



.4-Bit, 96kHz Sampling .-Sigma Stereo Audio -ANALOG CONVERTER

_TA-SIGMA DAC _: 16kHz - 96kHz D:

√6dB

Range: 0.62 x V_{cc} (Vp-p) PLING DIGITAL FILTER: Attenuation: –82dB Ripple: ±0.002dB

ال. Off

MULTI FUNCTIONS:

 Digital De-emphasis
 L/R Independent Digital Attenuation
 Soft Mute
 Zero Detect Mute
 Zero Flag
 Chip Select
 Reversible Output Phase

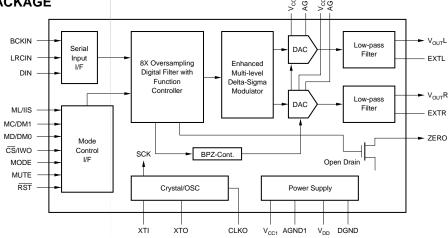
- +5V SINGLE SUPPLY OPERATION
- SMALL 28-LEAD SSOP PACKAGE

DESCRIPTION

The PCM1716 is designed for Mid to High grade Digital Audio applications which achieve 96kHz sampling rates with 24-bit audio data. PCM1716 uses a newly developed, enhanced multi-level delta-sigma modulator architecture that improves audio dynamic performance and reduces jitter sensitivity in actual applications.

The internal digital filter operates at 8x over sampling at a 96kHz sampling rate, with two kinds of roll-off performances that can be selected: sharp roll-off, or slow roll-off, as required for specific applications.

PCM1716 is suitable for Mid to High grade audio applications such as CD, DVD-Audio, and Music Instruments, since the device has superior audio dynamic performance, 24-bit resolution and 96kHz sampling.



International Airport Industrial Park • Mailing Address: PO Box 11400, Tucson, AZ 85734 • Street Address: 6730 S. Tucson Blvd., Tucson, AZ 85706 • Tel: (520) 746-1111 • Twx: 910-952-1111
Internet: http://www.burr-brown.com/ • FAXLine: (800) 548-6133 (US/Canada Only) • Cable: BBRCORP • Telex: 066-6491 • FAX: (520) 889-1510 • Immediate Product Info: (800) 548-6132

SPECIFICATIONS

All specifications at $+25^{\circ}\text{C}$, $+\text{V}_{\text{CC}}$ = $+\text{V}_{\text{DD}}$ = +5V, f_{S} = 44.1kHz, and 24-bit input data, SYSCLK = $384f_{\text{S}}$, unless otherwise noted.

		1	PCM1716			
PARAMETER	CONDITIONS	MIN	TYP	MAX	UNITS	
RESOLUTION			24		Bits	
DATA FORMAT Audio Data Interface Format Data Bit Length Audio Data Format Sampling Frequency (f _S)		16	Standard/l ² S 16/20/24 Selectable MSB First, 2's Com	p 96	kHz	
System Clock Frequency ⁽¹⁾ DIGITAL INPUT/OUTPUT LOGIC LEVEL			256/384/512/768f _S		-	
Input Logic Level V _{IH} V _{IL}		2.0		0.8	V	
Output Logic Level (CLKO) V _{OH} V _{OL}	$I_{OH} = 2mA$ $I_{OL} = 4mA$	4.5		0.5	V V	
CLKO PERFORMANCE ⁽²⁾ Output Rise Time Output Fall Time Output Duty Cycle	20 ~ 80% V _{DD} , 10pF 80 ~ 20% V _{DD} , 10pF 10pF Load		5.5 4 37		ns ns %	
DYNAMIC PERFORMANCE(3) (24-Bit Data)						
THD+N $V_{O} = 0 dB$ $V_{O} = -60 dB$ Dynamic Range Signal-to-Noise Ratio ⁽⁴⁾	$\begin{aligned} f_S &= 44.1 \text{kHz} \\ f_S &= 96 \text{kHz} \\ f_S &= 44.1 \text{kHz} \end{aligned}$ $\begin{aligned} f_S &= 44.1 \text{kHz} \\ f_S &= 44.1 \text{kHz} \end{aligned}$ $\begin{aligned} f_S &= 44.1 \text{kHz} \end{aligned}$	98 98	-97 -94 -42 106 103 106 103	-9 0	dB dB dB dB dB dB	
Channel Separation	$f_S = 44.1 \text{kHz}$ $f_S = 96 \text{kHz}$	96	102 101		dB dB	
DYNAMIC PERFORMANCE ⁽³⁾ (16-Bit Data) THD+N $V_0 = 0$ dB Dynamic Range	$f_S = 44.1 \text{kHz}$ $f_S = 96 \text{kHz}$ $f_S = 44.1 \text{kHz EIAJ A-weighted}$ $f_S = 96 \text{kHz A-weighted}$		-94 -92 98 97		dB dB dB dB	
DC ACCURACY Gain Error			±1.0	±3.0	% of FS	
Gain Mismatch: Channel-to-Channel Bipolar Zero Error	$V_O = 0.5V_{CC}$ at Bipolar Zero		±1.0 ±30	±3.0 ±60	% of FS mV	
ANALOG OUTPUT Output Voltage Center Voltage Load Impedance	Full Scale (0dB) AC Load	5	0.62 V _{CC} 0.5 V _{CC}		Vp-p V kΩ	
DIGITAL FILTER PERFORMANCE Filter Characteristics 1						
(Sharp Roll-Off) Passband Stopband Passband Ripple Stopband Attenuation	±0.002dB -3dB Stop Band = 0.546f _s	0.546f _S -75		0.454f _S 0.490f _S ±0.002	dB dB	
Filter Characteristics 2 (Slow Roll-Off) Passband	Stop Band = $0.567f_S$ $\pm 0.002dB$	-82		0.274f _S	dB	
Stopband Passband Ripple Stopband Attenuation Delay Time	−3dB Stopband = 0.732f _S	0.732f _S -82	30/f _S	0.454f _S ±0.002	dB dB sec	
De-emphasis Error INTERNAL ANALOG FILTER				±0.1	dB	
-3dB Bandwidth Passband Response	f = 20kHz		100 -0.16		kHz dB	
POWER SUPPLY REQUIREMENTS Voltage Range Supply Current: I _{CC} +I _{DD} Power Dissipation	$V_{DD,} V_{CC}$ $f_S = 44.1 \text{kHz}$ $f_S = 96 \text{kHz}$ $f_S = 44.1 \text{kHz}$	4.5	5 32 45 160	5.5 45 225	VDC mA mA mW	
TEMPERATURE RANGE	$f_S = 96kHz$		225		mW	
Operation Storage		-25 -55		+85 +100	°C °C	

NOTES: (1) Refer section of system clock. (2) External buffer is recommended. (3) Dynamic performance specs are tested with 20kHz low pass filter and THD+N specs are tested with 30kHz LPF, 400Hz HPF, Average Mode. (4) SNR is tested at internally infinity zero detection off.



PIN CONFIGURATION

LRCIN 1 DIN 2 BCKIN 3 CLKO 4 XTI 5 6 хто DGND 7 8 $\rm V_{\rm DD}$ V_{CC}2R AGND2R EXTR N Vou AGN

PACKA

PRODU PCM1

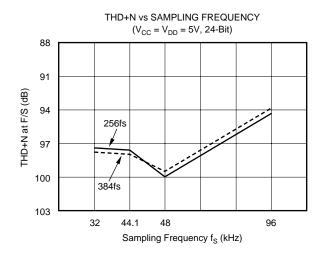
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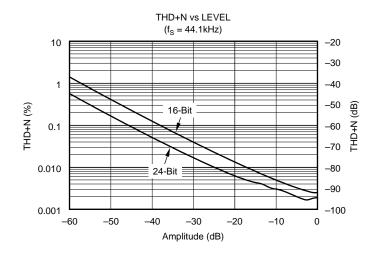
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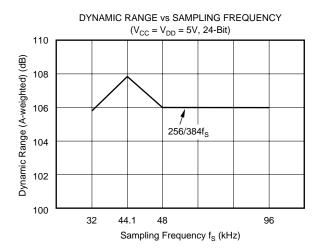


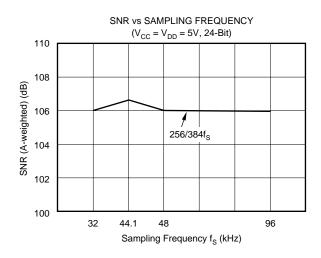
TYPICAL PERFORMANCE CURVES

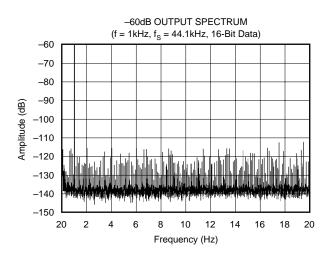
All specifications at +25°C, + V_{CC} = + V_{DD} = +5V, f_S = 44.1kHz, and 24-bit input data, SYSCLK = 384f_S, unless otherwise noted.

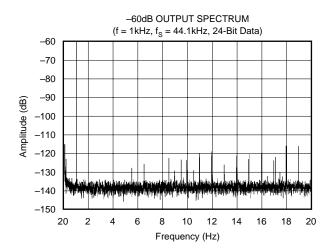




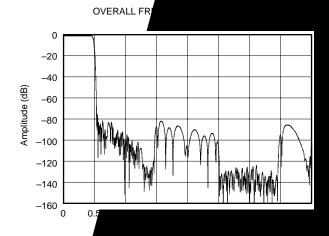


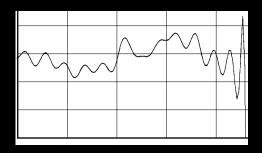


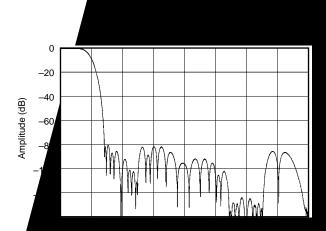




TYPICAL PERF







SYSTEM CLOCK

The system clock for PCM1716 must be either $256f_S$, $384f_S$, $512f_S$ or $768f_S$, where f_S is the audio sampling frequency (typically 32kHz, 44.1kHz, 48kHz, or 96kHz). But $768f_S$ at 96kHz is not accepted.

The system clock can be either a crystal oscillator placed between XTI (pin 5) and XTO (pin 6), or an external clock input to XTI. If an external system clock is used, XTO is open (floating). Figure 1 illustrates the typical system clock connections.

PCM1716 has a system clock detection circuit which automatically senses if the system clock is operating at $256f_S \sim 768f_S$. The system clock should be synchronized with LRCIN (pin 1) clock. LRCIN (left-right clock) operates at the sampling frequency f_S . In the event these clocks are not synchronized, PCM1716 can compensate for the phase difference internally. If the phase difference between left-right and system clocks is greater than 6-bit clocks (BCKIN), the synchronization is performed internally. While the synchronization is processing, the analog output is forced to a DC level at bipolar zero. The synchronization typically occurs in less than 1 cycle of LRCIN.

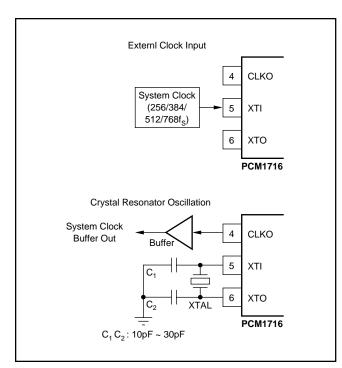


FIGURE 1. System Clock Connection.

Typical input system clock frequencies to the PCM1716 are shown in Table I, also, external input clock timing requirements are shown in Figure 2.

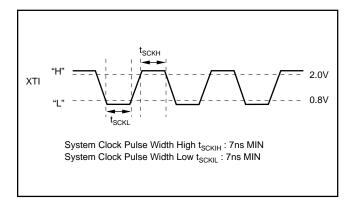


FIGURE 2. XTI Clock Timing.

DATA INTERFACE FORMATS

Digital audio data is interfaced to PCM1716 on pins 1, 2, and 3, LRCIN (left-right clock), DIN (data input) and BCKIN (bit clock). PCM1716 can accept both standard, I²S, and left justified data formats.

Figure 3 illustrates acceptable input data formats. Figure 4 shows required timing specification for digital audio data.

Reset

PCM1716 has both internal power-on reset circuit and the \overline{RST} pin (pin 22) which accepts an external forced reset by \overline{RST} = LOW. For internal power on reset, initialize (reset) is done automatically at power on V_{DD} >2.2V (typ). During internal reset = LOW, the output of the DAC is invalid and the analog outputs are forced to $V_{CC}/2$. Figure 5 illustrates the timing of the internal power on reset.

PCM1716 accepts an external forced reset when $\overline{RST}=L$. When $\overline{RST}=L$, the output of the DAC is invalid and the analog outputs are forced to $V_{CC}/2$ after internal initialization (1024 system clocks count after $\overline{RST}=H$.) Figure 6 illustrates the timing of the \overline{RST} pin.

Zero Out (pin 21)

If the input data is continuously zero for 65536 cycles of BCK, an internal FET is switched to "ON". The drain of the internal FET is the zero-pin, it will enable "wired-or" with external circuit. This zero detect function is available in both software mode and hardware mode.

	SYSTEM CLOCK FREQUENCY - MHz					
SAMPLING RATE FREQUENCY (f_{S}) - LRCIN	256f _S	384f _S	512f _S	768f _S		
32kHz	8.1920	12.2880	16.3840	24.5760		
44.1kHz	11.2896	16.9340	22.5792	33.8688 ⁽¹⁾		
48kHz	12.2880	18.4320	24.5760	36.8640 ⁽¹⁾		
96kHz	24.5760	36.8640 ⁽¹⁾	49.1520 ⁽¹⁾	_		

NOTE: (1) The Internal Crystal oscillator frequency cannot be larger than 24.576MHz.

TABLE I. Typical System Clock Frequency.



LRCIN (pin 1)	
BCKIN (pin 3)	
(1) 16-Bit Right	Justifie
DIN (pin 2)	14 1
(2) 20-Bit Right	Just
DIN (pin 2)	18
(3) 24-Bit Right	
DIN (pin 2)	
(4) 24-Bit Lef	
DIN (pin	
LRCIN (pir	
EKCIN (pir	
BCKIN (p	
(5)	
DIN	

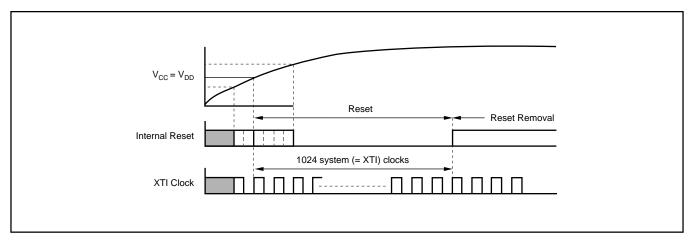


FIGURE 5. Internal Power-On Reset Timing.

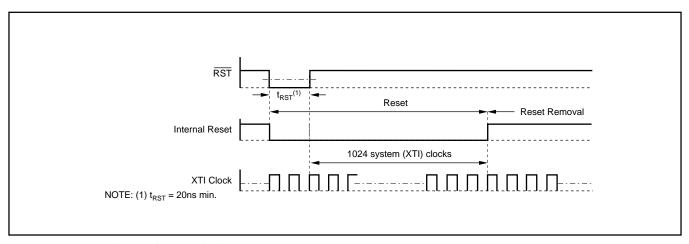


FIGURE 6. External Forced Reset Timing.

FUNCTIONAL DESCRIPTION

PCM1716 has several built-in functions including digital attenuation, digital de-emphasis, input data format selection, and others. These functions are software controlled. PCM1716 can be operated in two different modes, software mode or hardware mode. Software mode is a three-wire interface using pin 28 (ML), 27 (MC), and 26 (MD).

PCM1716 can also be operated in hardware mode, where static control signals are used on pin 28 (115, pin 27 (DM1), pin 26 (DM0) and pin 23 (IWO).

This basic operation mode as software or hardware can be selected by pin 24 (MODE) as shown in Table II.

MODE (pin 24) = H	Software Mode
MODE (pin 24) = L	Hardware Mode

TABLE II. Mode Control.

Table III indicates which functions are selectable within the users chosen mode. All of the functions shown are selectable within the software mode, but only de-emphasis control, soft mute and input data format may be selected when using PCM1716 in the hardware mode.

FUNCTION	SOFTWARE (Mode = H)	HARDWARE (Mode = L)
Input Data Format Selection	0	0
Input Data Bit Selection	0	0
Input LRCIN Polarity Selection	0	X
De-emphasis Control	0	0
Mute	0	0
Attenuation	0	X
Infinity Zero Mute Control	0	X
DAC Operation Control	0	X
Slow Roll-Off Selection	0	X
Output Phase Selection	0	Х
CLKO Output Selection	0	X

NOTE: O = Selectable, X: Not Selectable.

TABLE III. Mode Control, Selectable Functions.

HARDWARE MODE (MODE = L)

In hardware mode, the following function can be selected.

De-emphasis control

De-emphasis control can be selected by DM1 (pin 27) and DM0 (pin 26)

DM1 (Pin 27)	DM0
L	
L	1 1
Н	
Н	

TABLE IV. De-emphasis

Input Audio Data Fo

Input data format can (pin 23)

I ² S (Pin 28)	IWO (Pin
L	L
L	Н
Н	Ļ
Н	

TABLE V. Data

SOFT MUTE

Soft Mute fu



SOFT

PCM1 Table MD

REGISTER NAME	BIT NAME	DESCRIPTION
Register 0	AL (7:0) LDL A (1:0) res	DAC Attenuation Data for Lch Attenuation Data Load Control for Lch Register Address Reserved, should be "L"
Register 1	AR (7:0) LDR A (1:0) res	DAC Attenuation Data for Rch Attenuation Data Load Control for Rch Register Address Reserved, should be "L"
Register 2	MUT DEM OPE IW (1:0) res A (1:0) res	Left and Right DACs Soft Mute Control De-emphasis Control Left and Right DACs Operation Control Input Audio Data Bit and Format Select Reserved Register Address Reserved, should be "L"
Register 3	I ² S LRP ATC SRO REV CKO SF (1:0) IZD A (1:0) res	Audio Data Format Select Polarity of LRCIN Select Attenuator Control Slow Roll-Off Select Output Phase Select CLKO Output Select Sampling Rate Select Internal Zero Detection Circuit Control Register Address Reserved, should be "L"

TABLE VII. Register Functions

REGISTER 0 (A1 = 0, A0 = 0)

Register 0 is used to control left channel attenuation. Bits 0 - 7 (AL0 - AL7) are used to determine the attenuation level. The level of attenuation is given by:

ATT = 0.5 x (data-255) (dB)
FFh = -0dB
FEh = -0.5dB
:
:
01h = -127.5dB

$$00h = -\infty$$
 (= Mute)

ATTENUATION DATA LOAD CONTROL

Bit 8 (LDL) is used to control the loading of attenuation data in B0:B7. When LDL is set to 0, attenuation data will be loaded into AL0:AL7, but it will not affect the attenuation level until LDL is set to 1. LDR in Register 1 has the same function for right channel attenuation.

REGISTER 1 (A1 = 0, A0 = 1)

Register 1 is used to control right channel attenuation. As in Register 1, bits 0 - 7 (AR0 - AR7) control the level of attenuation.

REGISTER 2 (A1 = 1, A0 = 0)

B15	B14	B13	B12	B11	B10	B9	B8	B7	B6	B5	B4	B3	B2	B1	B0
res	res	res	res	res	A1	A0	res	res	res	res	IW1	IWO	OPE	DEM	MUTE

Register 2 is used to control soft mute, de-emphasis, operation enable, input resolution, and input audio data bit and format.

MUT (B0)	
MUT = L	Soft Mute OFF
MUT = H	Soft Mute ON

DEM (B1)	
DEM = L	De-emphasis OFF
DEM = H	De-emphasis ON

OPE (B2)	
OPE = L	Normal Operation
OPE = H	DAC Operation OFF

when OPE (B2) is "HIGH", the output of the DAC will be forced to bipolar zero, irrespective of any input data.

IWO (B3), IW1 (B4) and I^2S (B0) of Register 3

These resisters, IWO, IW1, I²S determine the input data word and input data format as shown below.

IW1	IW0	I ² S	Audio Interface				
0	0	0	16-Bit Standard (Right-Justified)				
0	1	0	20-Bit Standard (Right-Justified)				
1	0	0 0 24-Bit Standard (Right-Just					
1	1	0	24-Bit Left-Justified (MSB First)				
0	0	1	16-Bit I ² S				
0	1	1	24-Bit I ² S				
1	0	1	Reserved				
1	1	1	Reserved				

REGISTER 3 (A1 = 1, A0 = 1)

							B8								
res	res	res	res	res	A1	A0	IZD	SF1	SF0	СКО	REV	SRO	ATC	LRP	I ² S

REGISTER 3 (A1 = 1, A0 = 1)

Register 3 is used to control input data format and polarity, attenuation channel control, system clock frequency, sampling frequency, infinite zero detection, output phase, CLKO output, and slow roll-off.

Bit 8 is used to control the infinite zero detection function (IZD).

When IZD is "LOW", the zero detect circuit is off. Under this condition, no automatic muting will occur if the input is continuously zero. When IZD is "HIGH", the zero detect feature is enabled. If the input data is continuously zero for 65, 536 cycles of BCKIN, the output will be immediately

forced to a bipolar zero state feature is used to avoid noi input is DC. When the ou there may be an audible of detect feature to be disable external muting circuit.

IZD (B8)	
B8 = L	
B8 = H	
B0 = 11	Ц

Bits 6 (SF0) and 7 (frequency for De-er

SF1	SF0
0	0
0	1
1	0
1	1

CKO (B5) is or selected as B (1/2).

СК
Ck

REF (B4

THEORY OF OPERATION

The delta-sigma section of PCM1716 is based on a 8-level amplitude quantizer and a 4th-order noise shaper. This section converts the oversampled input data to 8-level delta-sigma format.

This newly developed, "Enhanced Multi-level Delta-Sigma" architecture achieves high-grade audio dynamic performance and sound quality.

A block diagram of the 8-level delta-sigma modulator is shown in Figure 9. This 8-level delta-sigma modulator has

the advantage of stability and clock jitter sensitivity over the typical one-bit (2 level) delta-sigma modulator.

The combined oversampling rate of the delta-sigma modulator and the internal 8-times interpolation filter is $64f_S$ for all system clock ratios ($256/384/512/768f_S$).

The theoretical quantization noise performance of the 8-level delta-sigma modulator is shown in Figure 10. This enhanced multi-level delta-sigma architecture also has advantages for input clock jitter sensitivity due to the multi-level quantizer, simulated jitter sensitivity is shown in Figure 11.

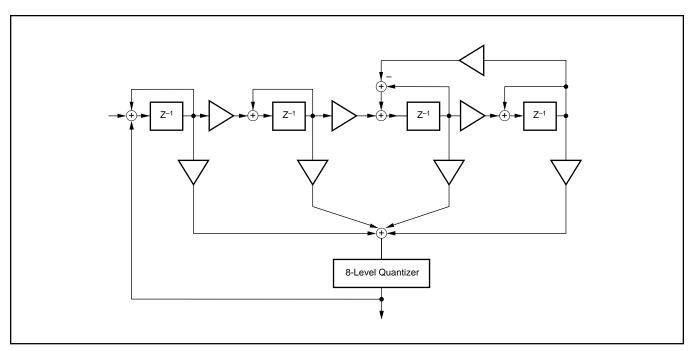


FIGURE 9. 8-Level Delta-Sigma Modulator.

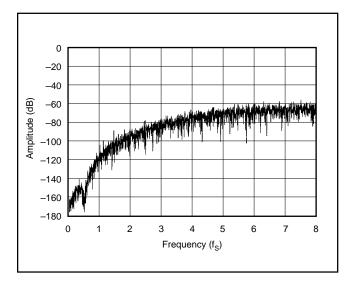


FIGURE 10. Quantization Noise Spectrum.

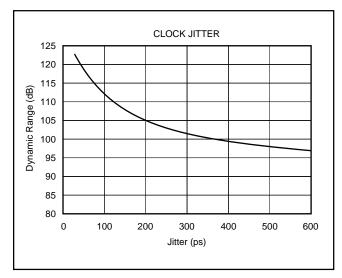


FIGURE 11. Jitter Sensitivity.

APPLICATION CONSIDERATION

DELAY TIME

There is a finite delay time converters, this is comm delta-sigma D/A convertorder number of the FIR rate. The following expCM1716:

For $f_S = 44$

Applications using CD audio, DVD generally are no sional application important for t

OUTPUT F

For testing PCM1716 the measure to use such SNR and specifications.

The part of the has

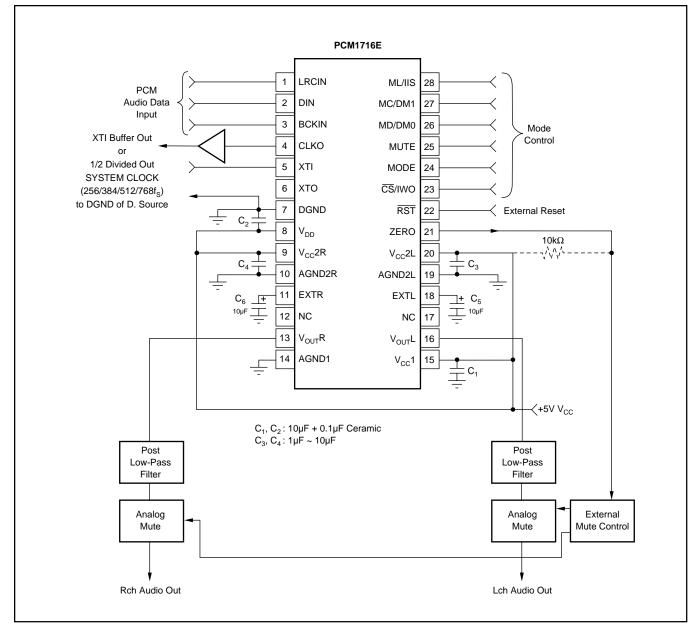


FIGURE 15. Typical Circuit Connection Diagram.





3-Oct-2003

PACKAGING INFORMATION

ORDERABLE DEVICE	STATUS(1)	PACKAGE TYPE	PACKAGE DRAWING	PINS	PACKAGE QTY
PCM1716E	ACTIVE	SSOP	DB	28	47
PCM1716E/2K	ACTIVE	SSOP	DB	28	2000

(1) The marketing status values are defined as follows:

ACTIVE: Product device recommended for new designs.

LIFEBUY: TI has announced that the device will be discontinued, and a lifetime-buy period is in effect.

NRND: Not recommended for new designs. Device is in production to support existing customers, but TI does not recommend using this part in a new design.

PREVIEW: Device has been announced but is not in production. Samples may or may not be available.

OBSOLETE: TI has discontinued the production of the device.

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