



## FEATURES

- AC '97 2.1 Compatible
- Industry Leading Mixed Signal Technology
- 20-bit Stereo Digital-to-Analog Converter and 18-bit Stereo Analog to Digital Converter with Sample Rate Conversion
- Four Analog Line-level Stereo Inputs for Connection from LINE IN, CD, VIDEO, and AUX
- Two Analog Line-level Mono Inputs for Modem Subsystem and Internal PC Beeper
- Mono Microphone Input Switchable from Two External Sources
- High Quality Pseudo Differential CD Input
- Dual Stereo Line-level Outputs
- Extensive Power Management Support
- Meets or Exceeds Microsoft's® PC 98 and PC 99 Audio Performance Requirements
- CrystalClear 3D Stereo Enhancement
- S/PDIF Digital Audio Output
- Sample Rate Converters

## ORDERING INFO

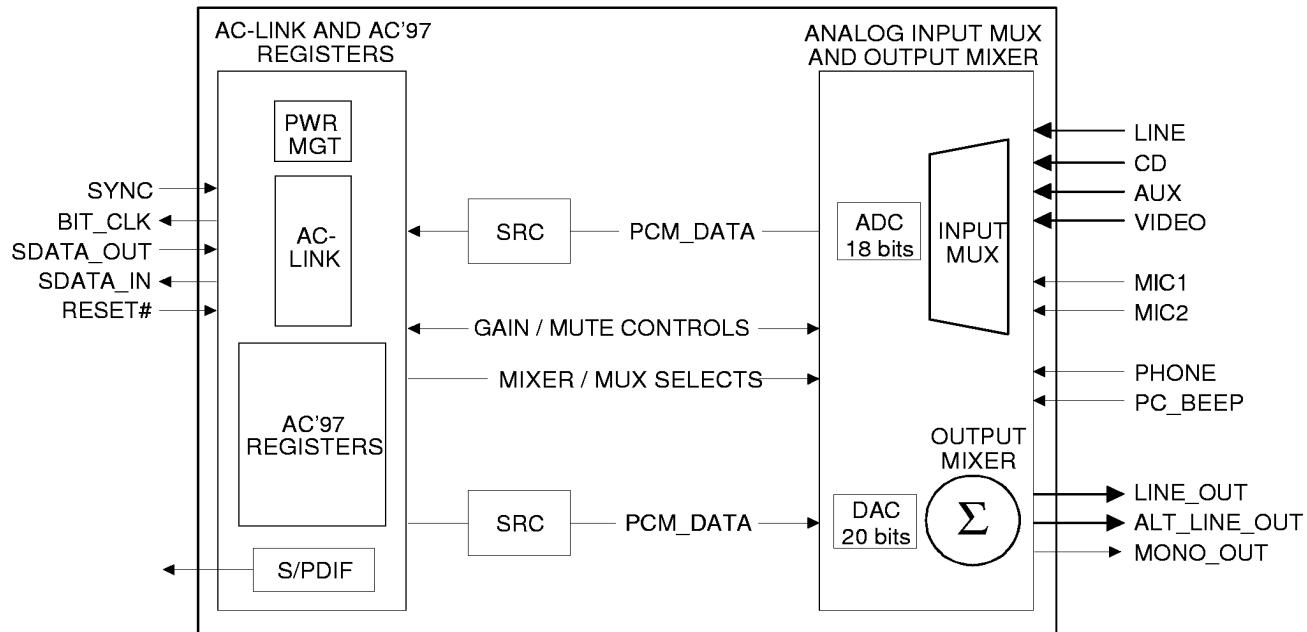
CS4299-KQ, 48-pin TQFP, 9x9x1 mm  
 CS4299-JQ, 48-pin TQFP, 9x9x1 mm

**CrystalClear™  
 SoundFusion™ Audio  
 Codec '97**

## DESCRIPTION

The CS4299 is an AC '97 2.1 compatible stereo audio codec designed for PC multimedia systems. Using the industry leading CrystalClear delta-sigma and mixed signal technology, the CS4299 paves the way for PC 98 and PC 99-compliant desktop, portable, and entertainment PCs, where high-quality audio is required.

The CS4299, when coupled with a PCI audio accelerator or core logic supporting the AC '97 interface, implements a cost effective, superior quality, audio solution. The CS4299 surpasses audio quality standards such as PC 98, PC 99, and AC '97 2.1.



## TABLE OF CONTENTS

<b>1.0 CHARACTERISTICS AND SPECIFICATIONS .....</b>	<b>4</b>
Analog Characteristics .....	4
Mixer Characteristics .....	5
Absolute Maximum Ratings .....	5
Recommended Operating Conditions .....	5
Digital Characteristics .....	5
Serial Port Timing .....	6
<b>2.0 GENERAL DESCRIPTION .....</b>	<b>9</b>
2.1 AC-Link .....	9
2.2 Control registers .....	9
2.3 Sample Rate Converters .....	10
2.4 Output Mixer .....	10
2.5 Input Mux .....	10
2.6 Volume Control .....	10
<b>3.0 AC '97 FRAME DEFINITION .....</b>	<b>12</b>
3.1 AC-Link Serial Data Output Frame .....	12
Serial Data Output Slot Tags (Slot 0) .....	12
Register Address (Slot 1) .....	12
Register Write Data (Slot 2) .....	13
Playback Data (Slots 3-10) .....	13
3.2 AC-Link Audio Input Frame .....	13
Serial Data Input Slot Tag Bits (Slot 0) .....	13
Read-Back Address Port (Slot 1) .....	14
Read-Back Data Port (Slot 2) .....	14
PCM Capture Data (Slot 3-10) .....	14
3.3 AC '97 Reset Modes .....	14
Cold AC '97 Reset .....	14
Warm AC '97 Reset .....	14
AC '97 Register Reset .....	15
3.4 AC-Link Protocol Violation - Loss of SYNC .....	15
<b>4.0 REGISTER INTERFACE .....</b>	<b>16</b>
Reset (Index 00h) .....	17
Master Volume (Index 02h) .....	17
Alternate Volume (Index 04h) .....	17
Master Mono Volume (Index 06h) .....	18
PC_BEEP Volume (Index 0Ah) .....	18
Phone Volume (Index 0Ch) .....	18
Microphone Volume (Index 0Eh) .....	18
Stereo Analog Mixer Input Gain (Index's 10h - 18h) .....	19
Input Mux Select (Index 1Ah) .....	19
Record Gain (Index 1Ch) .....	19

Microsoft is a registered trademark of Microsoft Corporation in the United States and/or other countries.

Intel is a registered trademark of Intel Corporation.

Crystal Clear and Sound Fusion are trademarks of Cirrus Logic.

Cirrus Logic, Inc. has made best efforts to ensure that the information contained in this document is accurate and reliable. However, the information is subject to change without notice and is provided "AS IS" without warranty of any kind (express or implied). No responsibility is assumed by Cirrus Logic, Inc. for the use of this information, nor for infringements of patents or other rights of third parties. This document is the property of Cirrus Logic, Inc. and implies no license under patents, copyrights, trademarks, or trade secrets. No part of this publication may be copied, reproduced, stored in a retrieval system, or transmitted, in any form or by any means (electronic, mechanical, photographic, or otherwise). Furthermore, no part of this publication may be used as a basis for manufacture or sale of any items without the prior written consent of Cirrus Logic, Inc. The names of products of Cirrus Logic, Inc. or other vendors and suppliers appearing in this document may be trademarks or service marks of their respective owners which may be registered in some jurisdictions. A list of Cirrus Logic, Inc. trademarks and service marks can be found at <http://www.cirrus.com>.

General Purpose (Index 20h) .....	20
3D Control (Index 22h) .....	20
Powerdown Control/Status (Index 26h) .....	21
Extended Audio ID (Index 28h) .....	21
Extended Audio Status / Control (Index 2Ah) .....	22
PCM Front DAC Rate (Index 2Ch) .....	22
PCM LR ADC Rate (Index 32h) .....	22
Slot Map (Index 5Eh) .....	23
S/PDIF Control (Index 68h) .....	24
Vendor ID1 (Index 7Ch) .....	24
Vendor ID2 (Index 7Eh) .....	25
<b>5.0 POWER MANAGEMENT</b> .....	<b>26</b>
<b>6.0 ANALOG HARDWARE DESCRIPTION</b> .....	<b>26</b>
6.1 Line-Level Inputs .....	26
6.2 Microphone Level Inputs .....	27
6.3 Mono Inputs .....	27
6.4 Line Level Outputs .....	28
6.5 Miscellaneous Analog Signals .....	29
6.6 Consumer IEC-958 Digital Interface (S/PDIF) .....	29
6.7 Power Supplies .....	30
<b>7.0 GROUNDING AND LAYOUT</b> .....	<b>30</b>
<b>8.0 PIN DESCRIPTIONS</b> .....	<b>32</b>
Digital I/O Pins .....	32
Analog I/O Pins .....	34
Filter and Reference Pins .....	36
Power Supplies .....	36
<b>9.0 PARAMETER AND TERM DEFINITIONS</b> .....	<b>38</b>
<b>10.0 REFERENCES</b> .....	<b>40</b>
<b>11.0 PACKAGE DIMENSIONS</b> .....	<b>41</b>

## 1.0 CHARACTERISTICS AND SPECIFICATIONS

**ANALOG CHARACTERISTICS** (Standard test conditions unless otherwise noted:  $T_{\text{ambient}} = 25^\circ \text{C}$ ,  $\text{AVdd} = 5.0 \text{ V} \pm 5\%$ ,  $\text{DVdd} = 3.3 \text{ V} \pm 5\%$ ; 1 kHz Input Sine wave; Sample Frequency,  $\text{Fs} = 48 \text{ kHz}$ ;  $Z_{\text{AL}} = 10 \text{ k}\Omega$ /680 pF load,  $C_{\text{DL}} = 18 \text{ pF}$  load (Note 1); Measurement bandwidth is 20 Hz - 20 kHz, 18-bit linear coding for ADC functions, 20-bit linear coding for DAC functions; Mixer registers set for unity gain.

Parameter (Note 2)	Symbol	Path (Note 3)	CS4299-KQ			CS4299-JQ			Unit
			Min	Typ	Max	Min	Typ	Max	
Full Scale Input Voltage									
Line Inputs		A-D	0.91	1.00	-	0.91	1.00	-	$\text{V}_{\text{RMS}}$
Mic Inputs (20dB = 0)		A-D	0.91	1.00	-	0.91	1.00	-	$\text{V}_{\text{RMS}}$
Mic Inputs (20dB = 1)		A-D	0.091	0.10	-	0.091	0.10	-	$\text{V}_{\text{RMS}}$
Full Scale Output Voltage									
Line, Alternate Line, and Mono Outputs		D-A	0.91	1.0	1.13	0.91	1.0	1.13	$\text{V}_{\text{RMS}}$
Frequency Response (Note 4)	FR								
Analog $\text{Ac} = \pm 0.5 \text{ dB}$		A-A	20	-	20,000	20	-	20,000	Hz
DAC $\text{Ac} = \pm 0.5 \text{ dB}$		D-A	20	-	20,000	20	-	20,000	Hz
ADC $\text{Ac} = \pm 0.5 \text{ dB}$		A-D	20	-	20,000	20	-	20,000	Hz
Dynamic Range									
Stereo Analog inputs to LINE_OUT	DR	A-A	90	95	-	-	90	-	$\text{dB FS A}$
Mono Analog inputs to LINE_OUT		A-A	85	90	-	-	85	-	$\text{dB FS A}$
DAC Dynamic Range		D-A	85	90	-	-	87	-	$\text{dB FS A}$
ADC Dynamic Range		A-D	85	90	-	-	85	-	$\text{dB FS A}$
DAC SNR (-20 dB FS input w/ CCIR-RMS filter on output)	SNR	D-A	-	70	-	-	-	-	dB
Total Harmonic Distortion + Noise (-3 dB FS input signal):									
Line/Alternate Line Output	THD+N	A-A	-	-90	-80	-	-	-74	$\text{dB FS}$
DAC		D-A	-	-87	-80	-	-	-74	$\text{dB FS}$
ADC (all inputs except phone/mic)		A-D	-	-84	-80	-	-	-74	$\text{dB FS}$
ADC (phone/mic)		A-D	-	-85	-74	-	-	-74	$\text{dB FS}$
Power Supply Rejection Ratio (1 kHz, 0.5 $\text{V}_{\text{RMS}}$ w/ 5 V DC offset)(Note 4)			40	60	-	-	40	-	dB
Interchannel Isolation			70	87	-	-	87	-	dB
Spurious Tone (Note 4)			-	-100	-	-	-100	-	$\text{dB FS}$
Input Impedance (Note 4)			10	-	-	10	-	-	$\text{k}\Omega$
External Load Impedance			10	-	-	10	-	-	$\text{k}\Omega$
Output Impedance (Note 4)			-	730	-	-	730	-	$\Omega$
Input Capacitance (Note 4)			-	5	-	-	5	-	$\text{pF}$
Vrefout			2.0	2.3	2.5	2.0	2.3	2.5	V

Notes:

1.  $Z_{\text{AL}}$  refers to the analog output pin loading and  $C_{\text{DL}}$  refers to the digital output pin loading.
2. Parameter definitions are given in the *Parameter and Term Definitions* section.
3. Path refers to the signal path used to generate this data. These paths are defined in the *Parameter and Term Definitions* section.
4. This specification is guaranteed by silicon characterization, it is not production tested.

**MIXER CHARACTERISTICS** (for CS4299-KQ only)

Parameter	Min	Typ	Max	Unit
Mixer Gain Range Span				
Line In, Aux, CD, Video, Mic1 Mic2, Phone, PC Beep, Alternate Line	-	46.5	-	dB
Mono Out	-	46.5	-	dB
Line Out	-	94.5	-	dB
Step Size				
All volume controls except PC Beep	-	1.5	-	dB
PC Beep	-	3.0	-	dB

**ABSOLUTE MAXIMUM RATINGS** (AVss1 = AVss2 = DVss1 = DVss2 = 0 V)

Parameter	Min	Typ	Max	Unit
Power Supplies				
+3.3 V Digital	-0.3	-	6.0	V
+5 V Digital	-0.3	-	6.0	V
Analog	-0.3	-	6.0	V
Total Power Dissipation			TBD	mW
Input Current per Pin			10	mA
Output Current per Pin			15	mA
Analog Input voltage			AVdd + 0.3	V
Digital Input voltage			DVdd + 0.3	V
Ambient Temperature			110	°C
Storage Temperature			150	°C

**RECOMMENDED OPERATING CONDITIONS** (AVss1 = AVss2 = DVss1 = DVss2 = 0 V)

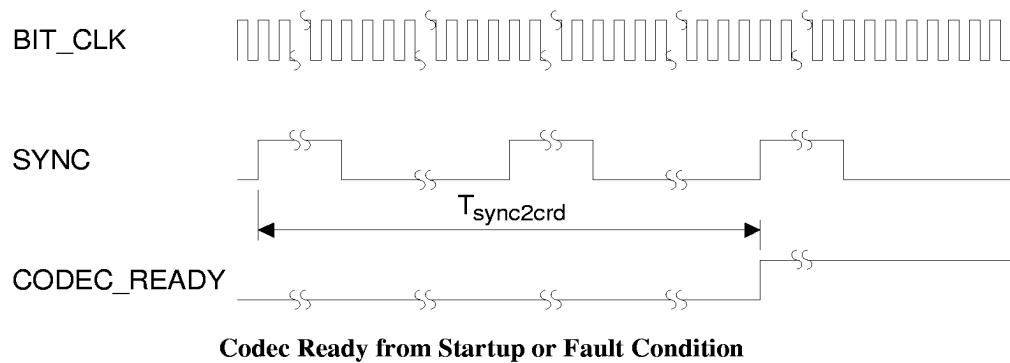
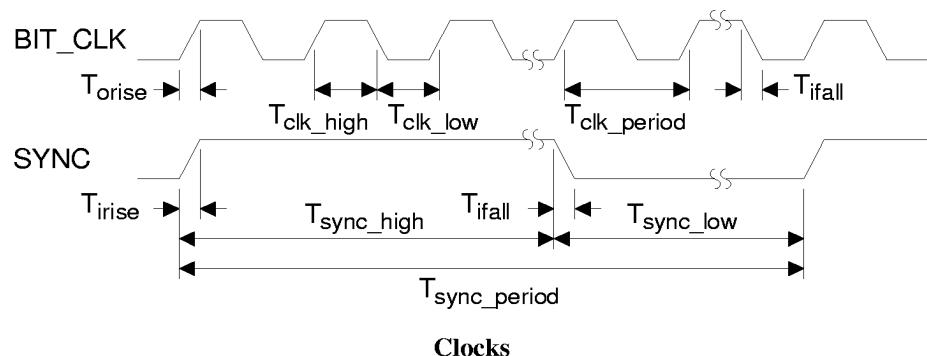
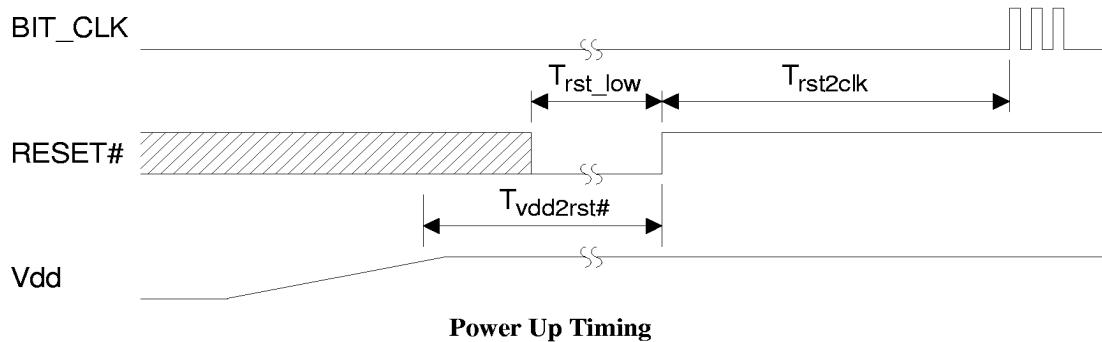
Parameter	Symbol	Min	Typ	Max	Unit
Power Supplies					
+3.3 V Digital	DVdd1, DVdd2	3.135	3.3	3.465	V
+5 V Digital	DVdd1, DVdd2	4.75	5	5.25	V
Analog	AVdd1, AVdd2	4.75	5	5.25	V
Operating Ambient Temperature		0	-	70	°C

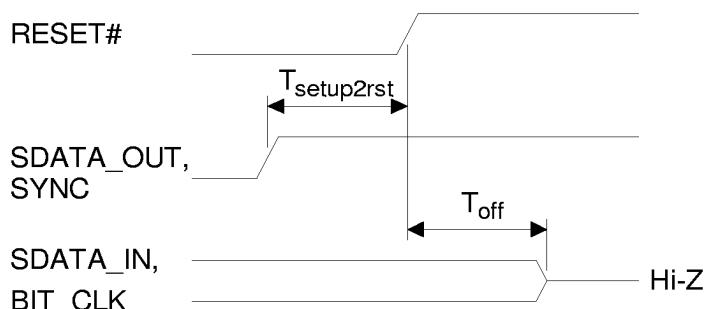
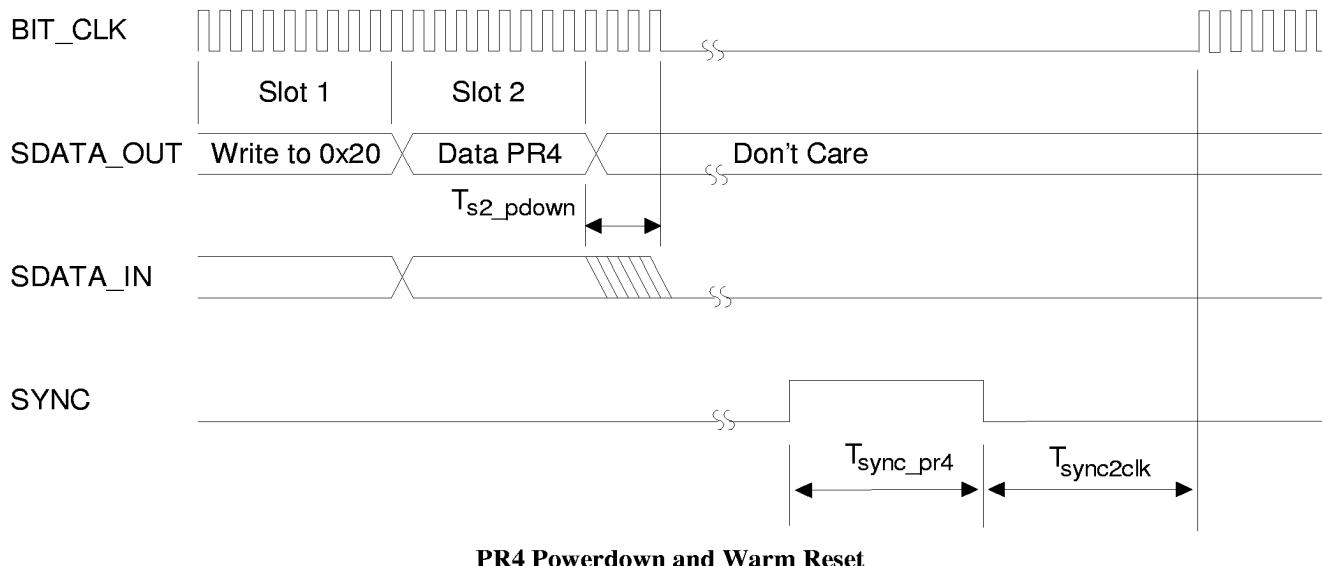
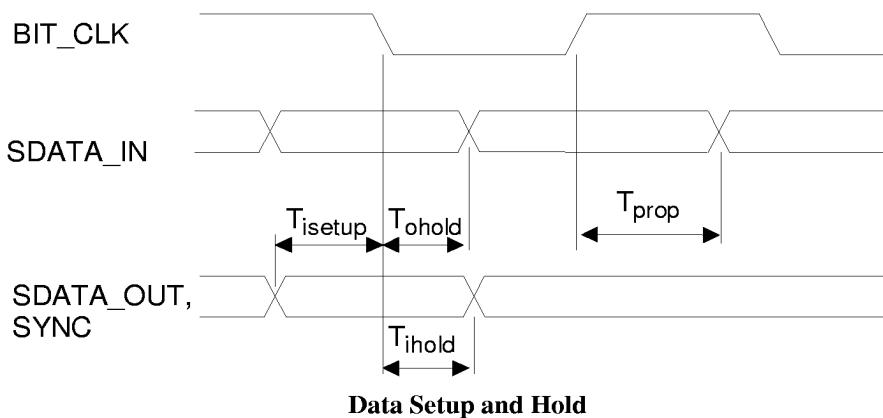
**DIGITAL CHARACTERISTICS** (AVss = DVss = 0 V (See *Grounding and Layout* section))

Parameter	Symbol	Min	Typ	Max	Unit
Low level input voltage	V <sub>il</sub>	-	-	0.35 x DVdd	V
High level input voltage	V <sub>ih</sub>	0.65 x DVdd	-	-	V
High level output voltage	V <sub>oh</sub>	0.90 x DVdd	0.99 x DVdd	-	V
Low level output voltage	V <sub>ol</sub>	-	0.03	0.10 x DVdd	V
Input Leakage Current (AC-link inputs)		-10	-	10	µA
Output Leakage Current (Tri-stated AC-link outputs)		-10	-	10	µA
Output buffer drive current	(Note 4)	-	TBD		µA

**SERIAL PORT TIMING** Standard test conditions unless otherwise noted:  $T_{\text{ambient}} = 25^\circ \text{ C}$ ,  $\text{AVdd} = 5.0 \text{ V}$ ,  $\text{DVdd} = 3.3 \text{ V}$ ;  $C_L = 55 \text{ pF}$  load.

Parameter	Symbol	Min	Typ	Max	Unit
<b>RESET Timing</b>					
RESET# active low pulse width	$T_{\text{rst\_low}}$	1.0	-	-	$\mu\text{s}$
RESET# inactive to BIT_CLK start-up delay	$T_{\text{rst2clk}}$	-	TBD	-	ms
1st SYNC active to CODEC READY set	$T_{\text{sync2rd}}$	-	41.6	-	$\mu\text{s}$
Vdd stable to Reset inactive	$T_{\text{vdd2rst\#}}$		TBD		
<b>Clocks</b>					
BIT_CLK frequency	$F_{\text{clk}}$	-	12.288	-	MHz
BIT_CLK period	$T_{\text{clk\_period}}$	-	81.4	-	ns
BIT_CLK output jitter (depends on XTAL_IN source)		-	-	750	ps
BIT_CLK high pulse width	$T_{\text{clk\_high}}$	36	40.7	45	ns
BIT_CLK low pulse width	$T_{\text{clk\_low}}$	36	40.7	45	ns
SYNC frequency	$F_{\text{sync}}$	-	48	-	kHz
SYNC period	$T_{\text{sync\_period}}$	-	20.8	-	$\mu\text{s}$
SYNC high pulse width	$T_{\text{sync\_high}}$	-	1.3	-	$\mu\text{s}$
SYNC low pulse width	$T_{\text{sync\_low}}$	-	19.5	-	$\mu\text{s}$
<b>Data Setup and Hold</b>					
Output Propagation delay from rising edge of BIT_CLK	$T_{\text{prop}}$	-	6	8	ns
Output hold from falling edge of BIT_CLK	$T_{\text{ohold}}$	5	-	-	ns
Input setup time from falling edge of BIT_CLK	$T_{\text{isetup}}$	10	-	-	ns
Input hold time from falling edge of BIT_CLK	$T_{\text{ihold}}$	0	-	-	ns
Input Signal rise time	$T_{\text{irise}}$	2	-	6	ns
Input Signal fall time	$T_{\text{ifall}}$	2	-	6	ns
Output Signal rise time (Note 4)	$T_{\text{orise}}$	2	4	6	ns
Output Signal fall time (Note 4)	$T_{\text{ofall}}$	2	4	6	ns
<b>Misc. Timing Parameters</b>					
End of Slot 2 to BIT_CLK, SDATA_IN low (PR4)	$T_{\text{s2\_pdown}}$	-	.34	1.0	$\mu\text{s}$
SYNC pulse width (PR4) Warm Reset	$T_{\text{sync\_pr4}}$	1.0	-	-	$\mu\text{s}$
SYNC inactive (PR4) to BIT_CLK start-up delay	$T_{\text{sync2clk}}$	162.8	244	-	ns
Setup to trailing edge of RESET# (ATE test mode) (Note 4)	$T_{\text{setup2rst}}$	15	-	-	ns
Rising edge of RESET# to Hi-Z delay (Note 4)	$T_{\text{off}}$	-	-	25	ns





## 2.0 GENERAL DESCRIPTION

The CS4299 is a mixed-signal serial Codec compliant to the *Audio Codec '97 Specification*, revision 2.1 [1]. It is designed to be paired with a digital interface, referred to as the Controller, which is typically located on the PCI bus or integrated within the system chip set. The Controller is responsible for all communications between the CS4299 and the remainder of the system. The Codec contains two distinct functional sections: Digital and Analog. The Digital section includes the AC-link registers, sample rate converters, power management support, and AC-link serial port interface logic. The analog section includes the analog input multiplexer (mux), stereo output mixer, mono output mixer, stereo ADCs, stereo DACs, and their associated volume controls.

### 2.1 AC-Link

All communication with the Codec is established with a 5-wire digital interface to the Controller chip as shown in Figure 1. All clocking for the serial communication is synchronous to the BIT\_CLK signal. BIT\_CLK is generated by the primary Codec and is used to slave the Controller and any secondary Codecs, if applicable. An AC-link audio frame is a sequence of 256 serial bits organized into 13 groups referred to as 'slots'. One frame consists of one 16-bit slot and twelve 20-bit slots. During each audio frame, data is passed bi-directionally between the Codec and the Controller. The input frame is driven from the Codec on the SDATA\_IN line. The

output frame is driven from the Controller SDATA\_OUT line. Both input and output frames contain the same number of bits and are organized with the same 'slot' configuration. The input and output frame have differing functions for each slot. The Controller synchronizes the beginning of a frame with the SYNC signal. In Figure 2 the position of each bit location within the frame is noted. The first bit position in a new serial data frame is F0 and the last bit position in the serial data frame is F255. When SYNC goes active (high) and is sampled active by the CS4299 (on the falling edge of BIT\_CLK), both devices are synchronized to a new serial data frame. The data on the SDATA\_OUT pin at this clock edge is the final bit of the previous frame's serial data. On the next rising edge of BIT\_CLK, the first bit of Slot 0 is driven by the Controller on the SDATA\_OUT pin. The CS4299 latches in this data, as the first bit of the frame, on the next falling edge of the BIT\_CLK clock signal. The Controller is also responsible for issuing reset via the RESET# signal. After being reset, the Codec is responsible for flagging the Controller that it is ready for operation after synchronizing its internal functions. The AC-link signals may be referenced to either 5 Volts or 3.3 Volts. The CS4299 must use the same digital supply voltage as the Controller chip.

### 2.2 Control registers

All read accesses to the Codec are generated by requesting a register address (index number) in slot 1 of a SDATA\_OUT frame. The following SDATA\_IN frame will contain the register content in its

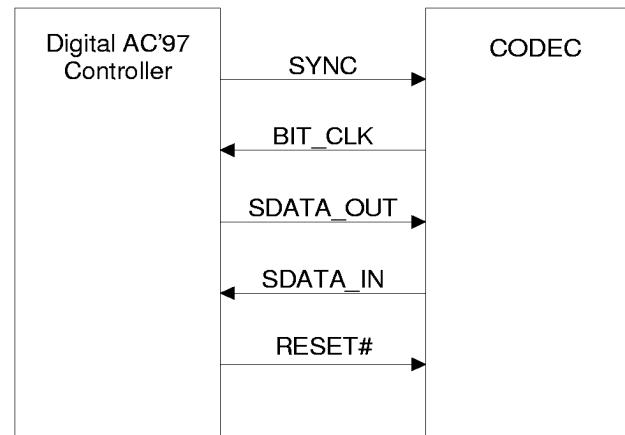
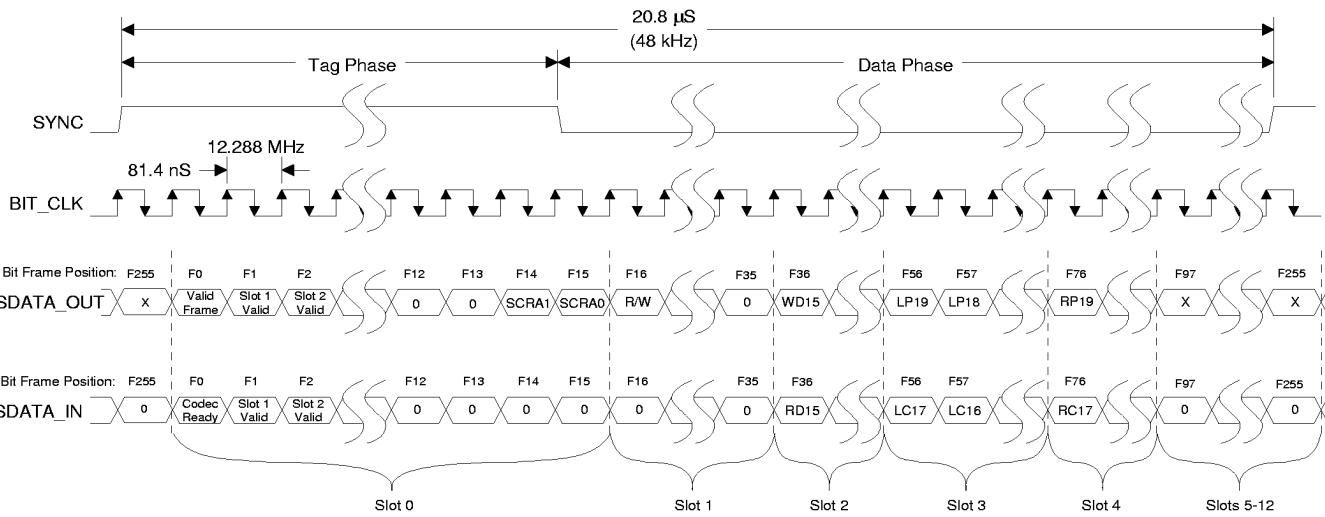


Figure 1. AC-link Connections



**Figure 2. AC-link Input and Output Framing**

slot 2. The write operation is identical with the index in slot 1 and the write data in slot 2. The *AC '97 Frame Definition* section details the function of each input and output frame. Individual register descriptions are found in the *Register Interface* section.

## 2.3 Sample Rate Converters

The sample rate converters (SRC) provide high accuracy digital filters supporting sample frequencies other than 48 kHz to be played or captured from the Codec. In addition to the 48 kHz rate, the CS4299 supports the Windows standard sample rates of 8.0 kHz, 11.025 kHz, 16 kHz, 22.05 kHz, 32 kHz, and 44.1 kHz.

## 2.4 Output Mixer

There are two output mixers on the CS4299 as illustrated in Figure 3. The stereo output mixer sums together the analog inputs to the CS4299 according to the settings in the volume control registers. The mono output mixer generates a monophonic sum of the left and right channels from the stereo input mixer. However, the mono output mixer does not include the PC\_BEEP and PHONE signals which are included in the stereo output mix. The stereo output mix is sent to the LINE\_OUT and ALT\_LINE\_OUT output pins of the CS4299. The mono output mix is sent to the MONO\_OUT output pin on the CS4299.

## 2.5 Input Mux

The input multiplexer controls which analog input is sent to the ADCs. The output of the input mux is converted to stereo 18-bit digital PCM data and sent to the Controller chip via the AC-link SDATA\_IN signal.

## 2.6 Volume Control

The Codec's volume control registers control analog input level to the input mixer, the master volume level, and the alternate volume level. All analog volume controls, except PC\_BEEP, implement controlled volume steps at nominally 1.5 dB per step. PC\_BEEP uses 3 dB steps. The analog inputs

allow a mixing range of +12 dB of signal gain to -34.5 dB of signal attenuation. The analog output volume controls allow from 0 dB to -94.5 dB of attenuation for LINE\_OUT and -46.5 dB for ALT\_LINE\_OUT and MONO\_OUT. The PC\_BEEP input volume control allows from 0 dB to -45 dB of attenuation.

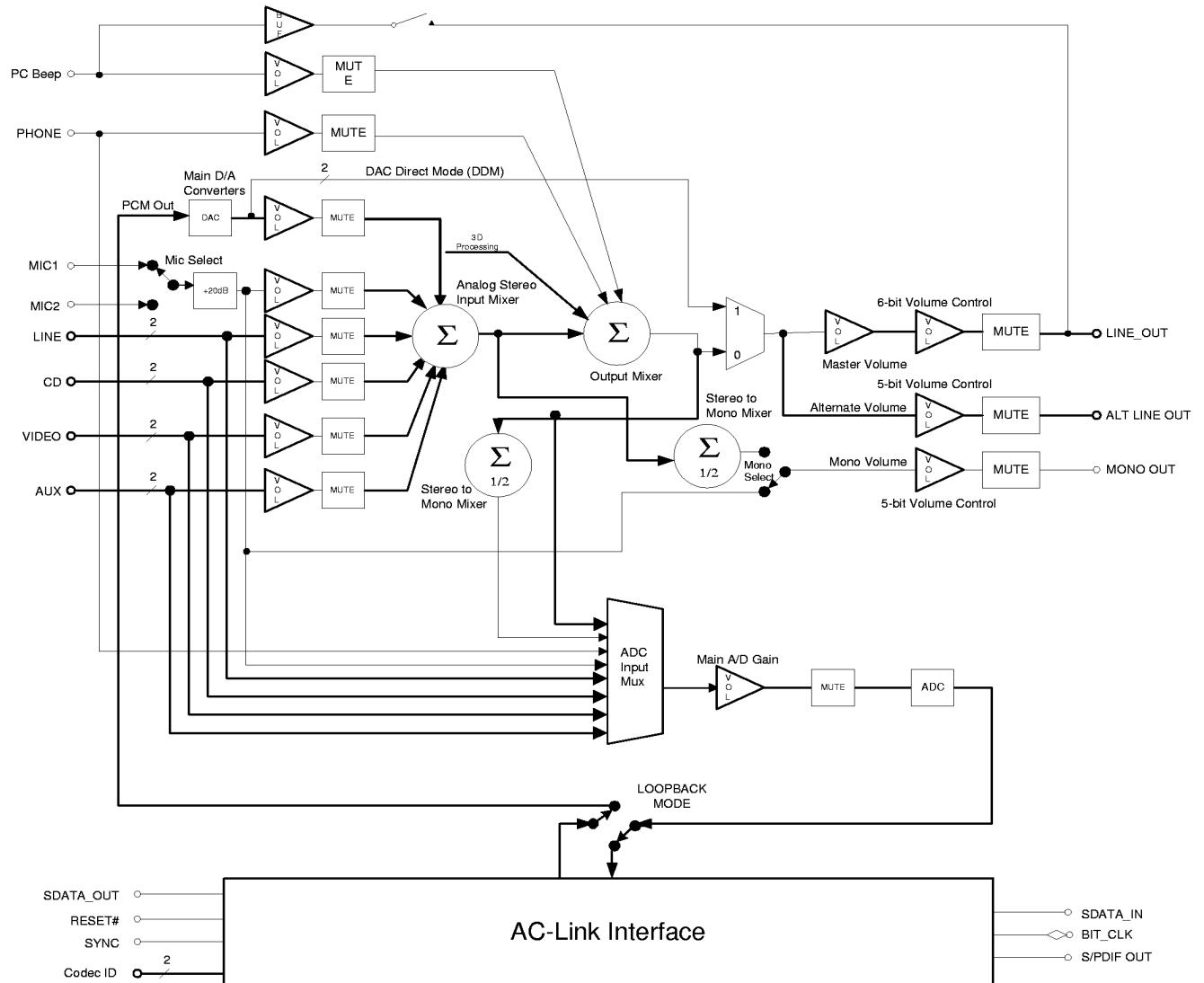


Figure 3. Mixer Diagram

### 3.0 AC '97 FRAME DEFINITION

The AC Link is a bi-directional serial port with thirteen time-division multiplexed slots in each direction. The first slot is 16 bits long and termed the tag slot. Bits in the tag slot determine if the Codec is ready and indicate which, if any, other slots contain valid data. Slots 1 through 12 are 20-bits long and can contain audio data. Slot 11 and Slot 12 are not utilized on the CS4299. The serial data line is defined from the Controller's perspective, NOT from the Audio Codec's perspective.

#### 3.1 AC-Link Serial Data Output Frame

In the serial data output frame, data is passed on the SDATA\_OUT pin TO the CS4299 FROM the Controller. Figure 2 illustrates the serial port timing.

##### *Serial Data Output Slot Tags (Slot 0)*

Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0
Valid Frame	Slot 1 Valid	Slot 2 Valid	Slot 3 Valid	Slot 4 Valid	Slot 5 Valid	Slot 6 Valid	Slot 7 Valid	Slot 8 Valid	Slot 9 Valid	Slot 10 Valid			SCRA 1	SCRA 0	

Valid Frame	Determines if any of the following slots contain either valid playback data for the Codec's DACs or data for read/write operation. When set, at least one of the other AC-link slots contain valid data. If this bit is clear, the remainder of the frame is ignored.
Slot [1:2] Valid	Indicates valid slot data when accessing the register set of the primary Codec (SCRA[1:0] = 00). For a read operation, Slot 1 Valid is set when <i>Register Address</i> (Slot 1) contains valid data. For a write operation, Slot 1 Valid and Slot 2 Valid are set indicating <i>Register Address</i> (Slot 1) and <i>Register Write Data</i> (Slot 2) contain valid data. The register address and write data must be valid within the same frame. SCRA[1:0] must be cleared when accessing the primary Codec. The physical address of a Codec is determined by the ID[1:0]# input pins which are reflected in the <i>Extended Audio ID</i> (Index 28h) register.
Slot [3:10] Valid	If a Slot Valid bit is set, the named slot contains valid audio data. If the bit is clear, the slot will be ignored. The Codec supports alternate slot mapping as defined in the AC '97 2.1 specification [1]. For more information, see the <i>Slot Map</i> (Index 5Eh) register.
SCRA[1:0]	Secondary Codec Register Access. Unlike the primary Codec, SCRA[1:0] indicate valid slot data when accessing the register set of a secondary Codec. The value set in SCRA[1:0] (01,10,11) determines which of the three possible secondary Codecs is accessed. For a read operation, the SCRA[1:0] bits are set when <i>Register Address</i> (Slot 1) contains valid data. For a write operation, SCRA[1:0] bits are set when <i>Register Address</i> (Slot 1) and <i>Register Write Data</i> (Slot 2) contain valid data. The write operation requires the register address and the write data to be valid within the same frame. SCRA[1:0] must be cleared when accessing the primary Codec. They must also be cleared during the idle period where no register read or write is pending. The physical address of a Codec is determined by the ID[1:0]# input pins which are reflected in the <i>Extended Audio ID</i> (Index 28h) register. The SCRA[1:0] bits are listed as the ID[1:0] bits in Slot 0 in the AC '97 specification.

##### *Register Address (Slot 1)*

Bit 19	18	17	16	15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0
R/W	RI6	RI5	RI4	RI3	RI2	RI1	RI0												

R/W	Read/Write. Determines if a read (R/W = 1) or write (R/W = 0) operation is requested. For a read operation, the following Input Frame will return the register index in the <i>Read-Back Address Port</i> (Slot 1) and the contents of the register in the <i>Read-Back Data Port</i> (Slot 2). A write operation does not return any valid data in the following frame. If the R/W bit = 0, data must be valid in both the <i>Register Address</i> (Slot 1) and the <i>Register Write Data</i> (Slot 2) during a frame when Slot [1:2] Valid or SCRA[1:0] are set.
-----	---

RI[6:0] Register index/address. Registers can only be accessed on word boundaries; RI0 must be set to 0. RI[6:0] must contain valid data during a frame when the Slot 1 Valid or SCRA[1:0] are set.

### Register Write Data (Slot 2)

Bit 19	18	17	16	15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0
WD15	WD14	WD13	WD12	WD11	WD10	WD9	WD8	WD7	WD6	WD5	WD4	WD3	WD2	WD1	WD0				

WD[15:0] Codec register data for write operations. For read operations, this data is ignored. If R/W = 0, data must be valid in both the *Register Address* (Slot 1) and the *Register Write Data* (Slot 2) during a frame when the Slot [1:2] Valid = 11 or either SCRA[1:0] bit is set. Splitting the register address and the write data across multiple frames is not permitted.

### Playback Data (Slots 3-10)

Bit 19	18	17	16	15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0
PD19	PD18	PD17	PD16	PD15	PD14	PD13	PD12	PD11	PD10	PD9	PD8	PD7	PD6	PD5	PD4	PD3	PD2	PD1	PD0

PD[19:0] 20-bit PCM playback (2's compliment) data for the left and right DACs and/or the S/PDIF transmitter. Any PCM data from the Controller less than 20 bits should be left justified in the slot and zero-padded. Table 8 on page 24 lists the definition of each respective slot. The mapping of a given slot to a DAC is determined by the state of the ID[1:0] bits found in the *Extended Audio ID* (Index 28h) register and by the SM[1:0] and AMAP bits found in the *Slot Map* (Index 5Eh) register.

## 3.2 AC-Link Audio Input Frame

In the serial data input frame, data is passed on the SDATA\_IN pin FROM the CS4299 to the AC '97 Controller. The data format for the input frame is very similar to the output frame. Figure 2 illustrates the serial port timing.

### Serial Data Input Slot Tag Bits (Slot 0)

Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0
Codec Ready	Slot 1 Valid	Slot 2 Valid	Slot 3 Valid	Slot 4 Valid	Slot 5 Valid	Slot 6 Valid	Slot 7 Valid	Slot 8 Valid	Slot 9 Valid	Slot 10 Valid					

Codec Ready Indicates the readiness of the CS4299's AC-link and Control and Status registers. Immediately after a Cold Reset this bit will be clear. Once the CS4299's clocks and voltages are stable, this bit will be set. Until the Codec Ready bit is set, no AC-link transactions should be attempted by the Controller. The Codec Ready bit does not indicate readiness of the DACs, ADCs, Vref, or any other analog function. Those must be checked in the *Powerdown Control/Status* (Index 26h) register by the Controller before any access is made to the mixer registers. Any accesses to the Codec while Codec Ready is clear is ignored.

#### Slot 1 Valid Tag

Indicates Slot 1 contains a valid read back address.

#### Slot 2 Valid Tag

Indicates Slot 2 contains valid register read data.

#### Slot [3:10] Valid Tag

Indicates Slot [3:10] contains valid capture data from the Codec's ADC.

### Read-Back Address Port (Slot 1)

Bit 19	18	17	16	15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0
	RI6	RI5	RI4	RI3	RI2	RI1	RI0	DRS3	DRS4	DRS5	DRS6	DRS7	DRS8	DRS9	DRS10				

RI[6:0] Register index. The Read-Back Address Port echoes the AC '97 Register address when a register read has been requested in the previous frame. The Codec will only echo the register index for a read access. Write accesses will not return valid data in Slot 1.

DRS[3:10] Disable Request for Slots 3 - 10. When the VRE bit in *Extended Audio Status / Control* (Index 2Ah) register is clear, these bits are always 0. When VRE is set, the SRCs are enabled and the Codec will set these bits when the SRCs do NOT need data. When the Controller sees these bits set, it must NOT send data on SDATA\_OUT in the following frame (not the current frame).

### Read-Back Data Port (Slot 2)

Bit 19	18	17	16	15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0
RD15	RD14	RD13	RD12	RD11	RD10	RD9	RD8	RD7	RD6	RD5	RD4	RD3	RD2	RD1	RD0				

RD[15:0] 16-bit register value. The Read-Back Data Port contains the register data requested by the Controller from the previous read request. All read requests will return the read address in the *Read-Back Address Port* (Slot 1) and the register data in the *Read-Back Data Port* (Slot 2) on the following serial data frame.

### PCM Capture Data (Slot 3-10)

Bit 19	18	17	16	15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0
CD17	CD16	CD15	CD14	CD13	CD12	CD11	CD10	CD9	CD8	CD7	CD6	CD5	CD4	CD3	CD2	CD1	CD0		

CD[17:0] 18-bit PCM (2's compliment) data. The mapping of a given slot to an ADC is determined by the state of the ID[1:0] bits found in the *Extended Audio ID* (Index 28h) register and the SM[1:0] and AMAP bits found in the *Slot Map* (Index 5Eh) register. The definition of each slot can be found in Table 7 on page 23.

The capture data in Slot [3:10] will only be valid when the respective slot valid bit is set in Slot 0.

### 3.3 AC '97 Reset Modes

Three methods of resetting the CS4299, as defined in the AC '97 Specification, are supported: *Cold AC '97 Reset*, *Warm AC '97 Reset*, and *AC '97 Register Reset*. A Cold AC '97 Reset is required to restart the AC-link when bit PR5 is set in the *Powerdown Control/Status* (Index 26h) register.

#### Cold AC '97 Reset

A Cold Reset is performed by asserting RESET# in accordance with the minimum timing specifications in the *Serial Port Timing* section. Once de-asserted, all of the Codec's registers will be reset to their default power-on states and the BIT\_CLK clock and SDATA\_IN signals will be reactivated. The timing of power-up/reset events is discussed in detail in the *Power Management* section.

#### Warm AC '97 Reset

The CS4299 may also be reactivated when the AC-link is powered down (refer to the PR4 bit description in the *Power Management* section) by a Warm Reset. A Warm Reset allows the AC-link to be reactivated without losing information in the Codec's registers. Warm Reset is initiated when the SYNC signal is driven high for at least 1  $\mu$ s and then driven low in the absence of the BIT\_CLK clock signal. The BIT\_CLK clock will not restart until at least 2 normal BIT\_CLK clock periods ( $\pm 162.8$  ns) after the SYNC signal is de-asserted.

### AC '97 Register Reset

The third reset mode provides a register reset to the CS4299. This is available only when the CS4299's AC-link is active and the Codec Ready bit is set. The Register Reset forces all registers to be reset to their default, power-up values. A Register Reset occurs when any value is written to the *Reset* (Index 00h) register.

### 3.4 AC-Link Protocol Violation - Loss of SYNC

The CS4299 is designed to handle SYNC protocol violations. The following are situations where the SYNC protocol has been violated:

- The SYNC signal is not sampled high for exactly 16 BIT\_CLK clock cycles at the start of an audio frame.
- The SYNC signal is not sampled high on the 256th BIT\_CLK clock period after the previous SYNC assertion.
- The SYNC signal goes active high before the 256th BIT\_CLK clock period after the previous SYNC assertion.

Upon loss of synchronization with the Controller, the Codec will mute all analog outputs and clear the Codec Ready bit in the serial data input frame until two valid frames are detected. During this detection period, the Codec will ignore all register reads and writes and will discontinue the transmission of PCM capture data.

## 4.0 REGISTER INTERFACE

Reg Num	Name	D15	D14	D13	D12	D11	D10	D9	D8	D7	D6	D5	D4	D3	D2	D1	D0	Default
00h	Reset		SE4	SE3	SE2	SE1	SE0	0	ID8	ID7	0	0	ID4	0	0	0	0	1990h
02h	Master Volume	Mute		ML5	ML4	ML3	ML2	ML1	ML0				MR5	MR4	MR3	MR2	MR1	MR0
04h	Alternate Line Out Volume	Mute			ML4	ML3	ML2	ML1	ML0				MR4	MR3	MR2	MR1	MR0	8000h
06h	Master Volume Mono	Mute											MM4	MM3	MM2	MM1	MM0	8000h
0Ah	PC_BEEP Volume	Mute											PV3	PV2	PV1	PV0		0000h
0Ch	Phone Volume	Mute											GN4	GN3	GN2	GN1	GN0	8008h
0Eh	Mic Volume	Mute									20dB		GN4	GN3	GN2	GN1	GN0	8008h
10h	Line In Volume	Mute		GL4	GL3	GL2	GL1	GL0					GR4	GR3	GR2	GR1	GR0	8808h
12h	CD Volume	Mute		GL4	GL3	GL2	GL1	GL0					GR4	GR3	GR2	GR1	GR0	8808h
14h	Video Volume	Mute		GL4	GL3	GL2	GL1	GL0					GR4	GR3	GR2	GR1	GR0	8808h
16h	Aux Volume	Mute		GL4	GL3	GL2	GL1	GL0					GR4	GR3	GR2	GR1	GR0	8808h
18h	PCM Out Vol	Mute		GL4	GL3	GL2	GL1	GL0					GR4	GR3	GR2	GR1	GR0	8808h
1Ah	Record Select					SL2	SL1	SL0						SR2	SR1	SR0		0000h
1Ch	Record Gain	Mute			GL3	GL2	GL1	GL0					GR3	GR2	GR1	GR0		8000h
20h	General Purpose		3D				MIX	MS	LPBK									0000h
22h	3D Control													S3	S2	S1	S0	0000h
26h	Powerdown Ctrl/Stat	EAP D	PR6	PR5	PR4	PR3	PR2	PR1	PR0					REF	ANL	DAC	ADC	000Fh
28h	Extended Audio ID	ID1	ID0					AM AP										VRA 0201h
2Ah	Extended Audio Ctrl																	VRE 0000h
2Ch	PCM Front DAC Rate	SR 15	SR 14	SR 13	SR 12	SR 11	SR 10	SR9	SR8	SR7	SR6	SR5	SR4	SR3	SR2	SR1	SR0	BB80h
32h	PCM Left/Right ADC Rate	SR 15	SR 14	SR 13	SR 12	SR 11	SR 10	SR9	SR8	SR7	SR6	SR5	SR4	SR3	SR2	SR1	SR0	BB80h
<b>Cirrus Defined Registers:</b>																		
5E	Slot Map Register									DD M	AME N		SM1	SM0				0080h
68	S/PDIF Enable	SPE N	V		Fs	L	CC6	CC5	CC4	CC3	CC2	CC1	CC0	Pre	Cop y	#Au dio	Pro	0000h
7Ch	Vendor ID1(CR)	F7	F6	F5	F4	F3	F4	F1	F0	S7	S6	S5	S4	S3	S2	S1	S0	4352h
7Eh	Vendor ID2(Y-)	T7	T6	T5	T4	T3	T2	T1	T0		PID2	PID1	PID0		RID2	RID1	RID0	5931h

**Table 1. Mixer Registers**

*Reset (Index 00h)*

D15	D14	D13	D12	D11	D10	D9	D8	D7	D6	D5	D4	D3	D2	D1	D0
	SE4	SE3	SE2	SE1	SE0	0	ID8	ID7	0	0	ID4	0	0	0	0

SE[4:0] 3D Stereo Enhancement Technique.

00110 - Crystal 3D Stereo Enhancement.

ID8 set 18-bit ADC resolution.

ID7 set 20-bit DAC resolution.

ID4 set Headphone out support. (Alternate Line Output)

Read-only data 1990h

Any write to this register causes a Register Reset of the Codec's registers forcing all registers to their default state. Reads return configuration information about the Codec.

*Master Volume (Index 02h)*

D15	D14	D13	D12	D11	D10	D9	D8	D7	D6	D5	D4	D3	D2	D1	D0
Mute		ML5	ML4	ML3	ML2	ML1	ML0			MR5	MR4	MR3	MR2	MR1	MR0

Mute Master mute for the LINE\_OUT\_L and the LINE\_OUT\_R output signals.

ML[5:0] Master Volume control for LINE\_OUT\_L pin. Least significant bit represents -1.5 dB with 00000 = 0 dB. The total range is 0 dB to -94.5 dB.

MR[5:0] Master Volume control for LINE\_OUT\_R pin. Least significant bit represents -1.5 dB with 00000 = 0 dB. The total range is 0 dB to -94.5 dB.

Default 8000h, corresponding to 0 dB attenuation and mute on.

*Alternate Volume (Index 04h)*

D15	D14	D13	D12	D11	D10	D9	D8	D7	D6	D5	D4	D3	D2	D1	D0
Mute		ML5	ML4	ML3	ML2	ML1	ML0			MR5	MR4	MR3	MR2	MR1	MR0

Mute Master mute for the ALT\_LINE\_OUT\_L and the ALT\_LINE\_OUT\_R output signals.

ML[4:0] Master Volume control for ALT\_LINE\_OUT\_L pin. Least significant bit represents -1.5 dB with 00000 = 0 dB. The total range is 0 dB to -46.5 dB.

ML5 Setting ML5 sets the left channel attenuation to -46.5 dB by forcing ML[4:0] to a 1 state. ML[5:0] will read back 01111 when ML5 has been set. See Table 2.

MR[4:0] Master Volume control for ALT\_LINE\_OUT\_R pin. Least significant bit represents -1.5 dB with 00000 = 0 dB. The total range is 0 dB to -46.5 dB.

MR5 Setting MR5 sets the right channel attenuation to -46.5 dB by forcing MR[4:0] to a 1 state. MR[5:0] will read back 011111 when MR5 has been set. See Table 2.

Default 8000h, corresponding to 0 dB attenuation and mute on.

ML[5:0]/MR[5:0]/MM[5:0] Write	ML[5:0]/MR[5:0]/MM[5:0] Read	Gain Level
000000	000000	0 dB
000001	000001	-1.5 dB
...	...	...
011111	011111	-46.5 dB
...	...	...
1xxxxx	011111	-46.5 dB

Table 2. Alternate Line-Out and Master Mono Attenuation

**Master Mono Volume (Index 06h)**

D15	D14	D13	D12	D11	D10	D9	D8	D7	D6	D5	D4	D3	D2	D1	D0
Mute										MM5	MM4	MM3	MM2	MM1	MM0

Mute When set, mutes the MONO\_OUT signal.

MM[4:0] Master Mono Attenuation. Least significant bit represents -1.5 dB with 00000 = 0 dB. The total range is 0 dB to -46.5 dB.

MM5 Setting MM5 sets the master mono attenuation to -46.5 dB by forcing MM[4:0] to a 1 state. MM[5:0] will read back 011111 when MM5 has been set. See Table 2.

Default 8000h, corresponding to 0 dB attenuation and Mute set.

**PC\_BEEP Volume (Index 0Ah)**

D15	D14	D13	D12	D11	D10	D9	D8	D7	D6	D5	D4	D3	D2	D1	D0
Mute										PV3	PV2	PV1	PV0		

Mute When set, mutes the PC\_BEEP signal.

PV[3:0] Volume Control for PC\_BEEP pin. Least significant bit represents -3 dB with 0000 = 0 dB. The total range is 0 dB to -45 dB.

Default 0000h, Unmuted, with 0 dB attenuation after the CS4299 is removed from the reset state.

**Phone Volume (Index 0Ch)**

D15	D14	D13	D12	D11	D10	D9	D8	D7	D6	D5	D4	D3	D2	D1	D0
Mute										GN4	GN3	GN2	GN1	GN0	

Mute When set mutes the Phone signal.

GN[4:0] Phone Volume Control. Least significant bit represents 1.5 dB with 01000 = 0 dB. The total range is 12 dB to -34.5 dB.

Default 8008h, 0 dB attenuation and Mute set.

**Microphone Volume (Index 0Eh)**

D15	D14	D13	D12	D11	D10	D9	D8	D7	D6	D5	D4	D3	D2	D1	D0
Mute									20dB		GN4	GN3	GN2	GN1	GN0

Mute When set, mutes MIC1/MIC2 signal.

GN[4:0] MIC1/MIC2 Volume Control. Least significant bit represents 1.5 dB with 01000 = 0 dB. The total range is 12 dB to -34.5 dB.

20dB Enables 20 dB microphone gain block.

Default 8008h, 0 dB attenuation and Mute set.

This register controls the gain level of the Microphone input source to the Input Mixer. It also controls the +20 dB gain block which connects to the input volume control and to the Input Record Mux. The selection of MIC1 or MIC2 input pins is controlled by the MS bit in the *General Purpose* (Index 20h) register. The gain mapping for this register is shown in Table 3.

GN4 - GN0	Gain Level	Mic Gain with 20dB = 1
00000	+12.0 dB	+32.0 dB
00001	+10.5 dB	30.5 dB
...	...	...
00111	+1.5 dB	21.5 dB
01000	0.0 dB	20.0 dB
01001	-1.5 dB	18.5 dB
...	...	...
11111	-34.5 dB	-14.5 dB

**Table 3. Analog Mixer Input Gain Values**

### Stereo Analog Mixer Input Gain (Index's 10h - 18h)

D15	D14	D13	D12	D11	D10	D9	D8	D7	D6	D5	D4	D3	D2	D1	D0
Mute			GL4	GL3	GL2	GL1	GL0				GR4	GR3	GR2	GR1	GR0

Mute When set mutes the respective input. Setting this bit mutes both right and left inputs.

GL[4:0] Left Volume Control. Least significant bit represents 1.5 dB with 01000 = 0 dB. The total range is 12 dB to -34.5 dB. See Table 3.

GR[4:0] Right Volume Control. Least significant bit represents 1.5 dB with 01000 = 0 dB. The total range is 12 dB to -34.5 dB. See Table 3.

Default 8808h, 0 dB gain with Mute enabled.

These registers control the gain levels of the analog input sources to the Input Mixer. The analog inputs associated with registers 10h-18h are found in Table 4.

Register Index	Function
10h	Line IN Volume
12h	CD Volume
14h	Video Volume
16h	Aux Volume
18h	PCM Out Volume

Table 4. Stereo Volume Register Index

### Input Mux Select (Index 1Ah)

D15	D14	D13	D12	D11	D10	D9	D8	D7	D6	D5	D4	D3	D2	D1	D0
					SL2	SL1	SL0						SR2	SR1	SR0

SL[2:0] Left Channel ADC input source select.

SR[2:0] Right Channel ADC input source select.

Default 0000h, MIC inputs selected for both channels.

When capturing PCM data, this register controls the input MUX for the ADCs. Table 5 below lists the possible values for each input.

Sx2 - Sx0	Record Source
0	MIC
1	CD Input
2	Video Input
3	AUX Input
4	Line Input
5	Stereo Mix
6	Mono Mix
7	Phone Input

Table 5. Input Mux Selection

### Record Gain (Index 1Ch)

D15	D14	D13	D12	D11	D10	D9	D8	D7	D6	D5	D4	D3	D2	D1	D0	
Mute				GL3	GL2	GL1	GL0						GR3	GR2	GR1	GR0

Mute When set, mutes the input to the ADCs.

GL[3:0] Left ADC gain. Least significant bit represents +1.5 dB with 0000 = 0 dB. The total range is 0 dB to +22.5 dB.

GR[3:0] Right ADC gain. Least significant bit represents +1.5 dB with 0000 = 0 dB. The total range is 0 dB to +22.5 dB.

Default 8000h, 0 dB gain with Mute on.

**General Purpose (Index 20h)**

D15	D14	D13	D12	D11	D10	D9	D8	D7	D6	D5	D4	D3	D2	D1	D0
		3D				MIX	MS	LPBK							

MIX Mono Output Path. When clear, the Mono Mix out (a mix of the 5 analog stereo sources plus PCM\_OUT) is selected for MONO\_OUT. When set, the MIC path is sent to Mono Out.

MS Microphone Select. Determines which of the two MIC inputs are passed to the mixer. When set, MIC2 input is selected; when clear MIC1 is selected.

LPBK Loopback. If set, enables ADC/DAC Loopback Mode.

3D 3D Enable. If set, enables the CrystalClear 3D stereo enhancement.

Default 0000h.

**3D Control (Index 22h)**

D15	D14	D13	D12	D11	D10	D9	D8	D7	D6	D5	D4	D3	D2	D1	D0
												S3	S2	S1	S0

S[3:0] Spacial Enhancement Depth. Spacial Enhancement is enabled by the 3D bit in the *General Purpose* (Index 20h) register.  
 0000 - No spacial enhancement.  
 1111 - Full special enhancement.

Default 0000h, no spacial enhancement added.

*Powerdown Control/Status (Index 26h)*

D15	D14	D13	D12	D11	D10	D9	D8	D7	D6	D5	D4	D3	D2	D1	D0
EAPD	PR6	PR5	PR4	PR3	PR2	PR1	PR0					REF	ANL	DAC	ADC

EAPD	External Amplifier Power Down. The EAPD pin follows this bit. Generally used to power-down external amplifiers.
PR6	When set, the alternate line-out buffer is powered down.
PR5	When set, the internal master clock is disabled. The only way to recover from setting this bit is through a cold AC '97 reset (driving the RESET# signal active).
PR4	When set, the AC link is powered down. The AC link can be restarted through a warm AC '97 reset using the SYNC signal, or a cold AC '97 reset using the RESET# signal (the primary codec only).
PR3	When set, the analog mixer and voltage reference are powered down. When clearing this bit, the ANL, ADC, and DAC bits should be checked before writing any mixer registers.
PR2	When set, the analog mixer is powered down (the voltage reference is still active). When clearing this bit, the ANL bit should be checked before writing any mixer registers.
PR1	When set, the DACs are powered down. When clearing this bit, the DAC bit should be checked before sending any data to the DACs.
PR0	When set, the ADCs and the ADC input muxes are powered down. When clearing this bit, no valid data will be sent down the AC link until the ADC bit goes high.
REF	Voltage Reference Ready Status. When set, indicates the voltage reference is at a nominal level.
ANL	Analog Ready Status. When set, the analog output mixer, input multiplexer, and volume controls are ready. When clear, no volume control registers should be written.
DAC	DAC Ready Status. When set, the DACs are ready to receive data across the AC link. When clear, the DACs will not accept any valid data.
ADC	ADC Ready Status. When set, the ADCs are ready to send data across the AC link. When clear, no data will be sent to the Controller.
Default	0000h, all blocks are powered on. The lower four bits will eventually change as the Codec finishes an initialization and calibration sequence.

The PR[6:0] and the EAPD bits are power-down control for different sections of the Codec as well as external amplifiers. The REF, ANL, DAC, and ADC bits are status bits which, when set, indicate that a particular section of the Codec is ready. After the Controller receives the Codec Ready bit in Slot 0, these status bits must be checked before writing to any mixer registers.

*Extended Audio ID (Index 28h)*

D15	D14	D13	D12	D11	D10	D9	D8	D7	D6	D5	D4	D3	D2	D1	D0
ID1	ID0					AMAP									VRA

ID[1:0]	Codec configuration ID. Primary is 00; Secondary is 01,10,or 11. This is a reflection of the ID[1:0]# configuration pins. The state of the ID# pins are determined at power-up and are the inverse of the ID bits in this register.
AMAP	AC Link Mapping. When set, this device supports the AC '97, revision 2.1, AC-link slot to audio DAC mapping. This is a reflection of the AMEN bit in the <i>Slot Map</i> (Index 5Eh) register.
VRA	Variable Rate Audio. This bit is set by default for the CS4299 indicating variable sample rates are supported.

Read-only data x201h. Where x is determined by the state of ID[1:0] input pins.

**Extended Audio Status / Control (Index 2Ah)**

D15	D14	D13	D12	D11	D10	D9	D8	D7	D6	D5	D4	D3	D2	D1	D0	VRE

**VRE** Variable Rate Audio Enable. When set, the VRE bit enables access to the sample rate registers *PCM FRONT DAC Rate* (Index 2Ch) register and *PCM LR ADC Rate* (Index 32h) register. When cleared, all accesses to the sample rate registers are ignored and will force the default value of the two PCM registers to BB80h, corresponding to 48 kHz sample rate. VRE is listed as the VRA bit in the *Extended Audio Status Control* register in the AC '97 specification.

**Default** 0000h.

**PCM Front DAC Rate (Index 2Ch)**

D15	D14	D13	D12	D11	D10	D9	D8	D7	D6	D5	D4	D3	D2	D1	D0	
SR15	SR14	SR13	SR12	SR11	SR10	SR9	SR8	SR7	SR6	SR5	SR4	SR3	SR2	SR1	SR0	

**SR[15:0]** Front DAC Sample Rate. Can only be written when the VRE bit of the *Extended Audio Status / Control* (Index 2Ah) register is set. If the VRE bit is clear, all writes are ignored and the register reads back BB80h; corresponding to a 48 kHz sample rate. If the VRE bit is set, the seven standard sample rates are available. If a value not listed in Table 6 is written, the closest sample rate is chosen. Writing a non-supported value will reflect the closest supported rate, not the written value.

**Default** BB80h, indicating 48 kHz sample rate..

Sample rate (Hz)	SR[15:0]	SR[15:12] decode
8,000	1F40	0 or 1
11,025	2B11	2
16,000	3E80	3
22,050	5622	4 or 5
32,000	7D00	6 or 7
44,100	AC44	8,9,Ah
48,000	BB80	Bh,Ch,Dh,Eh,Fh

**Table 6. Standard Sample Rates**

**PCM LR ADC Rate (Index 32h)**

D15	D14	D13	D12	D11	D10	D9	D8	D7	D6	D5	D4	D3	D2	D1	D0	
SR15	SR14	SR13	SR12	SR11	SR10	SR9	SR8	SR7	SR6	SR5	SR4	SR3	SR2	SR1	SR0	

**SR[15:0]** Left / Right ADC Sample Rate. If the VRE bit is clear, all writes are ignored and the register reads back BB80h; corresponding to a 48 kHz sample rate. If the VRE bit is set, the seven standard sample rates are available. If a value not listed in Table 6 is written, the closest sample rate is chosen. Writing a non-supported value will reflect the closest supported rate, not the written value.

**Default** BB80h, indicating 48 kHz sample rate.

*Slot Map (Index 5Eh)*

D15	D14	D13	D12	D11	D10	D9	D8	D7	D6	D5	D4	D3	D2	D1	D0
							DDM	AMEN		SM1	SM0				

DDM	DAC Direct Mode. This bit controls the source to the line and alternate line output drivers. When set, the Left and Right DAC directly drive the line and alternate line outputs by bypassing the audio mixer. When clear, the audio mixer is the source for the line and alternate line outputs.
AMEN	Alternate Slot Map Enable. This bit determines if the CS4299 responds to the Codec ID to slot mapping. If clear, the slot mapping will be determined by the state of the SM[1:0] bits. If set, the slot mapping will be determined by the state of the ID[1:0] bits of the <i>Extended Audio ID</i> (Index 28h) register.
SM[1:0]	Slot Map. Determine which AC-link Slots the ADC and DAC data are transferred through. See Table 7 and Table 8.
Default	0080h

AMEN, SM[1:0] and the ID[1:0] bits of the *Extended Audio ID* (Index 28h) register define the slots assigned for the capture ADCs and the playback DACs. The capture slot assignments are listed in Table 7 and the playback slot assignments are in Table 8.

Slot	Frame definition	ID1	ID0	AMAP*	SM1	SM0	ADC Mapping
3	Left Channel PCM Capture Data	X	X	0	0	0	Left
		0	X	1	X	X	Left
4	Right Channel PCM Capture Data	X	X	0	0	0	Right
		0	X	1	0	0	Right
5	Right Channel PCM Capture Data	X	X	0	0	1	Left
6	Microphone	X	X	0	0	1	Right
		1	1	1	X	X	Left
7	Left Channel PCM Capture Data	X	X	0	1	0	Left
		1	0	1	X	X	Left
8	Right Channel PCM Capture Data	X	X	0	1	0	Right
		1	0	1	X	X	Right
9	Left Channel PCM Capture Data	X	X	0	1	1	Left
		1	1	1	X	X	Right
10	Right Channel PCM Capture Data	X	X	0	1	1	Right
11	Unused						
12							

\* AMAP is controlled by AMEN located in *Slot Map* (Index 5Eh) register.

**Table 7. Capture Slot Assignments**

Slot	Frame definition	ID1	ID0	AMAP*	SM1	SM0	DAC Mapping
3	Left Channel PCM Playback Data	X	X	0	0	0	Left
		0	X	1	X	X	Left
4	Right Channel PCM Playback Data	X	X	0	0	0	Right
		0	X	1	X	X	Right
5	Modem Line 1 PCM Output Data	X	X	0	0	1	Left
6	PCM Center Channel Output Data	X	X	0	0	1	Right
		1	1	1	X	X	Left
7	PCM Left Surround Channel Output Data	X	X	0	1	0	Left
		1	0	1	X	X	Left
8	PCM Right Surround Channel Output Data	X	X	0	1	0	Right
		1	0	1	X	X	Right
9	PCM Low Frequency Effects Output Data	X	X	0	1	1	Left
		1	1	1	X	X	Right
10	Modem Line 1 PCM Output Data	X	X	0	1	1	Right
11	Unused						
12	Unused						

\* AMAP is controlled by AMEN located in *Slot Map* (Index 5Eh) register.

**Table 8. Playback Slot Assignment**

#### S/PDIF Control (Index 68h)

D15	D14	D13	D12	D11	D10	D9	D8	D7	D6	D5	D4	D3	D2	D1	D0
SPEN	V	0	0	L	CC6	CC5	CC4	CC3	CC2	CC1	CC0	Pre	Copy	#Audio	

Please contact Crystal Semiconductor for additional information on the S/PDIF Control Register.

Default 0000h.

#### Vendor ID1 (Index 7Ch)

D15	D14	D13	D12	D11	D10	D9	D8	D7	D6	D5	D4	D3	D2	D1	D0
F7	F6	F5	F4	F3	F2	F1	F0	S7	S6	S5	S4	S3	S2	S1	S0

F[7:0] First Character of Vendor ID.  
 43h - ASCII 'C' character.

S[7:0] Second Character of Vendor ID.  
 52h - ASCII 'R' character.

Read-only data 4352h.

*Vendor ID2 (Index 7Eh)*

D15	D14	D13	D12	D11	D10	D9	D8	D7	D6	D5	D4	D3	D2	D1	D0
T7	T6	T5	T4	T3	T2	T1	T0		PID2	PID1	PID0		RID2	RID1	RID0

T[7:0] Third Character of Vendor ID.  
59h - ASCII 'Y' character.

PID[2:0] Part ID.  
011 - CS4299.

RID[2:0] Revision.  
001 - Revision 'A'.

Read-only data 5931h.

The two Vendor ID registers provide a means to determine the manufacturer of the AC '97 Codec. The first three bytes of the ID registers contain the ASCII code for the first 3 letters of Crystal (CRY). The final byte of the Vendor ID2 register is divided into a Part ID field and a Revision field. Table 9 lists the Part ID's defined to date.

CID2-CID0	Part Name
000	CS4297
001	CS4297A
010	CS4294/CS4298
011	CS4299

**Table 9. Reg. 7Eh Defined Part ID's**

## 5.0 POWER MANAGEMENT

The *Powerdown Control/Status* register (Index 26h) controls the power management functions. Seven of these bits (bits 14:8) have defined functions. In effect, all portions of the Codec can be shut down individually and powered back up by a single cold or warm reset sequence. Table 10 shows the mapping of the power control bits to the functions they manage:

PR Bit	Function
PR0	Main ADC's and Input Mux Powerdown
PR1	Main DAC's Powerdown
PR2	Analog Mixer Powerdown (Vref on)
PR3	Analog Mixer Powerdown (Vref off)
PR4	AC-link Powerdown (BIT_CLK off)*
PR5	Internal Clock Disable
PR6	Alternate Line Out Buffer Powerdown

\* Applies only to primary Codec

**Table 10. Powerdown PR Bit Functions**

When, for example, PR0 is set, the main ADC's and the Input Mux are shut down and the ADC bit (bit 0 in the *Powerdown Control/Status* (Index 26h) register) is cleared indicating the ADCs are no longer in a ready state. The same is true for the DACs, the analog mixers, and the reference voltage (Vrefout). When the PR2 or PR3 bit of the mixer is cleared, the mixer section will begin a power-on process, and the corresponding powerdown status bit will be set when the hardware is ready.

Bit PR4, which shuts down the AC-link, causes the primary Codec to turn off the BIT\_CLK and drive SDATA\_IN low. It also ignores SYNC and SDATA\_OUT in their normal capacities. To restore operation to the part from this state, either a cold or a warm reset is required (see *Cold AC '97 Reset* and *Warm AC '97 Reset* sections). A cold reset will restore all mixer registers to their power-on default values. A warm reset will not alter the values of any mixer register (with the exception of clearing the PR4 bit of *Powerdown Control/Status* (Index 26h) register).

The PR5 bit is a global Codec powerdown that forces all internal clocks to shut down. A cold reset is the only way to restore operation to the CS4299 after a global powerdown.

The CS4299 does not automatically mute any input or output when the powerdown bits are set. The software driver controlling the AC '97 device must manage muting the input and output analog signals before putting the part into any power management state.

## 6.0 ANALOG HARDWARE DESCRIPTION

The analog hardware consist of four line-level stereo inputs, two selectable mono microphone inputs, two mono inputs, a mono output, and dual, independent stereo line outputs. This section describes the analog hardware needed to interface with these pins.

### 6.1 Line-Level Inputs

The analog inputs consist of four stereo analog inputs and four mono inputs. As shown in Figure 3 on page 11, the input to the ADCs comes from the Input Mux which selects one of the following: Phone (Mono), Aux, Video, CD, Mic1 or Mic2 (Mono), Line, Stereo Output Mix, or the Mono Out-

put Mix (Mono). Unused analog inputs should be connected together and then connected through a capacitor to analog ground or tied to the Vrefout line directly.

The analog input mixer is designed to accommodate five stereo inputs and one mono input. These inputs are: a stereo line-level input (LINE), a mono microphone input (MIC), a stereo CD-ROM input (CD), a stereo auxiliary line-level input (AUX), and the PCM output from the DACs. Each of the stereo inputs has separate volume controls for each channel and one mute control for each left/right pair. The mono microphone input has one mute and one volume control. All analog inputs to the CS4299, including CD\_GND, should be capacitively coupled to the input pins. Since many analog levels can be as large as 2 V<sub>RMS</sub>, the circuit shown in Figure 4 can be used to attenuate the analog input by 6 dB (to 1 V<sub>RMS</sub>) which is the maximum voltage allowed for all the stereo line-level inputs.

The CD line-level inputs have an extra pin, CD\_GND, which provides a pseudo-differential input for both CD\_L and CD\_R. This pin takes the common-mode noise out of the CD inputs when connected to the ground coming from the CD analog source. Connecting the CD pins as shown in Figure 5 provides extra attenuation of common mode noise coming from the CDROM drive, thereby producing a higher quality signal. One percent resistors are recommended since the better the resistors match, the better the common-mode attenuation of unwanted signals. If CD is not used, the inputs should be connected through AC capacitors to analog ground or connected to Vrefout.

## 6.2 Microphone Level Inputs

The microphone level inputs, MIC1 and MIC2, include a selectable -34.5 dB to +12 dB gain stage for interfacing to an external microphone. An additional 20 dB gain block is also available. Figure 6 illustrates a single-ended microphone input buffer circuit that will support lower gain dynamic microphones, and phantom-powered microphones that use the right channel (ring) of the jack for power.

## 6.3 Mono Inputs

The mono input, PC\_BEEP, is useful for mixing the output of the “beeper” (timer chip), provided in all PCs, with the rest of the audio signals. When the part is held in reset, the CS4299 passes the PC\_BEEP input directly to the line output. This allows the system sounds or “beep” to be available before the AC '97 interface has been activated. This feature is controlled by the BEEP\_EN pin. By default, (the BEEP\_EN pin unconnected or pulled high) the CS4299 enables the PC\_BEEP to line out path while the RESET pin is active. The attenuation control allows 16 levels in -3 dB steps. In addition, a mute control is provided. The attenuator is a single channel block with the resulting signal sent to the output mixer where it is mixed with the left and right outputs. Figure 7 illustrates a typical

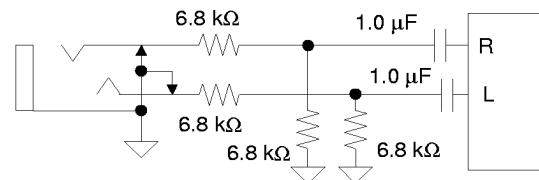


Figure 4. Line Inputs

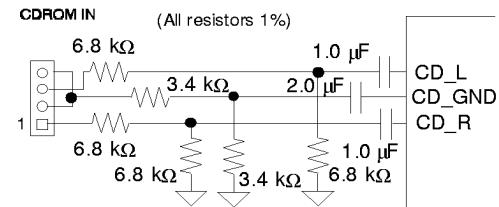
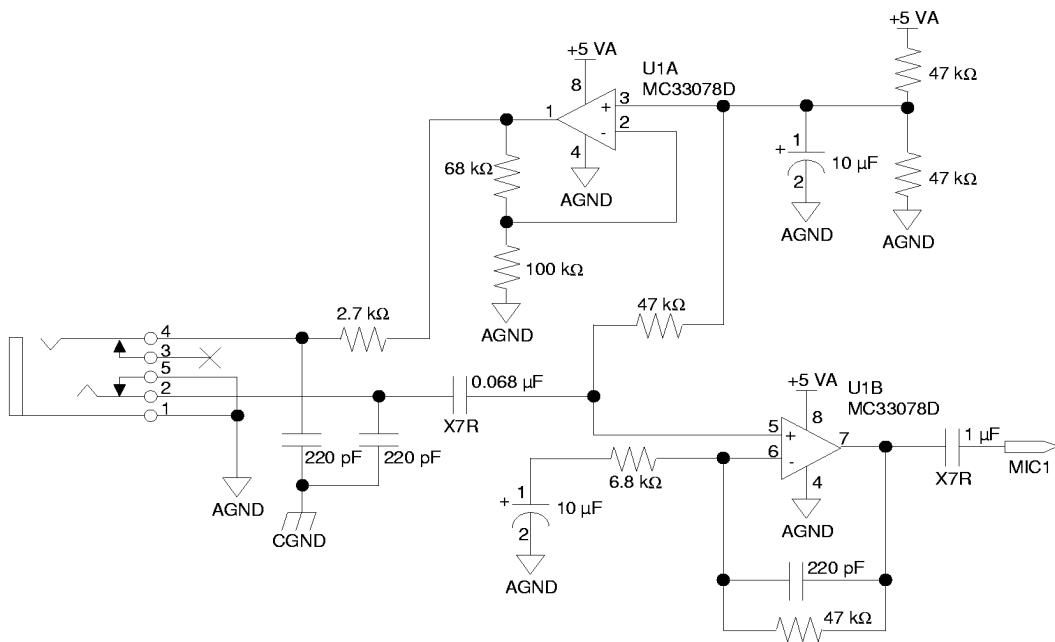


Figure 5. Differential CDROM In



**Figure 6. PC '99 Microphone Pre-amplifier**

input circuit for the PC\_BEEP input. If PC\_BEEP is driven from a CMOS gate, the  $4.7\text{ k}\Omega$  should be tied to analog ground instead of VA+. Although this input is described for a low-quality beeper, the input is of the same high-quality as all other analog inputs and may be used for other purposes.

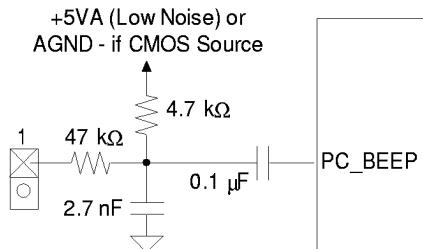
The mono input, PHONE, can be used to interface to the output of a modem analog front end (AFE) chip so that modem dialing signals and protocol negotiations may be monitored through the audio system. Like all other analog inputs, this pin must be AC coupled and the input signal must be limited to  $1\text{ V}_{\text{RMS}}$ .

#### 6.4 Line Level Outputs

The analog output section provides a stereo line-level output and an alternate stereo line-level output. LINE\_OUT\_L, LINE\_OUT\_R, ALT\_LINE\_OUT\_L, and ALT\_LINE\_OUT\_R outputs should be capacitively coupled to external circuitry.

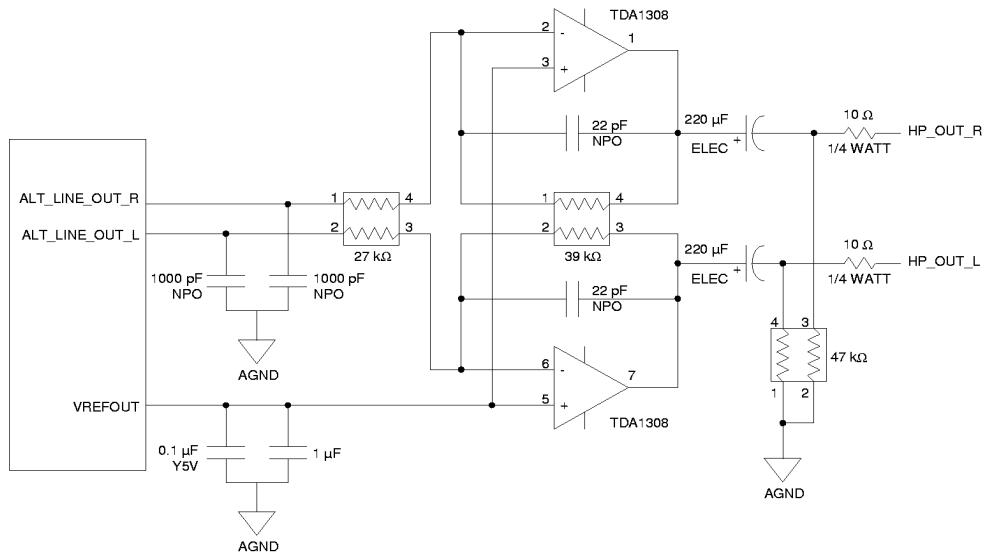
The mono output, MONO\_OUT, can be either a sum of the left and right output channels, attenuated by 6 dB to prevent clipping at full scale, or the selected MIC\_IN signal. The mono out channel can drive the PC internal mono speaker using an appropriate buffer circuit. The mute control is independent of the line outputs allowing the mono channel to mute the speaker without muting the line outputs.

Each of the 5 analog outputs, if used in the design, require 680 pF to 1000 pF NPO dielectric capacitors between the corresponding pin and analog ground. Each analog output is DC biased up to the



**Figure 7. Mono Input**

Vrefout voltage signal reference which is nominally 2.2 V. This requires that the output either be AC coupled to external circuitry (AC load must be greater than  $10\text{ k}\Omega$ ) or DC coupled to a buffer op-amp biased at the Vrefout voltage (see Figure 8 for the recommended headphone op-amp circuit).



**Figure 8. Headphones Driver**

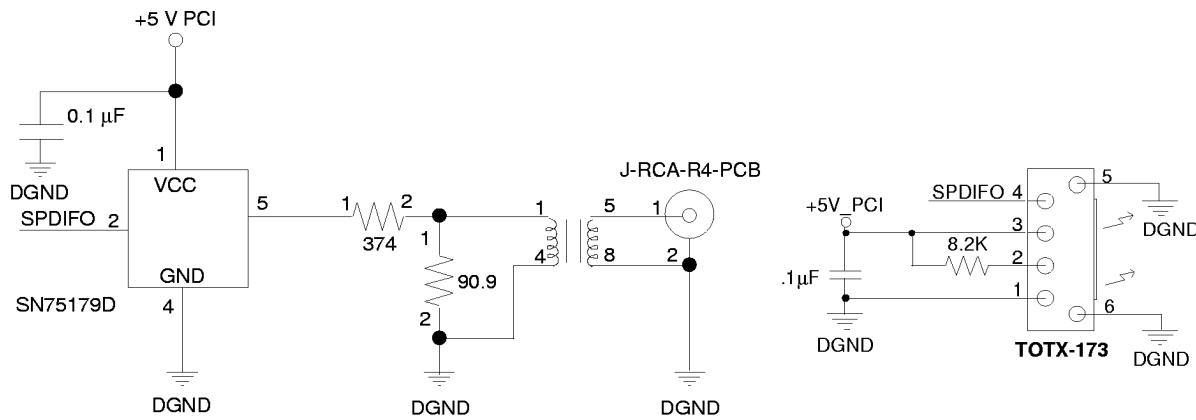
## 6.5 Miscellaneous Analog Signals

The AFLT1 and AFLT2 pins must have a 1000 pF NPO capacitor to analog ground. These capacitors, along with an internal resistor, provide a single-pole low-pass filter at the inputs to the ADCs. By placing these filters at the input to the ADCs, low-pass filters at each analog input pin are not necessary.

The REFFLT pin lowers the noise of the internal voltage reference. A 1 μF (must not be greater than 1 μF) and 0.1 μF capacitor to analog ground should be connected with a short, wide trace to this pin (see Figure 11 in the *Grounding and Layout* section for an example). No other connection should be made, as any coupling onto this pin will degrade the analog performance of the Codec. Likewise, digital signals should be kept away from REFFLT for similar reasons. The Vrefout pin is typically 2.2 V and provides a common mode signal for single-supply external circuits. Vrefout only supports light DC loads and should be buffered if AC loading is needed. For typical use, a 0.1 μF in parallel with a 1 μF capacitor should be connected to Vrefout.

## 6.6 Consumer IEC-958 Digital Interface (S/PDIF)

The CS4299 supports the industry standard IEC-958 consumer digital interface. Sometimes this interface is referred to as S/PDIF, which refers to an older version of this standard. This output provides an interface, external to the PC, for storing digital audio data or playing digital audio data to digital speakers. Figure 9 illustrates the circuit necessary for implementation of the IEC-958 optical or consumer interface.



**Figure 9. IEC-958 Interface Examples**

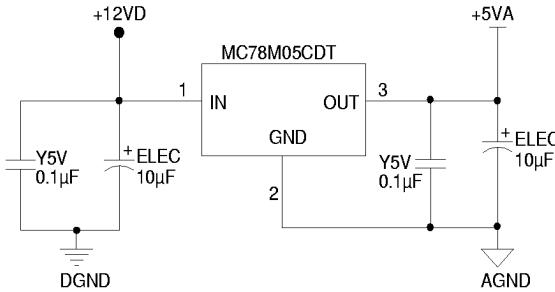
## 6.7 Power Supplies

The power supplies providing analog power should be as clean as possible to minimize coupling into the analog section which could degrade analog performance. The pins AVdd1 and AVdd2 supply power to all the analog circuitry on the CS4299. This 5 Volt analog supply should be generated from a voltage regulator (7805 type) connected to a +12 Volt supply. This helps isolate the analog circuitry from noise typically found on +5 V digital supplies which power many digital circuits in a PC environment. A typical voltage regulator circuit for analog power using an MC78M05CDT is shown in Figure 10. The digital power pins DVdd1 and DVdd2 should be connected to the same digital supply as the Controller's AC-link interface. Since the digital interface on the Codec may operate at either 3.3 V or 5 V, proper connection of these pins will depend on the digital power supply of the Controller.

## 7.0 GROUNDING AND LAYOUT

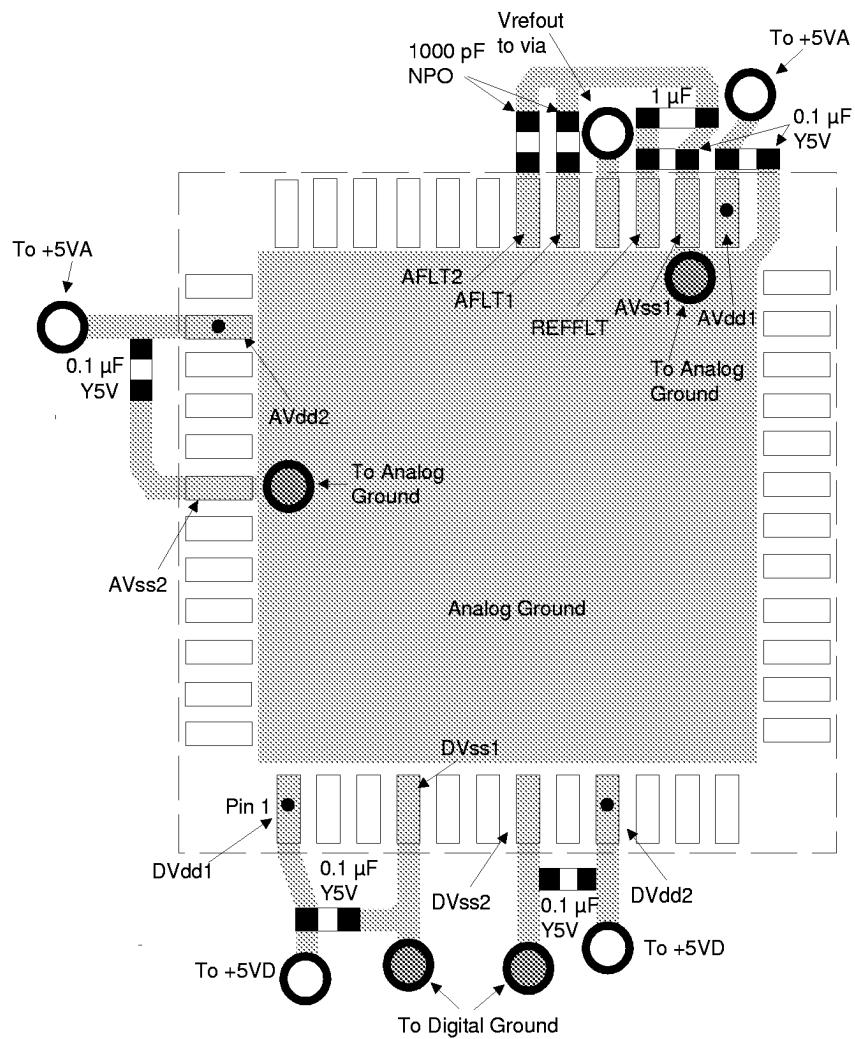
Figure 11 is the suggested layout for the Codec. The decoupling capacitors should be located physically as close to the pins as possible. Also note the routing of the REFFLT decoupling capacitors and the isolation of that ground strip.

It is strongly recommended that the device be located on a locally separate analog ground plane. This analog ground plane keeps noise from digital ground return currents from modulating the Codec's ground potential and degrading performance. The digital ground pins should be connected to the digital ground plane and kept separate from the analog ground connections of the Codec and any other external analog circuitry.



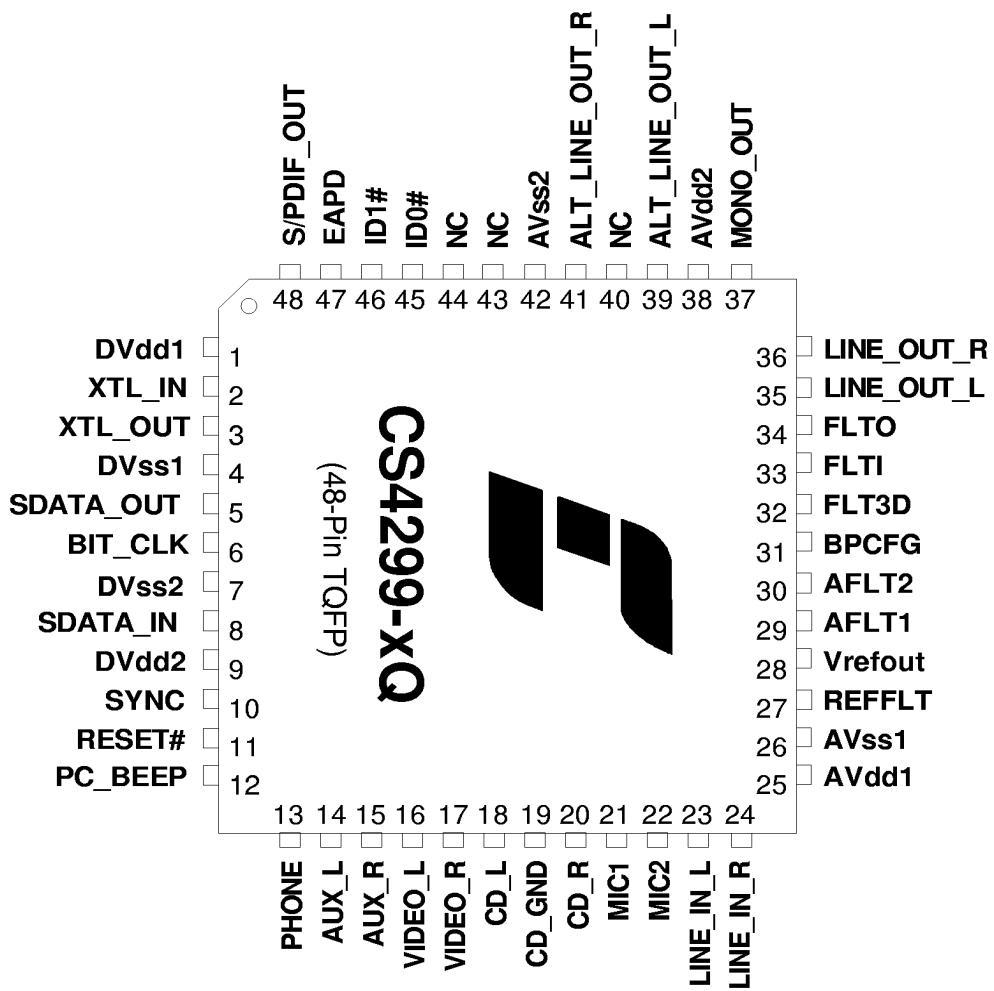
**Figure 10. Voltage Regulator**

The common connection point between the two ground planes (required to maintain a common ground voltage potential) should be located near the Codec just under the digital ground connections (vias). The AC-link digital interface connection traces should be routed such that the digital ground plane lies underneath these signals (on the internal ground layer) from the AC '97 Controller continuously to the Codec.



**Figure 11. Suggested Layout for the CS4299**

## 8.0 PIN DESCRIPTIONS



### Digital I/O Pins

#### **RESET# - AC '97 Chip Reset, Input**

This active low signal is the asynchronous cold reset input to the CS4299. The CS4299 must be reset before it can enter normal operating mode.

#### **SYNC - AC-Link Serial Port Sync pulse, Input**

This signal is the serial port timing signal for the AC-link. Its period is the reciprocal of the maximum sample rate, 48 kHz, and is generated by the AC '97 Controller synchronous to BIT\_CLK. SYNC is also an asynchronous input when the Codec is in a warm reset state. A series terminating resistor of  $47\ \Omega$  should be connected on this signal close to the Controller.

### **BIT\_CLK - AC-Link Serial Port Master Clock, Input/Output**

This input/output signal controls the master clock timing for the AC-link. When the Codec is in primary mode, this signal is a 12.288 MHz output clock signal divided down by two from the XTL\_IN input clock pin. When the Codec is in secondary mode, this signal is an input which controls the AC-link serial interface and generates all internal clocking. A series terminating resistor of  $47\ \Omega$  should be connected on this signal close to the primary Codec driving BIT\_CLK.

### **SDATA\_OUT - AC-Link Serial Data Input Stream to AC '97, Input**

This input signal transmits the control information and digital audio output streams to be sent to the DACs. The data is clocked into the Codec on the falling edge of BIT\_CLK. A series terminating resistor of  $47\ \Omega$  should be connected on this signal close to the Controller.

### **SDATA\_IN - AC-Link Serial Data Output Stream from AC '97, Output**

This output signal transmits the status information and digital audio input streams from the ADCs. The data is clocked out of the Codec on the rising edge of BIT\_CLK. A series terminating resistor of  $47\ \Omega$  should be connected on this signal as close to the Codec as possible.

### **XTL\_IN - Crystal Input**

When in primary mode, this pin requires either a crystal, with the other pin attached to XTL\_OUT, or an external CMOS clock. The crystal frequency must be 24.576 MHz and designed for fundamental mode, parallel resonance operation. When configured as a secondary Codec, all timing is derived from the BIT\_CLK input signal; this pin should be left floating.

### **XTL\_OUT - Crystal Output**

Used for a crystal placed between this pin and XLT\_IN. If an external clock is used on XTL\_IN, this pin must be left floating with no traces or components connected to it. When configured as a secondary Codec, this pin should be left floating.

### **ID1#, ID0# - Codec ID, Inputs**

These pins select the Codec ID and mode of operation for the Codec. Their value is sampled and latched on the rising edge of RESET#. These inputs use the digital supply bus for their value and contain internal pull-up resistors to the digital supply bus rail. The pins utilize inverted logic, so a value of '1:1' sets the Codec to primary mode while any other combination sets the Codec to secondary mode. In primary mode, the Codec is always clocked from an external crystal or an external oscillator connected to the XTL\_IN and/or XTL\_OUT pins with BIT\_CLK as an output. When either or both ID's are tied to analog ground, the Codec is in secondary mode and BIT\_CLK is always an input.

### **S/PDIF - IEC-958 Consumer Digital Output, Output**

This output provides a digital interface to devices external to the PC. With the appropriate buffer, the output can drive the IEC-958 consumer interface or directly drive an optical transmitter.

**BPCFG - PC Beep enable, Input**

This input controls the PC\_BEEP input when the device is held in reset. When unconnected or pulled high through an external pull-up resistor, the PC\_BEEP input is enabled. Grounding this input disables the PC\_BEEP when reset is active. The pin is pulled high through an internal 100 kΩ resistor.

**EAPD - External Amplifier Power Down, Output**

This signal is designated as a power down control for audio amplifiers external to the Codec. The output is determined by the EAPD bit and is low by default.

**Analog I/O Pins****PC\_BEEP - Analog Mono Source, Input**

This signal is generally used as an internal PC BEEP connection to the audio subsystem. This source is NOT input to the Stereo-to-Mono mixer. The maximum allowable input is 1 V<sub>RMS</sub> (sinusoidal). This input is internally biased at the Vrefout voltage reference and requires AC coupling to external circuitry. If this input is not used, it should be connected to the Vrefout pin or AC coupled to Analog ground.

**PHONE - Analog Mono Source, Input**

This signal is generally used as a voice modem connection to the audio subsystem. This source is NOT input to the Stereo-to-Mono mixer. The maximum allowable input is 1 V<sub>RMS</sub> (sinusoidal). This input is internally biased at the Vrefout voltage reference and requires AC coupling to external circuitry. If this input is not used, it should be connected to the Vrefout pin or AC coupled to analog ground.

**MIC1 - Analog Mono Source, Input**

This analog input is a monophonic source to the analog output mixer. It is generally used as a desktop microphone connection to the audio subsystem. This input is MUX-selectable to the input mixer with the MIC2 input source. The maximum allowable input is 1 V<sub>RMS</sub> (sinusoidal). This input is internally biased at the Vrefout voltage reference and requires AC coupling to external circuitry. If this input is not used, it should be AC coupled to analog ground.

**MIC2 - Analog Mono Source, Input**

This analog input is a monophonic source to the analog output mixer. It is generally used as an alternate microphone connection to the audio subsystem. This input is MUX-selectable to the input mixer with the MIC1 input source. The maximum allowable input is 1 V<sub>RMS</sub> (sinusoidal). This input is internally biased at the Vrefout voltage reference and requires AC coupling to external circuitry. If this input is not used, it should be AC coupled to analog ground.

**LINE\_IN\_L and LINE\_IN\_R - Analog Line Source, Inputs**

These inputs form a stereo input pair to the Codec. The maximum allowable input is 1 V<sub>RMS</sub> (sinusoidal). These inputs are internally biased at the Vrefout voltage reference. AC coupling to external circuitry is required. If these inputs are not used, they should both be connected to the Vrefout pin or both AC coupled, with separate AC coupling caps, to analog ground.

### **CD\_L and CD\_R - Analog CD Source, Inputs**

These inputs form a stereo input pair. Generally used for the Redbook CD audio connection to the audio subsystem. The maximum allowable input is 1 V<sub>RMS</sub> (sinusoidal). These inputs are internally biased at the Vrefout voltage reference. AC coupling to external circuitry is required. If these inputs are not used, they should both be connected to the Vrefout pin or both AC coupled, with separate AC coupling caps, to analog ground.

### **CD\_GND - Analog CD Common Source, Input**

This analog input is used to remove common mode noise from Redbook CD audio signals. The impedance on the input signal path should be one half the impedance on the CD\_L and CD\_R input paths. This pin requires AC coupling to external circuitry. If this input is not used, it should be connected to the Vrefout pin or AC coupled to analog ground.

### **VIDEO\_L and VIDEO\_R - Analog Video Audio Source, Inputs**

These inputs form a stereo input pair. It is generally used for the audio signal output of a video device. The maximum allowable input is 1 V<sub>RMS</sub> (sinusoidal). These inputs are internally biased at the Vrefout voltage reference. AC coupling to external circuitry is required. If these inputs are not used, they should both be connected to the Vrefout pin or both AC coupled, with separate AC coupling caps, to analog ground.

### **AUX\_L and AUX\_R - Analog Auxiliary Source, Inputs**

These inputs form a stereo input pair. The maximum allowable input is 1 V<sub>RMS</sub> (sinusoidal). These inputs are internally biased at the Vrefout voltage reference. AC coupling to external circuitry is required. If these inputs are not used, they should both be connected to the Vrefout pin or both AC coupled, with separate AC coupling caps, to analog ground.

### **LINE\_OUT\_L and LINE\_OUT\_R - Analog Line Level Outputs**

These signals are analog outputs from the stereo output mixer. The full scale output voltage for output is nominally 1 V<sub>RMS</sub> and is internally biased at the Vrefout voltage reference. It is required to either AC couple these pins to external circuitry or DC couple them to a buffer op-amp biased at the Vrefout voltage. These pins need a 680 pF to 1000 pF NPO capacitor attached to analog ground.

### **ALT\_LINE\_OUT\_L and ALT\_LINE\_OUT\_R - Analog Alternate Line Level Outputs**

These signals are analog outputs from the stereo output mixer. The full scale output voltage for each output is nominally 1 V<sub>RMS</sub> and is internally biased at the Vrefout voltage reference. It is required to either AC couple these pins to external circuitry or DC couple them to a buffer op-amp biased at the Vrefout voltage. These pins need a 680 pF to 1000 pF NPO capacitor attached to analog ground.

## MONO\_OUT, Analog Mono Line Level Output

This signal is an analog output from the Mono output mixer or MIC1/2. When the Mono output mixer is selected, the left and right channels are mixed from the output of the stereo input mixer. When the MIC mode is selected, MIC1 or MIC2 is routed to the Mono\_Out. The full scale output is nominally 1 V<sub>RMS</sub> and is internally biased at the Vrefout voltage reference. AC coupling to external circuitry is required. This pin needs a 680 pF to 1000 pF NPO capacitor attached to analog ground.

### Filter and Reference Pins

#### **REFFLT - Internal Reference Voltage, Input**

This is the voltage reference used internal to the part. A 0.1  $\mu$ F and a 1  $\mu$ F (must not be larger than 1  $\mu$ F) capacitor with short, wide traces must be connected to this pin. No other connections should be made to this pin.

#### **Vrefout - Voltage Reference, Output**

All analog inputs and outputs are centered around Vrefout which is nominally 2.2 Volts. This pin may be used to level shift external circuitry, however any external loading should be buffered.

#### **AFLT1 - Left Channel Antialiasing Filter Input**

This pin needs a 1000 pF NPO capacitor attached to analog ground.

#### **AFLT2 - Right Channel Antialiasing Filter Input**

This pin needs a 1000 pF NPO capacitor attached to analog ground.

#### **FLTI - FLTO - 3D Filter**

A 1000 pF capacitor must be attached between FLTI and FLTO if the 3D function is used.

#### **FILT3D - 3D Filter**

A 0.01  $\mu$ F capacitor must be attached from this pin to AGND if the 3D function is used.

### Power Supplies

#### **DVdd1, DVdd2 - Digital Supply Voltage**

Digital supply voltage for the AC-link section of the Codec. These pins can be tied to +5 V digital or to +3.3 V digital. The Codec and Controller's AC-link should share a common digital supply

#### **DVss1, DVss2 - Digital Ground**

Digital ground connection for the AC-link section of the Codec. These pins should be isolated from analog ground currents.

**AVdd1, AVdd2 - Analog Supply Voltage**

Analog supply voltage for the analog and mixed signal sections of the Codec. These pins must be tied to +5 Volt power supply. It is strongly recommended that +5 Volts be generated from a voltage regulator to ensure proper supply currents and noise immunity from the rest of the system.

**AVss1, AVss2 - Analog Ground**

Ground connection for the analog, mixed signal, and substrate sections of the Codec. These pins should be isolated from digital ground currents.

## 9.0 PARAMETER AND TERM DEFINITIONS

### AC '97 Specification

Refers to the *Audio Codec '97 Component Specification Ver 2.1* published by Intel® Corporation [1].

### AC '97 Controller or Controller

Refers to the control chip which interfaces to the Codec's AC-link. This has been also called *DC '97* for Digital Controller '97 [1].

### AC '97 Registers or Codec registers

Refers to the 64-field register map defined in the AC '97 Specification.

### ADC

Refers to a single Analog-to-Digital converter in the Codec. "ADCs" refers to the stereo pair of Analog-to-Digital converters.

### DAC

A single Digital-to-Analog converter in the Codec "DACs" refers to the stereo pair of Digital-to-Analog converters.

### SRC

Sample Rate converter. Converts data derived at one sample rate to a differing sample rate.

### Codec

Refers to the chip containing the ADCs, DACs, and analog mixer. In this data sheet, the Codec is the CS4299.

### FFT

Fast Fourier Transform.

### Resolution

The number of bits in the output words to the DACs, and in the input words to the ADCs.

### Differential Nonlinearity

The worst case deviation from the ideal code width. Units in LSB.

### dB FS A

dB FS is defined as dB relative to full-scale. The "A" indicates an A weighting filter was used.

## Frequency Response (FR)

FR is the deviation in signal level versus frequency. The 0 dB reference point is 1 kHz. The amplitude corner,  $A_c$ , lists the maximum deviation in amplitude above and below the 1 kHz reference point. The listed minimum and maximum frequencies are guaranteed to be within the  $A_c$  from minimum frequency to maximum frequency inclusive.

## Dynamic Range (DR)

DR is the ratio of the RMS full-scale signal level divided by the RMS sum of the noise floor, in the presence of a signal, available at any instant in time (no change in gain settings between measurements). Measured over a 20 Hz to 20 kHz bandwidth with units in dB FS A.

## Total Harmonic Distortion plus Noise (THD+N)

THD+N is the ratio of the RMS sum of all non-fundamental frequency components, divided by the RMS full-scale signal level. It is tested using a -3 dB FS input signal and is measured over a 20 Hz to 20 kHz bandwidth with units in dB FS.

## Signal to Noise Ratio (SNR)

SNR, similar to DR, is the ratio of an arbitrary sinusoidal input signal to the RMS sum of the noise floor, in the presence of a signal. It is measured over a 20 Hz to 20 kHz bandwidth with units in dB.

## S/PDIF

Sony/Phillips Digital Interface. This interface was established as a means of digitally interconnecting consumer audio equipment. The documentation for S/PDIF has been superseded by the IEC-958 consumer digital interface document.

## Interchannel Isolation

The amount of 1 kHz signal present on the output of the grounded AC-coupled line input channel with 1 kHz, 0 dB, signal present on the other line input channel. Units in dB.

## Interchannel Gain Mismatch

For the ADCs, the difference in input voltage to get an equal code on both channels. For the DACs, the difference in output voltages for each channel when both channels are fed the same code. Units in dB.

## PATHS

A-D: Analog in, through the ADC, onto the serial link.

D-A: Serial interface inputs through the DAC to the analog output.

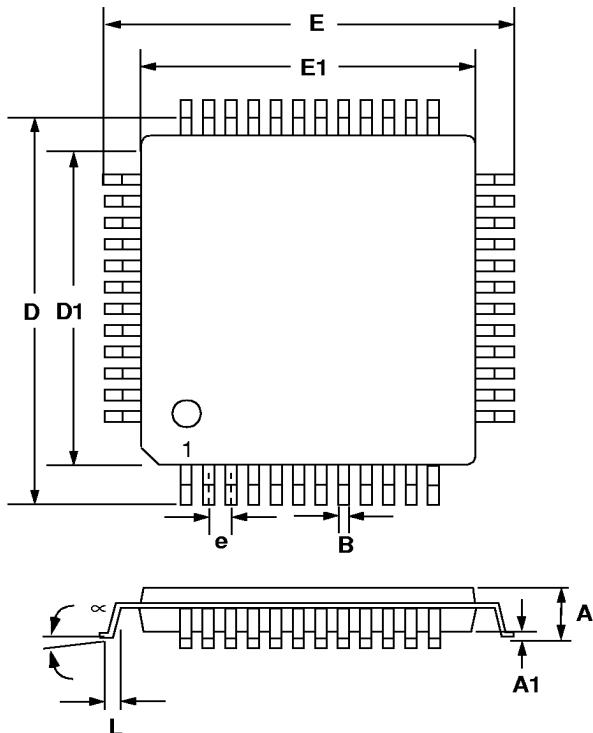
A-A: Analog in to Analog out (analog mixer).

## 10.0 REFERENCES

- 1) Intel, Audio Codec '97 Component Specification, Revision 2.1, May 22, 1998.  
<http://developer.intel.com/pc-suppl/platform/ac97/>

## 11.0 PACKAGE DIMENSIONS

**48L TQFP PACKAGE DRAWING**



DIM	INCHES		MILLIMETERS	
	MIN	MAX	MIN	MAX
A	----	0.063	----	1.600
A1	0.002	0.006	0.050	0.150
B	0.007	0.011	0.170	0.270
D	0.343	0.366	8.700	9.300
D1	0.272	0.280	6.900	7.100
E	0.343	0.366	8.700	9.300
E1	0.272	0.280	6.900	7.100
e*	0.016	0.024	0.400	0.600
L	0.018	0.030	0.450	0.750
$\infty$	0.000°	7.000°	0.000°	7.000°

\* Nominal pin pitch is 0.50 mm

Controlling dimension is mm.

JEDEC Designation: MS026