Interrupt Status Register 1 (Address 60h)

- Read Only

Refer to section "Interrupts" on page 70.

7	6	5	4	3	2	1	0
Reserved	MIC2_SDET	Reserved	THMOVLD	DIGMIXOVFL	Reserved	IPBOVFL	IPAOVFL

6.51.1 MIC2 Short Detect

- Read Only; not *sticky; ↓ or ↑ edge can trigger interrupt*

Indicates a short-to-ground condition across the MIC2 microphone module nodes, measured at the MIC2_SDET input pin. State transitions are debounced for 20 ms in both the rising and falling directions. Rising and falling transitions cause an interrupt, if the associated interrupt mask bit is set.

MIC2_SDET	Pin State
0	No short condition detected
1	Short condition detected

Note: MIC2_BIAS must be enabled (PDN_MIC2_BIAS = 0b and PDN = 0b) before MIC2 Short Detect occurrences are posted or used for automatic muting.

6.51.2 Thermal Overload Detect

- Read Only; sticky; ↑ edge can trigger interrupt

If thermal sensing is enabled, this bit indicates whether the current junction temperature has exceeded the thermal overload threshold. This status bit is sticky. Rising-edge state transitions cause an interrupt, if the associated interrupt mask bit is set.

THMOVLD	State
0	No thermal overload condition
1	Thermal overload condition detected

6.51.3 Digital Mixer Overflow

- Read Only; sticky; ↑edge can trigger interrupt

Indicates the over-range status in the digital mixer data path. This status bit is sticky. Rising-edge state transitions cause an interrupt, if the associated interrupt mask bit is set.

DIGMIXOVFL	PCM Overflow Status
0	No digital clipping has occurred in the data path of the digital mixer
1	Digital clipping has occurred in the data path of the digital mixer

6.51.4 Input Path x Overflow

- Read Only; sticky; ↑edge can trigger interrupt

Indicates the over-range status in the input path channel x signal path. This status bit is sticky. Rising-edge state transitions cause an interrupt if the associated interrupt mask bit is set.

IPxOVFL	Input Path Overflow Status
0	No clipping has occurred anywhere in the Input Path signal path
1	Clipping has occurred in the Input Path signal path

6.52 Interrupt Status Register 2 (Address 61h)

- Read Only

Refer to section “Interrupts” on page 70.

7	6	5	4	3	2	1	0
Reserved	Reserved	VASRC_DOLK	VASRC_DILK	AASRC_DOLK	AASRC_DILK	XASRC_DOLK	XASRC_DILK

6.52.1 Voice ASRC Data Out Lock

- Read Only; sticky; ↑edge can trigger interrupt

Indicates the lock status of the ASRC for the voice data out. This status bit is sticky. Rising-edge state transitions cause an interrupt, if the associated interrupt mask bit is set.

VASRC_DOLK	Status
0	Unlocked
1	Locked

6.52.2 Voice ASRC Data In Lock

- Read Only; sticky; ↑edge can trigger interrupt

Indicates the lock status of the ASRC for the voice data in. This status bit is sticky. Rising-edge state transitions cause an interrupt, if the associated interrupt mask bit is set.

VASRC_DILK	Status
0	Unlocked
1	Locked

6.52.3 Audio ASRC Data Out Lock

- Read Only; sticky; ↑edge can trigger interrupt

Indicates the lock status of the ASRC for the audio data out. This status bit is sticky. Rising-edge state transitions cause an interrupt, if the associated interrupt mask bit is set.

AASRC_DOLK	**Status**
0	Unlocked
1	Locked

6.52.4 Audio ASRC Data In Lock

- Read Only; sticky; ↑edge can trigger interrupt

Indicates the lock status of the ASRC for the audio data in. This status bit is sticky. Rising-edge state transitions cause an interrupt, if the associated interrupt mask bit is set.

AASRC_DILK	**Status**
0	Unlocked
1	Locked

6.52.5 Auxiliary ASRC Data Out Lock

- Read Only; sticky; ↑edge can trigger interrupt

Indicates the lock status of the ASRC for the auxiliary data out. This status bit is sticky. Rising-edge state transitions cause an interrupt, if the associated interrupt mask bit is set.

XASRC_DOLK	**Status**
0	Unlocked
1	Locked

6.52.6 Auxiliary ASRC Data In Lock

- Read Only; sticky; ↑edge can trigger interrupt

Indicates the lock status of the ASRC for the auxiliary data in. This status bit is sticky. Rising-edge state transitions cause an interrupt, if the associated interrupt mask bit is set.

XASRC_DILK	**Status**
0	Unlocked
1	Locked

6.53 Fast Mode 1 (Address 7Eh)

7	6	5	4	3	2	1	0
FM15	FM14	FM13	FM12	FM11	FM10	FM9	FM8

6.53.1 Fast Mode Bits 15:8

See "Fast Start Mode" on page 73.

6.54 Fast Mode 2 (Address 7Fh)

7	6	5	4	3	2	1	0
FM7	FM6	FM5	FM4	FM3	FM2	FM1	FM0

6.54.1 Fast Mode Bits 7:0

See "Fast Start Mode" on page 73.

7. PCB LAYOUT CONSIDERATIONS

7.1 Power Supply

As with any high-resolution converter, the CS42L73 requires careful attention to power supply and grounding arrangements for its potential performance to be realized. Figure 1 on page 17 shows the recommended power arrangements, with VA and VCP connected to clean supplies. VL, which powers the digital circuitry, may be run from the system logic supply. Alternatively, VL may be powered from the analog supply via a ferrite bead. In this case, no additional devices should be powered from VL.

7.2 Grounding

Extensive use of power and ground planes, ground plane fill in unused areas and surface mount decoupling capacitors are recommended. Decoupling capacitors must be as close to the pins of the CS42L73 as possible. The low value ceramic capacitor must be closest to the pin and must be mounted on the same side of the board as the CS42L73 to minimize inductance effects. All signals, especially clocks, must be kept away from the FILT+, ANA_VQ, and SPK_VQ pins in order to avoid unwanted coupling into the modulators. The FILT+, ANA_VQ, SPK_VQ, +VCP_FILT and -VCP_FILT capacitors must be positioned to minimize the electrical path from each respective pin to AGND (PGND with respect to SPK_VQ).

7.3 Layout With Fine-Pitch, Ball-Grid Packages

PCB layouts with fine-pitch, ball-grid packages, such as those available for the CS42L73, can benefit from using the particular layout approaches. This is especially true when routing to/from balls inside the outer ring of the package's ball array. Using the via-in-pad, filled-micro-via, and blind-via technologies can ease routing congestion, allowing access to all the packages balls. For more detailed layout assistance, please contact Cirrus Logic.

8. PERFORMANCE DATA

Note, unless otherwise noted, the data/plots in this section are for nominal supply voltages (VA = VCP = VL = 1.80 V, VP = 3.70 V), a 25 °C ambient temperature, and were taken using the connections shown in the “Typical Connection Diagram” on page 17.

8.1 Analog Input Path Attributes

8.1.1 PGA Analog Volume Nonlinearity (DNL and INL)

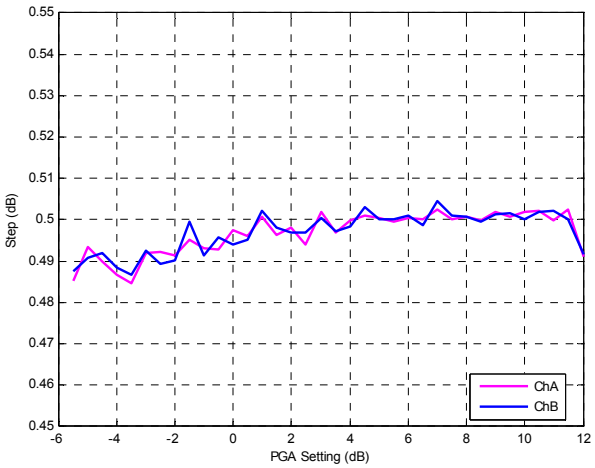


Figure 42. PGA DNL

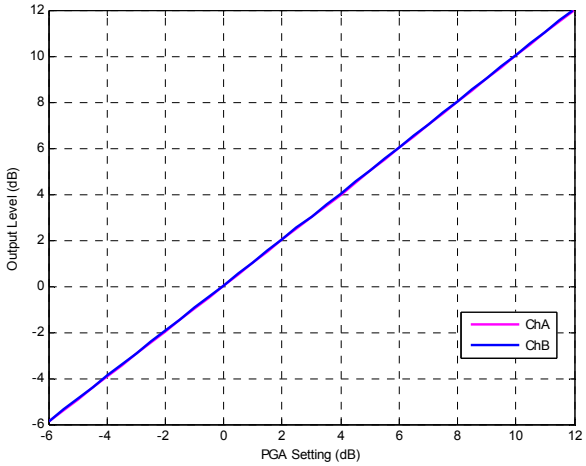


Figure 43. PGA INL

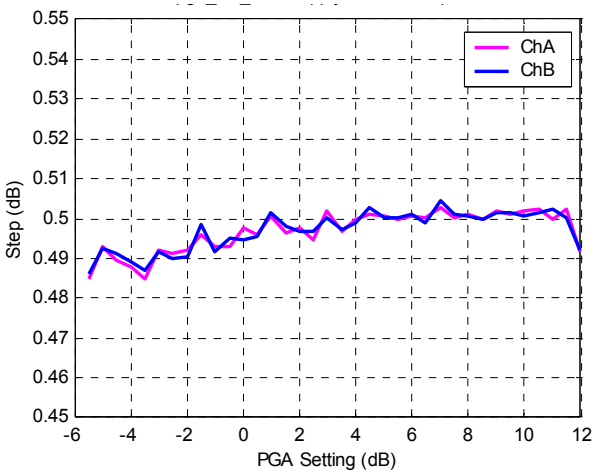


Figure 44. PGA + Preamp (+10 dB) DNL

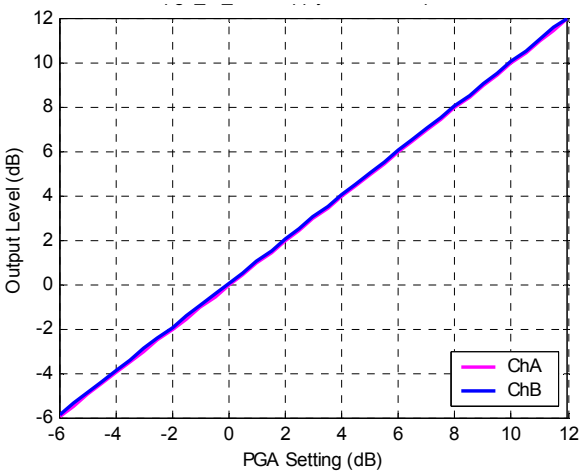


Figure 45. PGA + Preamp (+10 dB) INL

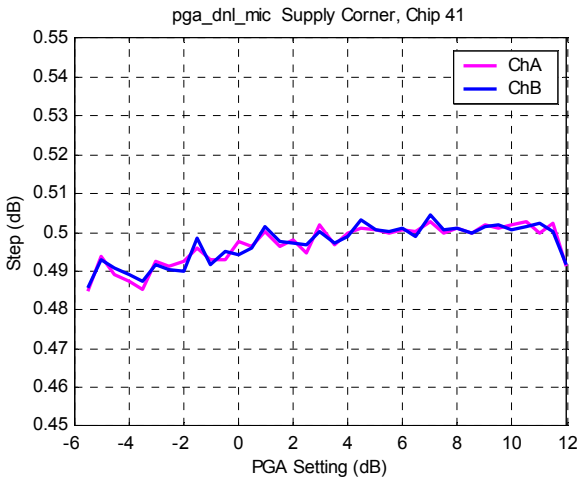


Figure 46. PGA + Preamp (+20 dB) DNL

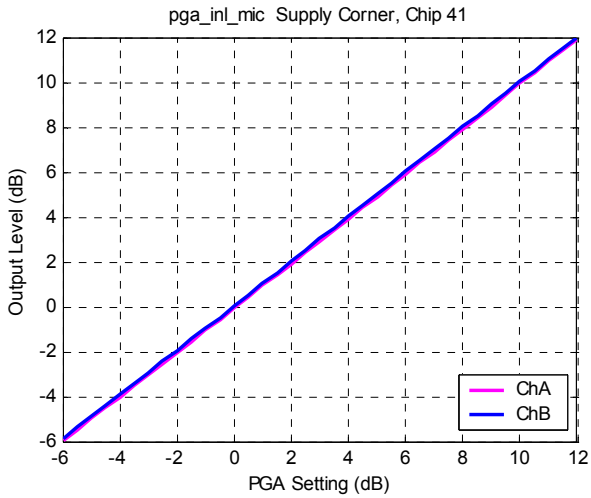


Figure 47. PGA + Preamp (+20 dB) INL

8.2 Analog Mic/Line ADC and Digital Mic Input Path Attributes

8.2.1 Input Path Digital LPF Response

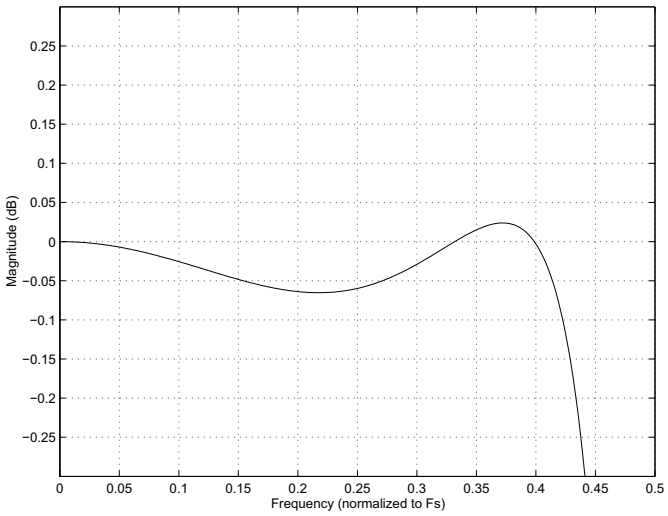


Figure 48. Input Path LPF Frequency Response

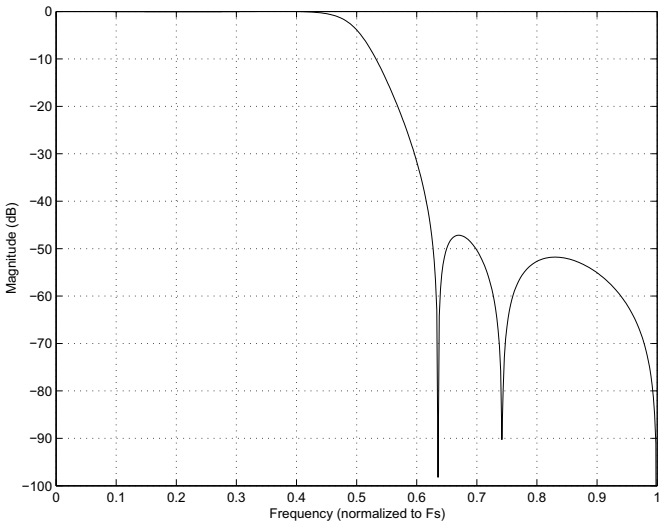


Figure 49. Input Path LPF Stopband Rejection

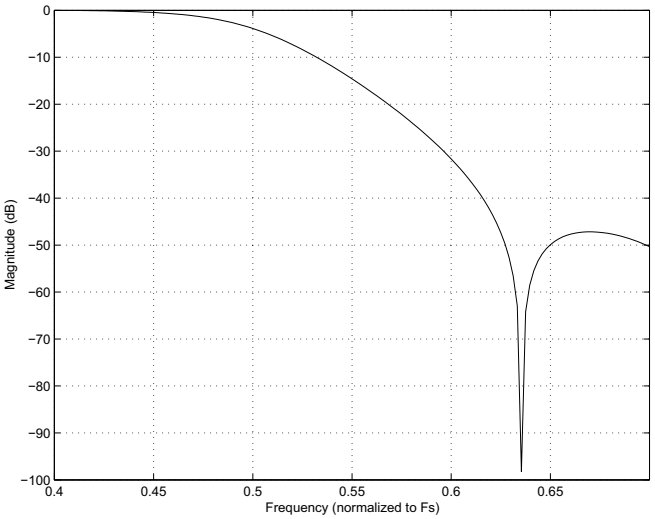


Figure 50. Input Path LPF Transition Band

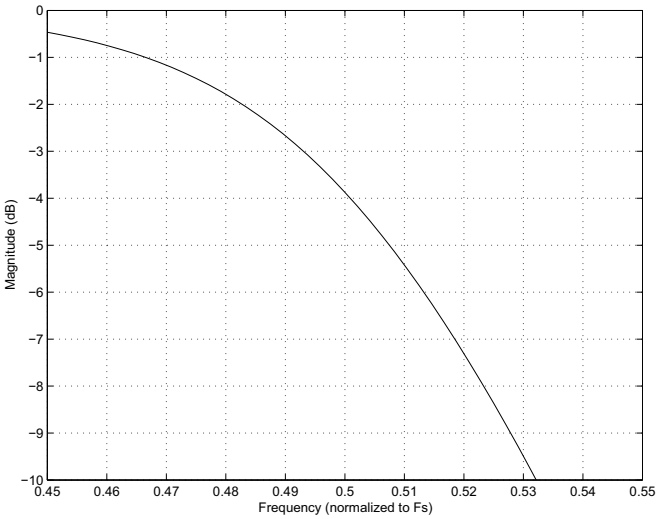


Figure 51. Input Path LPF Transition Band Detail

8.2.2 Input Path Digital HPF Response

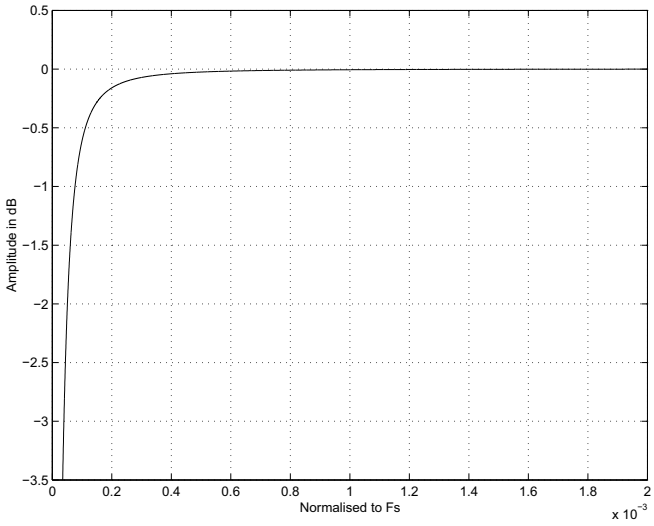


Figure 52. Input Path HPF Frequency Response

8.3 Core Circuitry Attributes

8.3.1 ASRC Attributes

8.3.1.1 Response

The following curves illustrate the ASRC frequency response. The horizontal frequency axis is normalized to the external Fs (Fs_{ext}, the serial port sample rate), because for all external Fs values, the plots are exactly the same—only the scaling of the horizontal axis changes. Hence, the frequency for a given point on the curves and a given external sample rate is the normalized frequency axis point multiplied by the external frequency.

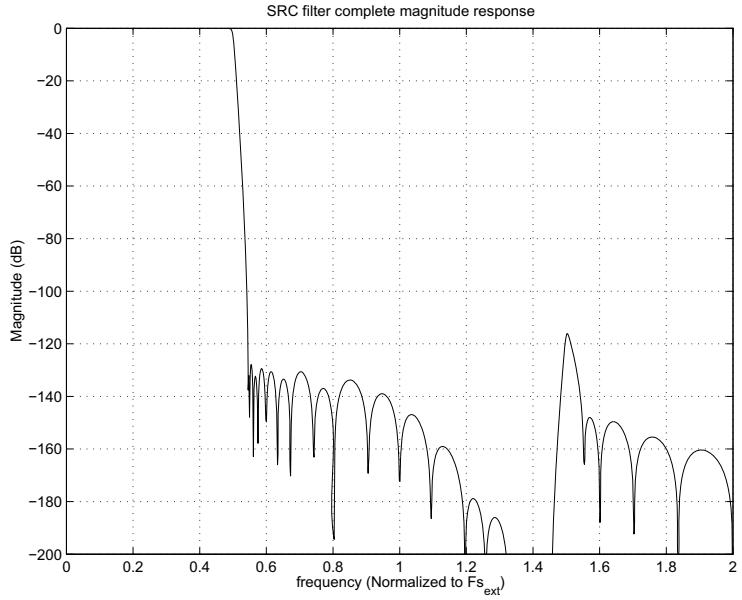


Figure 53. ASRC Frequency Response

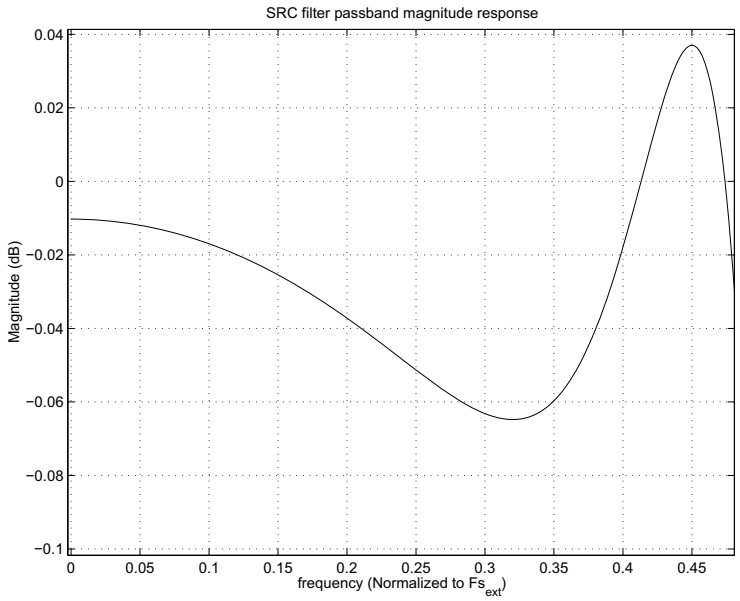


Figure 54. ASRC Passband Frequency Response

8.3.1.2 Group Delay

The group-delay equations for the ASRCs are specified in "ASRC Digital Filter Characteristics" on page 25. The following chart illustrates, for the extreme supported internal sample rates (Fs) and standard audio sample rates, the input (from the serial ports to the core) and output (from the core to the serial ports) group delays through the ASRCs.

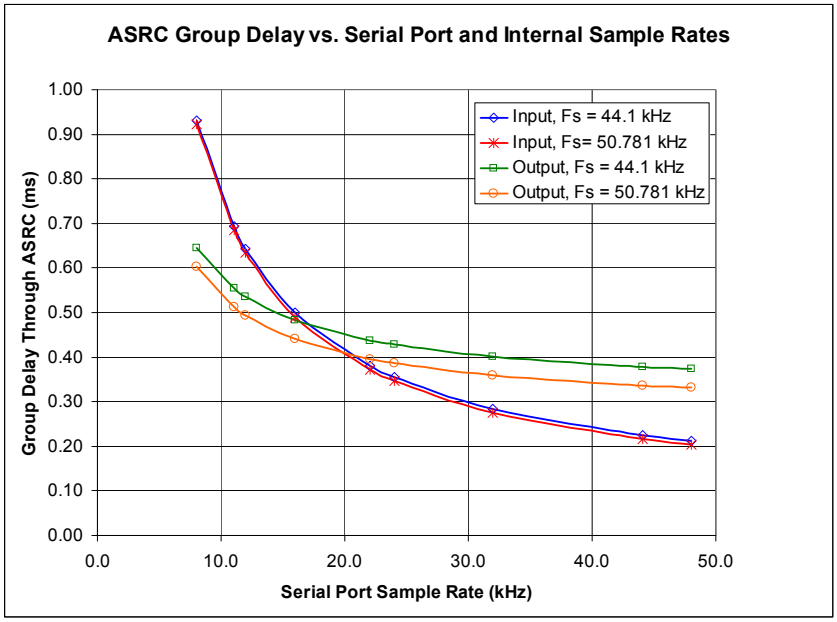


Figure 55. ASRC Group Delay vs. Serial Port and Internal Sample Rates

8.3.1.3 Lock Time

The following table outlines the ASRC lock times for the extremes of the standard audio serial port sample rates for all sample rate programming configurations and directions. If the table asks if the serial port sample rate has been programmed, it is asking if registers "XSP Sample Rate" on page 93, "ASP Sample Rate" on page 90, and/or "VSP Sample Rate" on page 93 (whichever are applicable) have been properly programmed (vs. being set to "don't know") before the ASRC attempts to lock.

Table 15. ASRC Lock Times

Serial Port Sample Rate Programmed (Y/N)?	ASRC Direction	Serial Port Sample Rate (kHz)	ASRC Lock Time (ms)
Y	Any	Any	≤ 19
N	From core to serial port	8	93
N		48	11
N	From serial port to core	8	63
N		48	43

8.4 Analog Output Paths Attributes

8.4.1 DAC Digital LPF Response

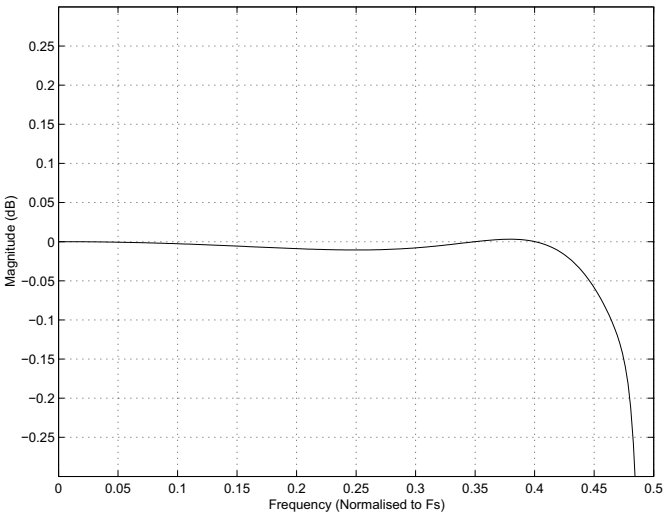


Figure 56. DAC LPF Frequency Response

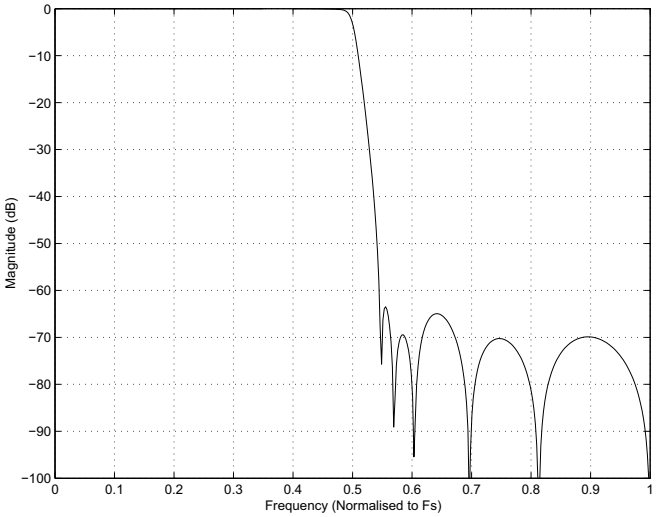


Figure 57. DAC LPF Stopband Rejection to 1x Fs

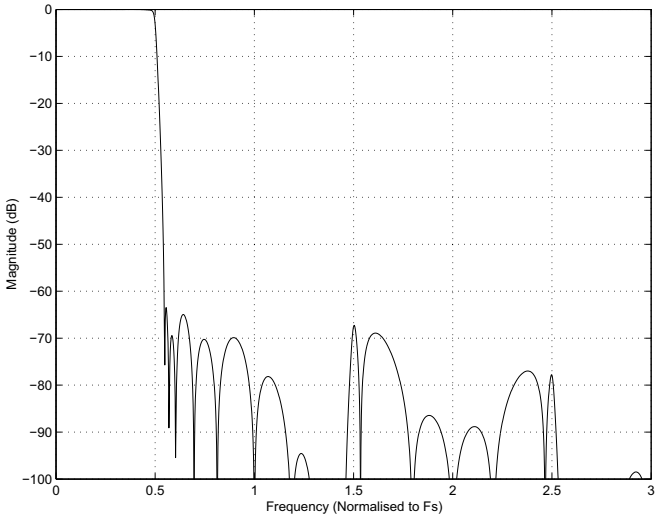


Figure 58. DAC LPF Stopband Rejection to 3x Fs

8.4.2 DAC HPF Response

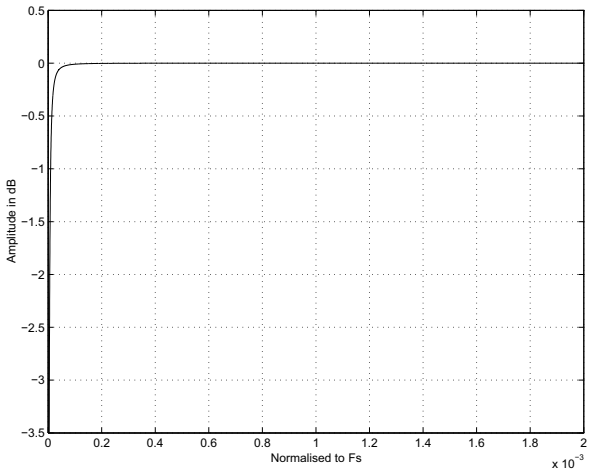


Figure 59. DAC HPF Frequency Response

8.4.3 Output Analog Volume Nonlinearity (DNL and INL)

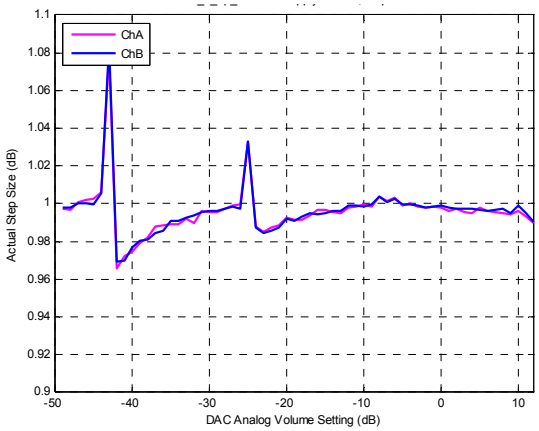


Figure 60. HPOUTx DNL (-50 to +12 dB)

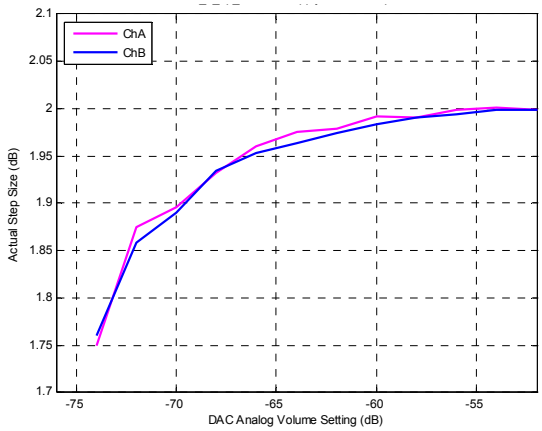


Figure 61. HPOUTx DNL (-76 to -52 dB)

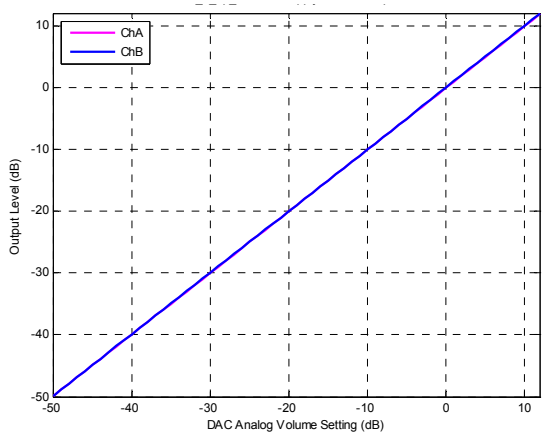


Figure 62. HPOUTx INL (-50 to +12 dB)

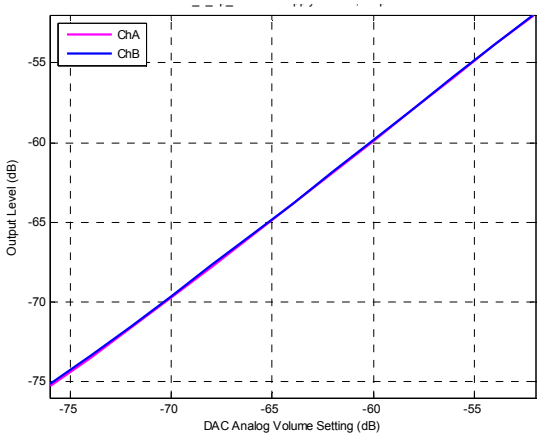


Figure 63. HPOUTx INL (-76 to -52 dB)

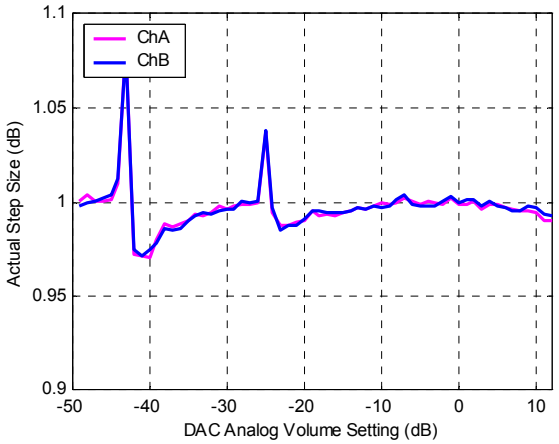


Figure 64. LINEOUTx DNL (-50 to +12 dB)

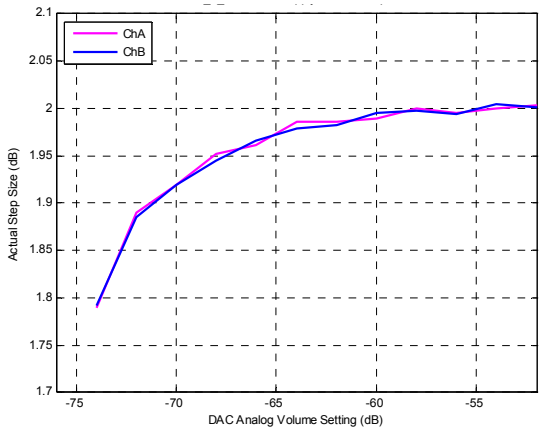


Figure 65. LINEOUTx DNL (-76 to -52 dB)

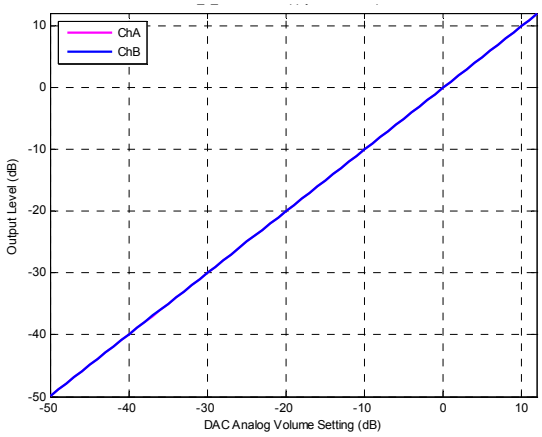


Figure 66. LINEOUTx INL (-50 to +12 dB)

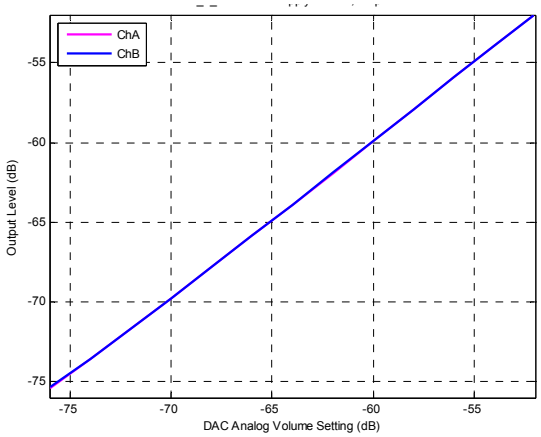


Figure 67. LINEOUTx INL (-76 to -52 dB)

8.4.4 Startup Times

Table 16 lists the startup times for the analog output paths. For each case, the device has been previously configured for a known Fs with all clocks applied (see "Lock Time" on page 130 for the length of this initial period). The times recorded in the table are from when the PDN bit is cleared until the output path produces full-scale audio at the output pins of the device. If a soft ramp is enabled, the setting is 1/8 dB per Fs. The values in Table 16 assume fast start mode is not being used; see Table 13 in Section 4.17.

Table 16. Analog Output Startup Times

Output Path	Mixer Soft Ramp	Digital Soft Ramp	Analog Zero Cross	Startup Time (ms)
HPOUT/LINEOUT	Enabled	Enabled	Enabled	70
	Disabled	Enabled	Enabled	70
	Disabled	Disabled	Enabled	52
	Disabled	Disabled	Disabled	52
SPKOUT/SPKLINEOUT	Enabled	Enabled	Enabled	140
	Disabled	Disabled	Enabled	120
EARSPKOUT	Enabled	Enabled	Enabled	46
	Disabled	Disabled	Enabled	30

9. PARAMETER DEFINITIONS

Dynamic Range

The ratio of the rms value of the signal to the rms sum of all other spectral components over the specified bandwidth. Dynamic range is a signal-to-noise ratio measurement over the specified band width made with a -60 dB signal. 60 dB is added to resulting measurement to refer the measurement to full-scale. This technique ensures that the distortion components are below the noise level and do not affect the measurement. This measurement technique has been accepted by the Audio Engineering Society, AES17-1991, and the Electronic Industries Association of Japan, EIAJ CP-307. Dynamic range is expressed in decibel units.

Frequency Response

A measure of the amplitude response variation from 10 Hz to 20 kHz relative to the amplitude response at 1 kHz. Frequency Response is expressed in decibel units.

Gain Drift

The change in gain value with temperature, expressed in ppm/°C units.

Interchannel Gain Mismatch

The gain difference between left and right channel pairs. Interchannel Gain Mismatch is expressed in decibel units.

Interchannel Isolation

A measure of crosstalk between the left and right channel pairs. Interchannel Isolation is measured for each channel at the converter's output with no signal to the input under test and a full-scale signal applied to the other channel. Interchannel isolation is expressed in decibel units.

Load Resistance and Capacitance

The recommended minimum resistance and maximum capacitance required for the internal op-amp's stability and signal integrity. The load capacitance effectively moves the band-limiting pole of the amp in the output stage. Increasing the load capacitance beyond the recommended value can cause the internal op-amp to become unstable.

Offset Error

The deviation of the mid-scale transition (111...111 to 000...000) from the ideal.

Output Offset Voltage

Describes the DC offset voltage present at the amplifier’s output when its input signal is in a MUTE state. The offset exists due to CMOS process limitations and is proportional to analog volume settings. When measuring the offset out the line amplifier, the line amplifier is ON while the headphone amplifier is OFF; when measuring the offset out the headphone amplifier, the headphone amplifier is ON while the line amplifier is OFF.

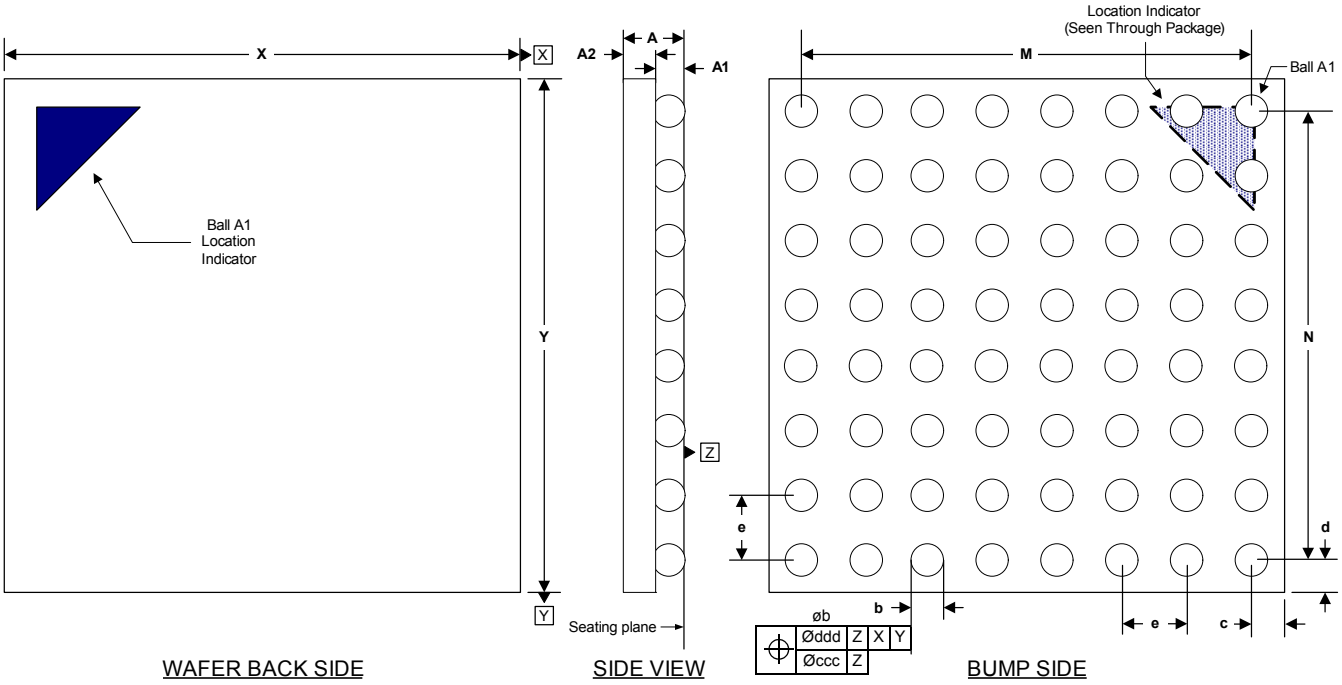
Total Harmonic Distortion + Noise (THD+N)

The ratio of the unweighted rms value of the signal to the rms sum of all other spectral components over the specified bandwidth (typically 10 Hz to 20 kHz), including distortion components. THD+N is measured at –1 dBFS for the analog input and 0 dB for the analog output as suggested in AES17-1991 Annex A. THD+N is expressed in decibel units.

10.PACKAGE DIMENSIONS

10.1 WLCSP Package

64-Ball WLCSP (3.44 x 3.44 mm Body) Package Drawing



Notes:
- Dimensioning and tolerances per ASME Y 14.5M–1994.
- The Ball A1 position indicator is for illustration purposes only and may not be to scale.
- Dimension “b” applies to the solder sphere diameter and is measured at the midpoint between the package body and the seating plane Datum Z.

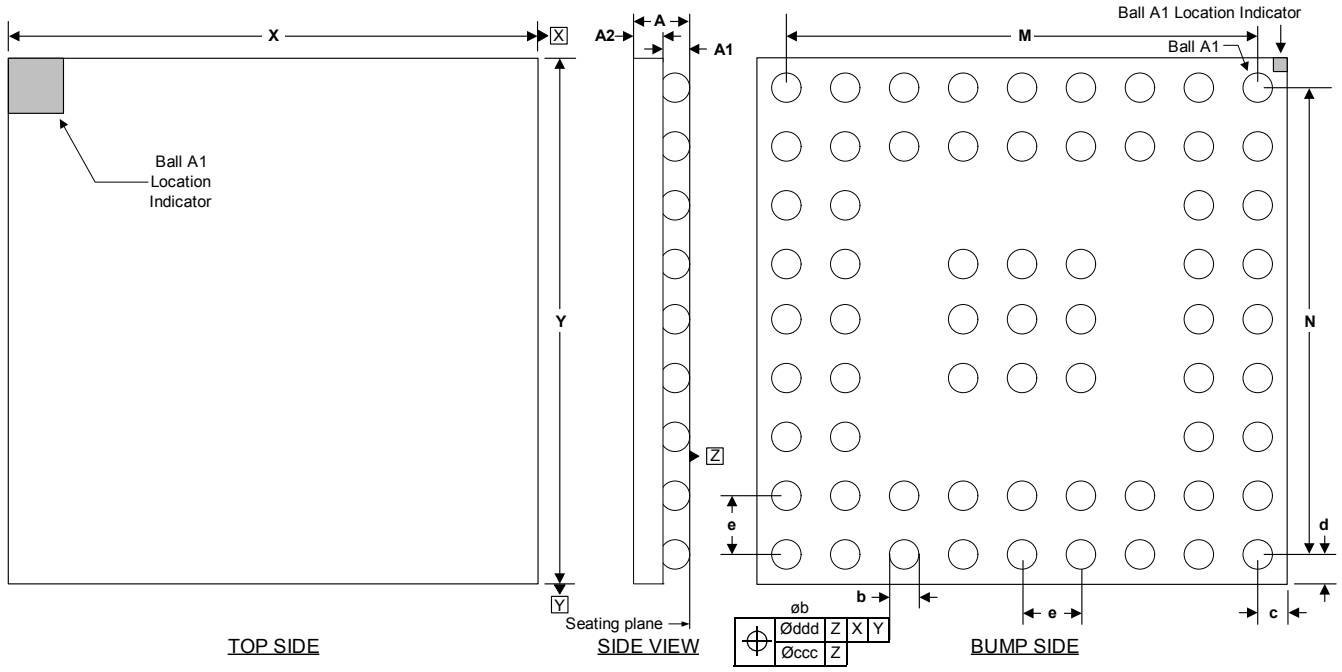
Table 17. WLCSP Package Dimensions

Dimension	Millimeters		
A	0.450	0.505	0.560
A1	0.170	0.200	0.230
A2	0.280	0.305	0.330
M	BSC	2.800	BSC
N	BSC	2.800	BSC
b	0.230	0.260	0.290
c	REF	0.320	REF
d	REF	0.320	REF
e	BSC	0.400	BSC
X	3.415	3.440	3.465
Y	3.415	3.440	3.465
ccc = 0.05 ddd = 0.15			

Note: Controlling dimension is millimeters.

10.2 FBGA Package

65-Ball FBGA (5 x 5 mm Body) Package Drawing



Notes:
- Dimensioning and tolerances per ASME Y 14.5M–1994.
- The Ball A1 position indicator is for illustration purposes only and may not be to scale.
- Dimension “b” applies to the solder sphere diameter and is measured at the midpoint between the package body and the seating plane Datum Z.

Table 18. FBGA Package Dimensions

Dimension	Millimeters		
	Minimum	Nominal	Maximum
A	0.74	0.87	1.00
A1	0.16	0.21	0.26
A2	0.58	0.66	0.74
M	BSC	4.00	BSC
N	BSC	4.00	BSC
b	0.27	0.30	0.37
c	REF	0.50	REF
d	REF	0.50	REF
e	BSC	0.50	BSC
X	4.90	5.00	5.10
Y	4.90	5.00	5.10
ccc = 0.05 ddd = 0.15			

Note: Controlling dimension is millimeters.

11.THERMAL CHARACTERISTICS

Parameter (Notes 1 and 2)	Symbol	Min	Typ	Max	Units
WLCSP Package					
Junction to Ambient Thermal Impedance	θ_{JA}	-	43	-	°C/Watt
Junction to Printed Circuit Board Thermal Impedance	θ_{JB}	-	10	-	°C/Watt
FBGA Package					
Junction to Ambient Thermal Impedance	θ_{JA}	-	58	-	°C/Watt
Junction to Printed Circuit Board Thermal Impedance	θ_{JB}	-	12	-	°C/Watt

Notes:
1. Test printed circuit-board assembly (PCBA) constructed in accordance with JEDEC standard JESD51-9. Two-signal, two-plane (2s2p) PCB used.
2. Test conducted with still air, in accordance with JEDEC standards JESD51, JESD51-2A, and JESD51-8.

12.ORDERING INFORMATION

Product	Description	Package	Halogen Free	Pb Free	Grade	Temperature Range	Container	Order #
CS42L73	Ultralow Power Mobile Audio and Telephony CODEC	64 Ball WLCSP	Yes	Yes	Commercial	–40 to +85 °C	Tape and Reel	CS42L73-CWZR
		65 Ball FBGA	—				Tray	CS42L73-CRZ
							Tape and Reel	CS42L73-CRZR

13.REFERENCES

- NXP Semiconductors (founded by Philips Semiconductor), UM10204 Rev. 03, 19 June 2007, *The I²C-Bus Specification and User Manual*.
  http://www.nxp.com
- Japan Electronics and Information Technology Industries Association (JEITA), CP-2905B Rev. 2003.05, *Methods of Measurement for Battery Duration on Portable Audio Equipment*.
  http://www.jeita.or.jp/english
- Joint Electron Device Engineering Council (JEDEC), JESD51 Rev. December 1995, *Methodology for the Measurement of Component Packages (Single Semiconductor Device).*
  http://www.jedec.org
- Joint Electron Device Engineering Council (JEDEC), JESD51-2A Rev. January 2008, *Integrated Circuits Thermal Test Method Environmental Conditions—Natural Convection (Still Air).*
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- Joint Electron Device Engineering Council (JEDEC), JESD51-8 Rev. October 1999, *Integrated Circuit Thermal Test Method Environmental Conditions — Junction-to-Board*.
  http://www.jedec.org
- Joint Electron Device Engineering Council (JEDEC), JESD51-9 Rev. July 2000, *Test Boards for Area Array Surface Mount Package Thermal Measurements*.
  http://www.jedec.org

14.REVISION HISTORY

Revision	Changes
F1	• Removed all references to PDN_LDO. • Updated Stereo DAC to Headphone Amplifier: High HP power output and Mono DAC to Speakerphone Amplifier: High output power on page 2. • Added (Note 4) to Absolute Maximum Ratings on page 20. Added a cross-reference to this note in Section 4.12.1 and Section 4.12.6. • Added Analog Output Current Limiter On Threshold specification to DC Electrical Characteristics on page 21. • Clarified Input Impedance (Note 17), MIC1/MIC2 and Input Impedance (Note 17), LINEINA/LINEINB test conditions on page 23 as 1 kHz. • Changed expected input impedance variance in (Note 17) on page 24 to ±20%. • Updated frequency response, passband, and group delay specifications Stereo-ADC and Dual-Digital-Mic Digital Filter Characteristics on page 24. • Updated Mic Bias PSRR in Mic BIAS Characteristics on page 26. • Updated HP Output full-scale voltage and Power in Serial Port to Stereo HP Output Characteristics on page 27. • Added Output Impedance to Serial Port to Stereo HP Output Characteristics on page 27. • Updated Speakerphone output voltage and power characteristics in Serial Port-to-Mono Speakerphone Output Characteristics on page 31. • Updated Speakerphone PSRR in Serial Port-to-Mono Speakerphone Output Characteristics on page 31. • Updated Stereo/Mono DAC Interpolation and On-Chip Digital/Analog Filter Characteristics table on page 33 to separate EAR/HP/LINE and SPK/SPKLINEOUT specifications. • Updated frequency response, passband, and group delay specifications in Stereo/Mono DAC Interpolation and On-Chip Digital/Analog Filter Characteristics on page 33. • Removed old note 46 (pertaining to DAC high-pass filter). • Updated $\overline{\text{MIC2\_SDET}}$ high level input voltage specification in Digital Interface Specifications and Characteristics on page 35. • Updated DMIC_CLK rise time and DMIC_SD setup and hold time specifications and conditions in Switching Specifications—Digital Mic Interface on page 37. • Updated Figure 11 to add markings for 90%/10% and VIH/VIL in Switching Specifications—Digital Mic Interface on page 37. • Generalized statement about oversampling at the end of the first paragraph in Section 4.1.1, “Basic Architecture.” • Added Section 4.4, “Pseudodifferential Outputs.” • Added Section 4.5. “Class H Amplifier.” • Added Section 4.6, “DAC Limiter.” • Added Section 4.7, “Analog Output Current Limiter.” • Updated wait time for Step 3 in Section 4.12.3, “Power-Down Sequence (xSP to HP/LO).” • Added step to set the DISCHG_FILT bit to Section 4.12.6, “Final Power-Down Sequence.” • Added Section 4.17, “Fast Start Mode.” • Added Section 4.18, “Headphone High-Impedance Mode.” • Added Section 6.1, “Fast Mode Enable (Address 00h),” Section 6.53, “Fast Mode 1 (Address 7Eh),” and Section 6.54, “Fast Mode 2 (Address 7Fh).” • Added note to Section 6.17.1, “Digital Swap/Mono,” register description. • Corrected register names referred to in the body of Section 6.41.1, “ALC Release Rate for Channels A and B.” • Updated INL DNL plots in Section 8.1.1 and Section 8.4.3. • Added Section 8.4.4, “Startup Times.” • Updated package dimensions formatting and added ccc/ddd positional tolerance values to Section 10. • Removed the CS42L73-CWZ option from ordering information table and added WLCSP halogen-free statement in Section 12.

Contacting Cirrus Logic Support

For all product questions and inquiries, contact a Cirrus Logic Sales Representative.
To find one nearest you, go to www.cirrus.com.