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ANALOG DEVICES

SigmaDSP Stereo, Low Power, 96 kHz, 24-Bit Audio Codec with Integrated PLL

ADAU1461

FEATURES

SigmaDSP 28-/56-bit, 50 MIPS digital audio processor
Fully programmable with SigmaStudio graphical tool
24-bit stereo audio ADC and DAC: >98 dB SNR
Sampling rates from 8 kHz to 96 kHz
Low power: 17 mW record, 18 mW playback, 48 kHz
6 analog input pins, configurable for single-ended or
    differential inputs
Flexible analog input/output mixers
Stereo digital microphone input
Analog outputs: 2 differential stereo, 2 single-ended stereo,
    1 mono headphone output driver
PLL supporting input clocks from 8 MHz to 27 MHz
Analog automatic level control (ALC)
Microphone bias reference voltage
Analog and digital I/O: 3.3 V
I²C and SPI control interfaces
Digital audio serial data I/O: stereo and time-division
    multiplexing (TDM) modes
Software-controllable clickless mute
GPIO pins for digital controls and outputs
32-lead, 5 mm × 5 mm LFCSP
−40°C to +105°C operating temperature range
Qualified for automotive applications

APPLICATIONS

Automotive head units
Automotive amplifiers
Navigation systems
Rear-seat entertainment systems

GENERAL DESCRIPTION

The ADAU1461 is a low power, stereo audio codec with integrated digital audio processing that supports stereo 48 kHz record and playback at 35 mW from a 3.3 V analog supply. The stereo audio ADCs and DACs support sample rates from 8 kHz to 96 kHz as well as a digital volume control.

The SigmaDSP® core features 28-bit processing (56-bit double precision). The processor allows system designers to compensate for the real-world limitations of microphones, speakers, amplifiers, and listening environments, resulting in a dramatic improvement in the perceived audio quality through equalization, multiband compression, limiting, and third-party branded algorithms.

The SigmaStudio™ graphical development tool is used to program the ADAU1461. This software includes audio processing blocks such as filters, dynamics processors, mixers, and low level DSP functions for fast development of custom signal flows.

The record path includes an integrated microphone bias circuit and six inputs. The inputs can be mixed and muxed before the ADC, or they can be configured to bypass the ADC. The ADAU1461 includes a stereo digital microphone input.

The ADAU1461 includes five high power output drivers (two differential and three single-ended), supporting stereo headphones, an earpiece, or other output transducer. AC-coupled or capless configurations are supported. Individual fine level controls are supported on all analog outputs. The output mixer stage allows for flexible routing of audio.
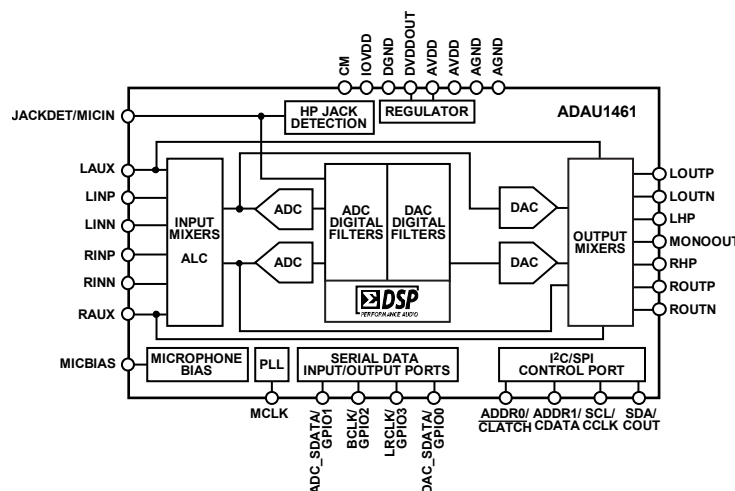
FUNCTIONAL BLOCK DIAGRAM

Figure 1.

ADAU1461* PRODUCT PAGE QUICK LINKS

Last Content Update: 02/23/2017

COMPARABLE PARTS 🖵

View a parametric search of comparable parts.

DOCUMENTATION 🖵

Application Notes

- AN-1006: Using the EVAL-ADUSB2EBZ
- AN-951: Using Hardware Controls with SigmaDSP GPIO Pins

Data Sheet

- ADAU1461: SigmaDSP Stereo, Low Power, 96 kHz, 24-Bit Audio Codec with Integrated PLL Data Sheet

SOFTWARE AND SYSTEMS REQUIREMENTS 🖵

- ADAU1361 Sound CODEC Linux Driver
- Firmware Loader for SigmaDSPs

TOOLS AND SIMULATIONS 🖵

- SigmaDSP Processors: Software and Tools
- ADAU1361/ADAU1761 IBIS Models

DESIGN RESOURCES 🖵

- ADAU1461 Material Declaration
- PCN-PDN Information
- Quality And Reliability
- Symbols and Footprints

DISCUSSIONS 🖵

View all ADAU1461 EngineerZone Discussions.

SAMPLE AND BUY 🖵

Visit the product page to see pricing options.

TECHNICAL SUPPORT 🖵

Submit a technical question or find your regional support number.

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REVISION HISTORY

6/10—Revision 0: Initial Version

SPECIFICATIONS

Supply voltage (AVDD) = 3.3 V, T_A = 25°C, master clock = 12.288 MHz (48 kHz f_S, 256 × f_S mode), input sample rate = 48 kHz, measurement bandwidth = 20 Hz to 20 kHz, word width = 24 bits, C_{LOAD} (digital output) = 20 pF, I_{LOAD} (digital output) = 2 mA, V_{IH} = 2 V, V_{IL} = 0.8 V, unless otherwise noted. Performance of all channels is identical, exclusive of the interchannel gain mismatch and interchannel phase deviation specifications.

ANALOG PERFORMANCE SPECIFICATIONS, T_A = 25°C

IOVDD = 3.3 V ± 10%.

Table 1.

Parameter	Test Conditions/Comments	Min	Typ	Max	Unit
ANALOG-TO-DIGITAL CONVERTERS	ADC performance excludes mixers and PGA				
ADC Resolution	All ADCs		24		Bits
Digital Attenuation Step			0.375		dB
Digital Attenuation Range			95		dB
INPUT RESISTANCE					
Single-Ended Line Input	−12 dB gain		80.4		kΩ
	0 dB gain		21		kΩ
	6 dB gain		10.5		kΩ
PGA Inverting Inputs	−12 dB gain		84.5		kΩ
	0 dB gain		53		kΩ
	35.25 dB gain		1.7		kΩ
PGA Noninverting Inputs	All gains		105		kΩ
SINGLE-ENDED LINE INPUT					
Full-Scale Input Voltage (0 dB)			1.0 (2.83)		V rms (V p-p)
Dynamic Range	20 Hz to 20 kHz, −60 dB input				
With A-Weighted Filter (RMS)		83.5	99		dB
No Filter (RMS)		83	96		dB
Total Harmonic Distortion + Noise	−1 dBFS		−90	−71	dB
Signal-to-Noise Ratio					
With A-Weighted Filter (RMS)			99		dB
No Filter (RMS)			96		dB
Input Mixer Gain per Step	−12 dB to +6 dB range	2.89	3	3.07	dB
Mute Attenuation	LINPG[2:0], LINNG[2:0] = 000, RINPG[2:0], RINNG[2:0] = 000, MX1AUXG[2:0], MX2AUXG[2:0] = 000		−85.5	−77	dB
Interchannel Gain Mismatch		−0.3	+0.032	+0.3	dB
Offset Error		−5	0	+5	mV
Gain Error		−17	−12	−8	%
Interchannel Isolation			68		dB
Power Supply Rejection Ratio	CM capacitor = 20 µF, 100 mV p-p @ 1 kHz		67		dB
PSEUDO-DIFFERENTIAL PGA INPUT					
Full-Scale Input Voltage (0 dB)			1.0 (2.83)		V rms (V p-p)
Dynamic Range	20 Hz to 20 kHz, −60 dB input				
With A-Weighted Filter (RMS)		94	98		dB
No Filter (RMS)		91	95		dB
Total Harmonic Distortion + Noise	−1 dBFS		−89	−83	dB
Signal-to-Noise Ratio					
With A-Weighted Filter (RMS)			98		dB
No Filter (RMS)			95		dB
PGA Boost Gain Error	20 dB gain setting (RDBOOST[1:0], LDBOOST[1:0] = 10)	−8	+0.4	+8	dB

ADAU1461

Parameter	Test Conditions/Comments	Min	Typ	Max	Unit
Mute Attenuation	PGA muted				
	LDMUTE, RDMUTE = 0		−76	−73	dB
	RDBOOST[1:0], LDBOOST[1:0] = 00		−87	−82	dB
Interchannel Gain Mismatch		−0.6	−0.073	+0.6	dB
Offset Error		−6	0	+6	mV
Gain Error		−24	−14	−3	%
Interchannel Isolation			83		dB
Common-Mode Rejection Ratio	100 mV rms, 1 kHz		−58		dB
	100 mV rms, 20 kHz	−52	−48	−44	dB
FULL DIFFERENTIAL PGA INPUT	Differential PGA inputs				
Full-Scale Input Voltage (0 dB)			1.0 (2.83)		V rms (V p-p)
Dynamic Range	20 Hz to 20 kHz, −60 dB input				
With A-Weighted Filter (RMS)		94	98		dB
No Filter (RMS)		91	95		dB
Total Harmonic Distortion + Noise	−1 dBFS		−78	−74	dB
Signal-to-Noise Ratio					
With A-Weighted Filter (RMS)			98		dB
No Filter (RMS)			95		dB
PGA Boost Gain Error	20 dB gain setting (RDBOOST[1:0], LDBOOST[1:0] = 10)	−8	−0.15	+8	dB
Mute Attenuation	PGA muted				
	LDMUTE, RDMUTE = 0		−76	−73	dB
	RDBOOST[1:0], LDBOOST[1:0] = 00		−87	−82	dB
Interchannel Gain Mismatch		−0.3	−0.0005	+0.3	dB
Offset Error		−6	0	+6	mV
Gain Error		−17	−14	−9	%
Interchannel Isolation			83		dB
Common-Mode Rejection Ratio	100 mV rms, 1 kHz		−58		dB
	100 mV rms, 20 kHz	−52	−48	−44	dB
MICROPHONE BIAS	MBIEN = 1				
Bias Voltage					
0.65 × AVDD	MBI = 1, MPERF = 0	2.00	2.145	2.19	V
	MBI = 1, MPERF = 1	2.04	2.13	2.21	V
0.90 × AVDD	MBI = 0, MPERF = 0	2.89	2.97	3.04	V
	MBI = 0, MPERF = 1	2.89	2.99	3.11	V
Bias Current Source	MBI = 0, MPERF = 1			3	mA
Noise in the Signal Bandwidth	1 kHz to 20 kHz				
	MBI = 0, MPERF = 0		42		nV/√Hz
	MBI = 0, MPERF = 1		85		nV/√Hz
	MBI = 1, MPERF = 0		25		nV/√Hz
	MBI = 1, MPERF = 1	13	22	36	nV/√Hz
DIGITAL-TO-ANALOG CONVERTERS	DAC performance excludes mixers and headphone amplifier				
DAC Resolution	All DACs		24		Bits
Digital Attenuation Step			0.375		dB
Digital Attenuation Range			95		dB
DAC TO LINE OUTPUT					
Full-Scale Output Voltage (0 dB)			0.92 (2.60)		V rms (V p-p)
Dynamic Range	20 Hz to 20 kHz, −60 dBFS input, line output mode				
With A-Weighted Filter (RMS)		95	101		dB
No Filter (RMS)		93.5	98		dB

Parameter	Test Conditions/Comments	Min	Typ	Max	Unit
Total Harmonic Distortion + Noise	0 dBFS, 10 kΩ load				
Line Output Mode			−92	−77	dB
Headphone Output Mode			−89	−79	dB
Signal-to-Noise Ratio	Line output mode				
With A-Weighted Filter (RMS)			101		dB
No Filter (RMS)			98		dB
Mute Attenuation					
Mixer 3 and Mixer 4 Muted	MX3RM, MX3LM, MX4RM, MX4LM = 0, MX3AUXG[3:0], MX4AUXG[3:0] = 0000, MX3G1[3:0], MX3G2[3:0] = 0000, MX4G1[3:0], MX4G2[3:0] = 0000		−85	−78	dB
Mixer 5, Mixer 6, and Mixer 7 Muted	MX5G3[1:0], MX5G4[1:0], MX6G3[1:0], MX6G4[1:0], MX7[1:0] = 00		−89	−80	dB
All Volume Controls Muted	LOUTM, ROUTM = 0		−82	−74	dB
	MONOM, LHPM, RHPM = 0		−74	−69	dB
Interchannel Gain Mismatch		−0.3	−0.005	+0.3	dB
Offset Error		−22	0	+22	mV
Gain Error		−10	+3	+10	%
Interchannel Isolation	1 kHz, 0 dBFS input signal		100		dB
Power Supply Rejection Ratio	CM capacitor = 20 µF, 100 mV p-p @ 1 kHz		70		dB
DAC TO HEADPHONE/EARPIECE OUTPUT	LOUTx, ROUTx, LHP, RHP in headphone output mode; P_O = output power per channel				
Full-Scale Output Voltage (0 dB)	Scales linearly with AVDD		0.92 (2.60)		V rms (V p-p)
Total Harmonic Distortion + Noise	−4 dBFS, 16 Ω load, P_O = 21.1 mW		−82		dB
	−4 dBFS, 32 Ω load, P_O = 10.6 mW		−82		dB
Capless Headphone Mode	−2 dBFS, 16 Ω load		−78	−71	dB
	−2 dBFS, 32 Ω load		−75	−65	dB
Headphone Output Mode	0 dBFS, 10 kΩ load		−86	−77	dB
Interchannel Isolation	1 kHz, 0 dBFS input signal, 32 Ω load				
	Referred to GND		73		dB
	Referred to CM (capless headphone mode)		50		dB
Power Supply Rejection Ratio	CM capacitor = 20 µF, 100 mV p-p @ 1 kHz		67		dB
REFERENCE					
Common-Mode Reference Output	CM pin	1.62	1.65	1.67	V

ANALOG PERFORMANCE SPECIFICATIONS, −40°C < T$_A$ < +105°C

IOVDD = 3.3 V ± 10%.

Table 2.

Parameter	Test Conditions/Comments	Min	Typ	Max	Unit
SINGLE-ENDED LINE INPUT					
Dynamic Range	20 Hz to 20 kHz, −60 dB input				
With A-Weighted Filter (RMS)		74			dB
No Filter (RMS)		71			dB
Total Harmonic Distortion + Noise	−1 dBFS			−67	dB
Input Mixer Gain per Step	−12 dB to +6 dB range	2.88		3.09	dB
Mute Attenuation	LINPG[2:0], LINNG[2:0] = 000, RINPG[2:0], RINNG[2:0] = 000, MX1AUXG[2:0], MX2AUXG[2:0] = 000			−77	dB
Interchannel Gain Mismatch		−0.5		+0.5	dB
Offset Error		−5		+5	mV
Gain Error		−22		−6	%

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Parameter	Test Conditions/Comments	Min	Typ	Max	Unit
PSEUDO-DIFFERENTIAL PGA INPUT					
Dynamic Range	20 Hz to 20 kHz, −60 dB input				
With A-Weighted Filter (RMS)		94			dB
No Filter (RMS)		91			dB
Total Harmonic Distortion + Noise	−1 dBFS			−75	dB
PGA Boost Gain Error	20 dB gain setting (RDBOOST[1:0], LDBOOST[1:0] = 10)	−11		−7	dB
Mute Attenuation	PGA muted				
	LDMUTE, RDMUTE = 0			−73	dB
	RDBOOST[1:0], LDBOOST[1:0] = 00			−82	dB
Interchannel Gain Mismatch		−0.6		+0.6	dB
Offset Error		−6		+6	mV
Gain Error		−24		−3	%
Common-Mode Rejection Ratio	100 mV rms, 1 kHz	−64		−38	dB
	100 mV rms, 20 kHz	−53		−43	dB
FULL DIFFERENTIAL PGA INPUT	Differential PGA inputs				
Dynamic Range	20 Hz to 20 kHz, −60 dB input				
With A-Weighted Filter (RMS)		89			dB
No Filter (RMS)		86			dB
Total Harmonic Distortion + Noise	−1 dBFS			−70	dB
PGA Boost Gain Error	20 dB gain setting (RDBOOST[1:0], LDBOOST[1:0] = 10)	−11		−7	dB
Mute Attenuation	PGA muted				
	LDMUTE, RDMUTE = 0			−73	dB
	RDBOOST[1:0], LDBOOST[1:0] = 00			−82	dB
Interchannel Gain Mismatch		−0.4		+0.4	dB
Offset Error		−6		+6	mV
Gain Error		−21		−7	%
Common-Mode Rejection Ratio	100 mV rms, 1 kHz	−64		−38	dB
	100 mV rms, 20 kHz	−53		−43	dB
MICROPHONE BIAS	MBIEN = 1				
Bias Voltage					
0.65 × AVDD	MBI = 1, MPERF = 0	1.85		2.45	V
	MBI = 1, MPERF = 1	1.87		2.45	V
0.90 × AVDD	MBI = 0, MPERF = 0	2.65		3.40	V
	MBI = 0, MPERF = 1	2.65		3.40	V
Noise in the Signal Bandwidth	1 kHz to 20 kHz	11		36	nV/√Hz
DAC TO LINE OUTPUT					
Dynamic Range	20 Hz to 20 kHz, −60 dB input, line output mode				
With A-Weighted Filter (RMS)		85			dB
No Filter (RMS)		78			dB
Total Harmonic Distortion + Noise	0 dBFS, 10 kΩ load				
Line Output Mode				−76	dB
Headphone Output Mode				−78	dB
Mute Attenuation					
Mixer 3 and Mixer 4 Muted	MX3RM, MX3LM, MX4RM, MX4LM = 0, MX3AUXG[3:0], MX4AUXG[3:0] = 0000, MX3G1[3:0], MX3G2[3:0] = 0000, MX4G1[3:0], MX4G2[3:0] = 0000			−77	dB
Mixer 5, Mixer 6, and Mixer 7 Muted	MX5G3[1:0], MX5G4[1:0], MX6G3[1:0], MX6G4[1:0], MX7[1:0] = 00			−77	dB
All Volume Controls Muted	LOUTM, ROUTM = 0			−74	dB
	MONOM, LHPM, RHPM = 0			−69	dB

Parameter	Test Conditions/Comments	Min	Typ	Max	Unit
Interchannel Gain Mismatch		−0.3		+0.3	dB
Offset Error		−22		+22	mV
Gain Error		−10		+10	%
DAC TO HEADPHONE/EARPIECE OUTPUT	LOUTx, ROUTx, LHP, RHP in headphone output mode; P_O = output power per channel				
Total Harmonic Distortion + Noise					
Capless Headphone Mode	−2 dBFS, 16 Ω load			−61	dB
	−2 dBFS, 32 Ω load			−63	dB
Headphone Output Mode	0 dBFS, 10 kΩ load			−76	dB
REFERENCE					
Common-Mode Reference Output	CM pin	1.47		1.83	V

POWER SUPPLY SPECIFICATIONS

Master clock = 12.288 MHz, input sample rate = 48 kHz, input tone = 1 kHz, ADC input @ −1 dBFS, DAC input @ 0 dBFS, −40°C < T_A < +105°C, IOVDD = 3.3 V ± 10%. For total power consumption, add the IOVDD current listed in Table 3.

Table 3.

Parameter	Test Conditions/Comments	Min	Typ	Max	Unit
SUPPLIES					
Voltage	DVDDOUT		1.56		V
	AVDD	2.97	3.3	3.65	V
	IOVDD	2.97	3.3	3.65	V
Digital I/O Current (IOVDD)	20 pF capacitive load on all digital pins				
Slave Mode	f_S = 48 kHz		0.48		mA
	f_S = 96 kHz		0.9		mA
	f_S = 8 kHz		0.13		mA
Master Mode	f_S = 48 kHz		1.51		mA
	f_S = 96 kHz		3		mA
	f_S = 8 kHz		0.27		mA
Analog Current (AVDD)					
Record Stereo Differential to ADC					
	PLL bypass		5.24		mA
	Integer PLL		6.57		mA
DAC Stereo Playback to Line Output	10 kΩ load				
	PLL bypass		5.55		mA
	Integer PLL		6.90		mA
DAC Stereo Playback to Headphone	32 Ω load				
	PLL bypass		30.9		mA
	Integer PLL		32.25		mA
DAC Stereo Playback to Capless Headphone	32 Ω load				
	PLL bypass		56.75		mA
	Integer PLL		58		mA

DIGITAL FILTERS

Table 4.

Parameter	Mode	Factor	Min	Typ	Max	Unit
ADC DECIMATION FILTER	All modes, typ @ 48 kHz					
Pass Band		0.4375 f_S		21		kHz
Pass-Band Ripple				±0.015		dB
Transition Band		0.5 f_S		24		kHz
Stop Band		0.5625 f_S		27		kHz
Stop-Band Attenuation				67		dB
Group Delay		22.9844/f_S		479		μs
DAC INTERPOLATION FILTER						
Pass Band	48 kHz mode, typ @ 48 kHz	0.4535 f_S		22		kHz
	96 kHz mode, typ @ 96 kHz	0.3646 f_S		35		kHz
Pass-Band Ripple	48 kHz mode, typ @ 48 kHz				±0.01	dB
	96 kHz mode, typ @ 96 kHz				±0.05	dB
Transition Band	48 kHz mode, typ @ 48 kHz	0.5 f_S		24		kHz
	96 kHz mode, typ @ 96 kHz	0.5 f_S		48		kHz
Stop Band	48 kHz mode, typ @ 48 kHz	0.5465 f_S		26		kHz
	96 kHz mode, typ @ 96 kHz	0.6354 f_S		61		kHz
Stop-Band Attenuation	48 kHz mode, typ @ 48 kHz			69		dB
	96 kHz mode, typ @ 96 kHz			68		dB
Group Delay	48 kHz mode, typ @ 48 kHz	25/f_S		521		μs
	96 kHz mode, typ @ 96 kHz	11/f_S		115		μs

DIGITAL INPUT/OUTPUT SPECIFICATIONS

−40°C < T_A < +105°C, IOVDD = 3.3 V ± 10%.

Table 5.

| Parameter	Test Conditions/Comments	Min	Typ	Max	Unit
INPUT SPECIFICATIONS					
Input Voltage High (V_{IH})		0.7 × IOVDD			V
Input Voltage Low (V_{IL})				0.3 × IOVDD	V
Input Leakage					
Pull-Ups/Pull-Downs Disabled	I_{IH} @ V_{IH} = 3.3 V	−0.17		+0.17	μA
	I_{IL} @ V_{IL} = 0 V	−0.17		+0.17	μA
	I_{IL} @ V_{IL} = 0 V (MCLK pin)	−13.5		−0.5	μA
Pull-Ups Enabled	I_{IH} @ V_{IH} = 3.3 V	−0.7		+0.7	μA
	I_{IL} @ V_{IL} = 0 V	−13.5		−0.5	μA
Pull-Downs Enabled	I_{IH} @ V_{IH} = 3.3 V	2.7		8.3	μA
	I_{IL} @ V_{IL} = 0 V	−0.18		+0.18	μA
Input Capacitance				5	pF
OUTPUT SPECIFICATIONS					
Output Voltage High (V_{OH})	I_{OH} = 2 mA @ 3.3 V	0.8 × IOVDD			V
Output Voltage Low (V_{OL})	I_{OL} = 2 mA @ 3.3 V			0.1 × IOVDD	V

DIGITAL TIMING SPECIFICATIONS

−40°C < T$_A$ < +105°C, IOVDD = 3.3 V ± 10%.

Table 6. Digital Timing

| Parameter | Limit | | Unit | Description |
	t$_{MIN}$	t$_{MAX}$		
MASTER CLOCK				
t$_{MP}$	74	488	ns	MCLK period, 256 × f$_S$ mode.
t$_{MP}$	37	244	ns	MCLK period, 512 × f$_S$ mode.
t$_{MP}$	24.7	162.7	ns	MCLK period, 768 × f$_S$ mode.
t$_{MP}$	18.5	122	ns	MCLK period, 1024 × f$_S$ mode.
SERIAL PORT				
t$_{BIL}$	5		ns	BCLK pulse width low.
t$_{BIH}$	5		ns	BCLK pulse width high.
t$_{LIS}$	5		ns	LRCLK setup. Time to BCLK rising.
t$_{LIH}$	5		ns	LRCLK hold. Time from BCLK rising.
t$_{SIS}$	5		ns	DAC_SDATA setup. Time to BCLK rising.
t$_{SIH}$	5		ns	DAC_SDATA hold. Time from BCLK rising.
t$_{SODM}$		50	ns	ADC_SDATA delay. Time from BCLK falling in master mode.
SPI PORT				
f$_{CCLK}$		10	MHz	CCLK frequency.
t$_{CCPL}$	10		ns	CCLK pulse width low.
t$_{CCPH}$	10		ns	CCLK pulse width high.
t$_{CLS}$	5		ns	$\overline{\text{CLATCH}}$ setup. Time to CCLK rising.
t$_{CLH}$	10		ns	$\overline{\text{CLATCH}}$ hold. Time from CCLK rising.
t$_{CLPH}$	10		ns	$\overline{\text{CLATCH}}$ pulse width high.
t$_{CDS}$	5		ns	CDATA setup. Time to CCLK rising.
t$_{CDH}$	5		ns	CDATA hold. Time from CCLK rising.
t$_{COD}$		50	ns	COUT three-stated. Time from $\overline{\text{CLATCH}}$ rising.
I²C PORT				
f$_{SCL}$		400	kHz	SCL frequency.
t$_{SCLH}$	0.6		µs	SCL high.
t$_{SCLL}$	1.3		µs	SCL low.
t$_{SCS}$	0.6		µs	Setup time; relevant for repeated start condition.
t$_{SCH}$	0.6		µs	Hold time. After this period, the first clock is generated.
t$_{DS}$	100		ns	Data setup time.
t$_{SCR}$		300	ns	SCL rise time.
t$_{SCF}$		300	ns	SCL fall time.
t$_{SDR}$		300	ns	SDA rise time.
t$_{SDF}$		300	ns	SDA fall time.
t$_{BFT}$	0.6		µs	Bus-free time. Time between stop and start.
DIGITAL MICROPHONE				R$_{LOAD}$ = 1 MΩ, C$_{LOAD}$ = 14 pF.
t$_{DCF}$		10	ns	Digital microphone clock fall time.
t$_{DCR}$		10	ns	Digital microphone clock rise time.
t$_{DDV}$	22	30	ns	Digital microphone delay time for valid data.
t$_{DDH}$	0	12	ns	Digital microphone delay time for data three-stated.

DIGITAL TIMING DIAGRAMS

Figure 2. Serial Input Port Timing

Figure 3. Serial Output Port Timing
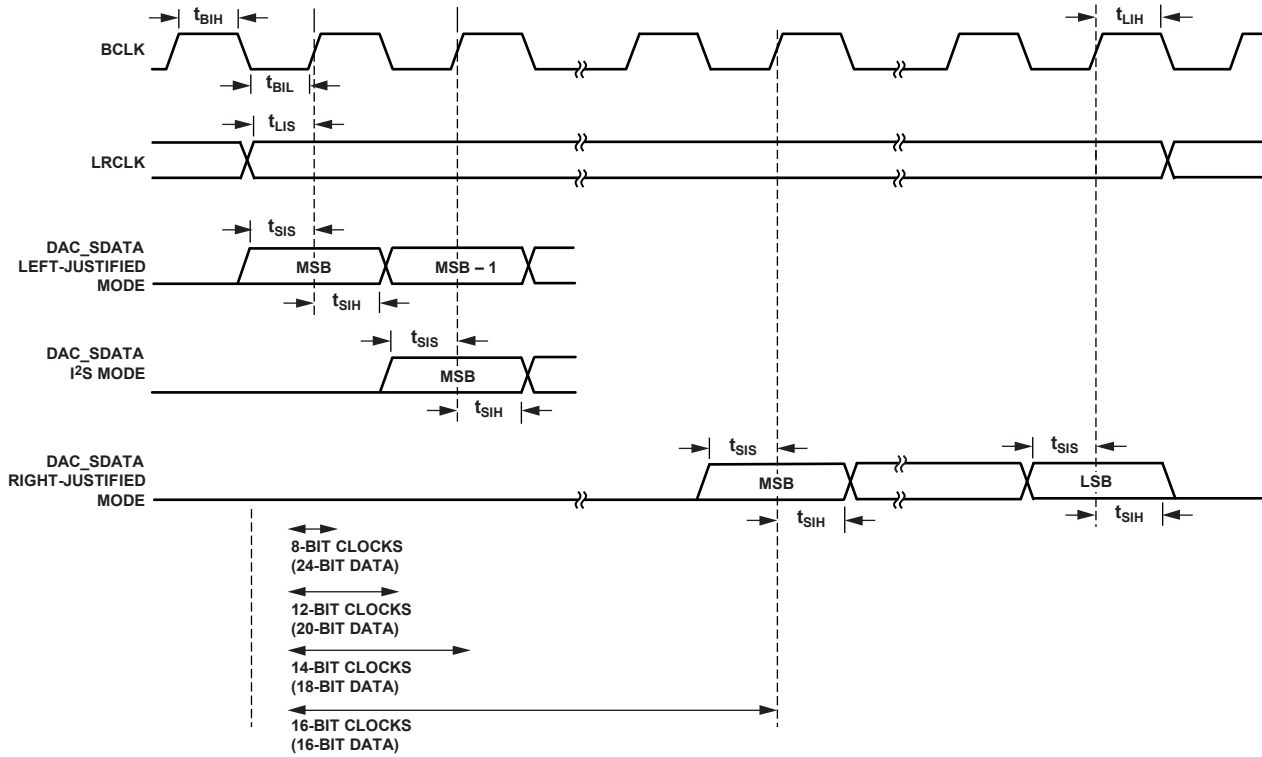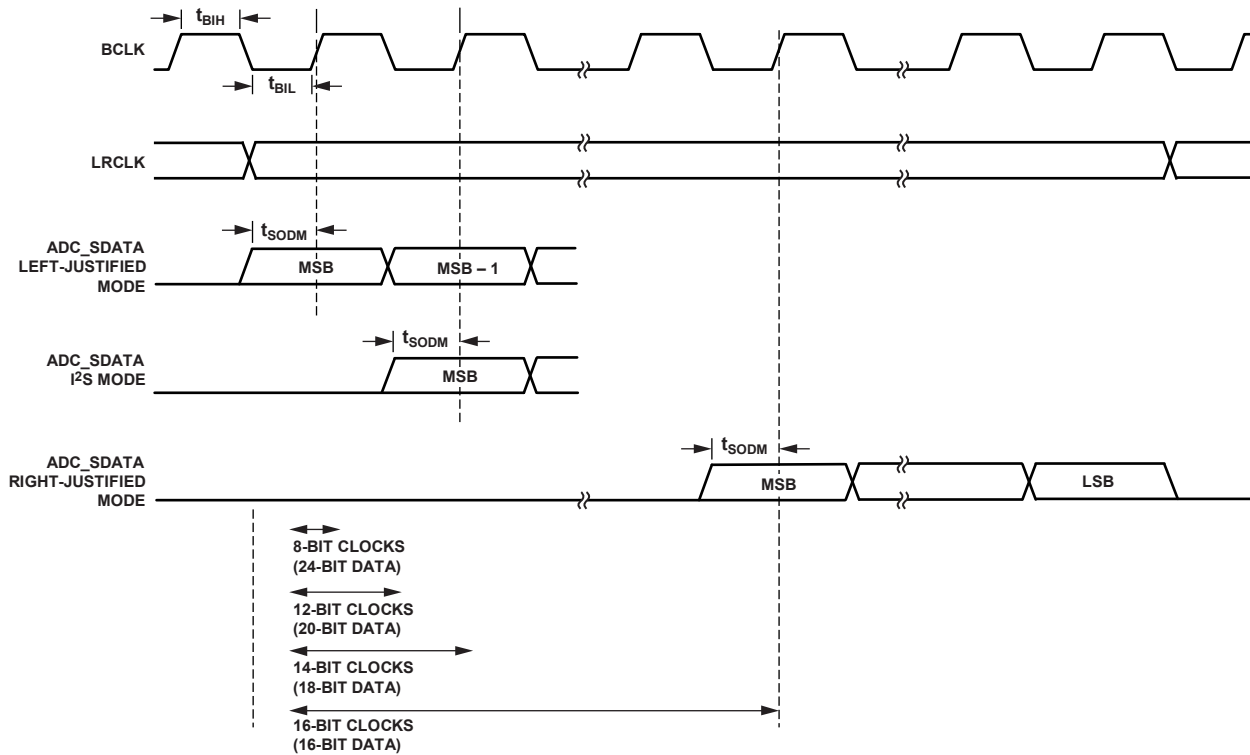
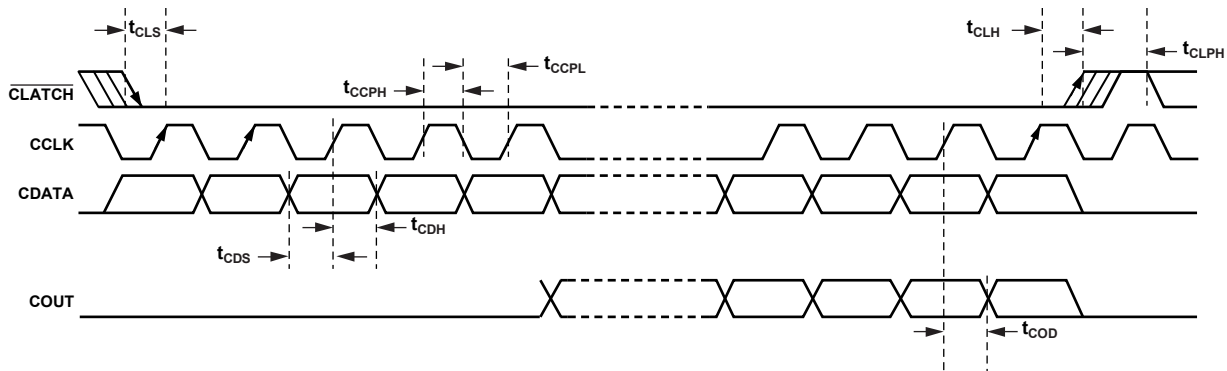*Figure 4. SPI Port Timing*
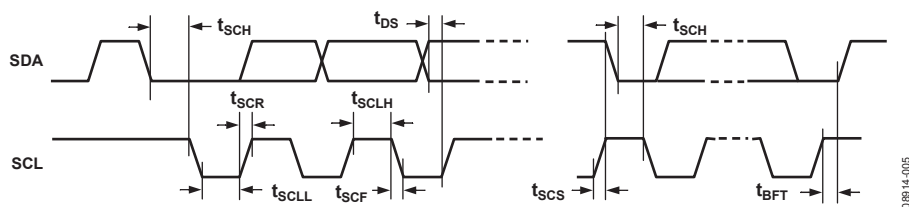
Figure 5. I²C Port Timing
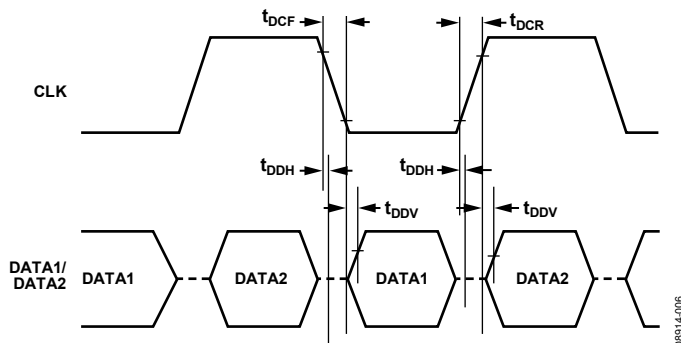
Figure 6. Digital Microphone Timing

ABSOLUTE MAXIMUM RATINGS

Table 7.

Parameter	Rating
Power Supply (AVDD)	−0.3 V to +3.65 V
Input Current (Except Supply Pins)	±20 mA
Analog Input Voltage (Signal Pins)	−0.3 V to AVDD + 0.3 V
Digital Input Voltage (Signal Pins)	−0.3 V to IOVDD + 0.3 V
Operating Temperature Range	−40°C to +105°C
Storage Temperature Range	−65°C to +150°C

Stresses above those listed under Absolute Maximum Ratings may cause permanent damage to the device. This is a stress rating only; functional operation of the device at these or any other conditions above those indicated in the operational section of this specification is not implied. Exposure to absolute maximum rating conditions for extended periods may affect device reliability.

THERMAL RESISTANCE

θ_{JA} represents thermal resistance, junction-to-ambient; θ_{JC} represents thermal resistance, junction-to-case. All characteristics are for a 4-layer board.

Table 8. Thermal Resistance

Package Type	θ_{JA}	θ_{JC}	Unit
32-Lead LFCSP	50.1	17	°C/W

ESD CAUTION

ESD (electrostatic discharge) sensitive device. Charged devices and circuit boards can discharge without detection. Although this product features patented or proprietary protection circuitry, damage may occur on devices subjected to high energy ESD. Therefore, proper ESD precautions should be taken to avoid performance degradation or loss of functionality.

PIN CONFIGURATION AND FUNCTION DESCRIPTIONS
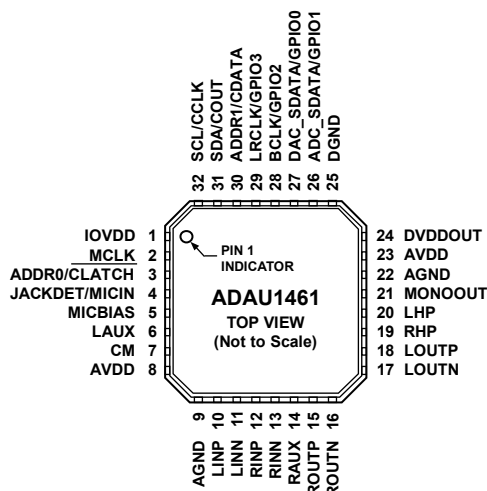
NOTES
1. THE EXPOSED PAD IS CONNECTED INTERNALLY TO THE ADAU1461 GROUNDS. FOR INCREASED RELIABILITY OF THE SOLDER JOINTS AND MAXIMUM THERMAL CAPABILITY, IT IS RECOMMENDED THAT THE PAD BE SOLDERED TO THE GROUND PLANE.

Figure 7. Pin Configuration

Table 9. Pin Function Descriptions

Pin No.	Mnemonic	Type[1]	Description
1	IOVDD	PWR	Supply for Digital Input and Output Pins. The digital output pins are supplied from IOVDD, which also sets the highest input voltage that should be seen on the digital input pins. IOVDD should be set to 3.3 V. The current draw of this pin is variable because it is dependent on the loads of the digital outputs. IOVDD should be decoupled to DGND with a 100 nF capacitor and a 10 µF capacitor.
2	MCLK	D_IN	External Master Clock Input.
3	ADDR0/CLATCH	D_IN	I²C Address Bit 0 (ADDR0). SPI Latch Signal (CLATCH). Must go low at the beginning of an SPI transaction and high at the end of a transaction. Each SPI transaction can take a different number of CCLKs to complete, depending on the address and read/write bit that are sent at the beginning of the SPI transaction.
4	JACKDET/MICIN	D_IN	Detect Insertion/Removal of Headphone Plug (JACKDET). Digital Microphone Stereo Input (MICIN).
5	MICBIAS	A_OUT	Bias Voltage for Electret Microphone.
6	LAUX	A_IN	Left Channel Single-Ended Auxiliary Input. Biased at AVDD/2.
7	CM	A_OUT	AVDD/2 V Common-Mode Reference. A 10 µF to 47 µF standard decoupling capacitor should be connected between this pin and AGND to reduce crosstalk between the ADCs and DACs. This pin can be used to bias external analog circuits, as long as they are not drawing current from CM (for example, the noninverting input of an op amp).
8	AVDD	PWR	3.3 V Analog Supply for DAC and Microphone Bias. This pin should be decoupled locally to AGND with a 100 nF capacitor.
9	AGND	PWR	Analog Ground. The AGND and DGND pins can be tied together on a common ground plane. AGND should be decoupled locally to AVDD with a 100 nF capacitor.
10	LINP	A_IN	Left Channel Noninverting Input or Single-Ended Input 0. Biased at AVDD/2.
11	LINN	A_IN	Left Channel Inverting Input or Single-Ended Input 1. Biased at AVDD/2.
12	RINP	A_IN	Right Channel Noninverting Input or Single-Ended Input 2. Biased at AVDD/2.
13	RINN	A_IN	Right Channel Inverting Input or Single-Ended Input 3. Biased at AVDD/2.
14	RAUX	A_IN	Right Channel Single-Ended Auxiliary Input. Biased at AVDD/2.
15	ROUTP	A_OUT	Right Line Output, Positive. Biased at AVDD/2.
16	ROUTN	A_OUT	Right Line Output, Negative. Biased at AVDD/2.
17	LOUTN	A_OUT	Left Line Output, Negative. Biased at AVDD/2.
18	LOUTP	A_OUT	Left Line Output, Positive. Biased at AVDD/2.

ADAU1461

Pin No.	Mnemonic	Type[1]	Description
19	RHP	A_OUT	Right Headphone Output. Biased at AVDD/2.
20	LHP	A_OUT	Left Headphone Output. Biased at AVDD/2.
21	MONOOUT	A_OUT	Mono Output or Virtual Ground for Capless Headphone. Biased at AVDD/2 when set as mono output.
22	AGND	PWR	Analog Ground. The AGND and DGND pins can be tied together on a common ground plane. AGND should be decoupled locally to AVDD with a 100 nF capacitor.
23	AVDD	PWR	3.3 V Analog Supply for ADC, Output Driver, and Input to Digital Supply Regulator. This pin should be decoupled locally to AGND with a 100 nF capacitor.
24	DVDDOUT	PWR	Digital Core Supply Decoupling Point. The digital supply is generated from an on-board regulator and does not require an external supply. DVDDOUT should be decoupled to DGND with a 100 nF capacitor and a 10 μF capacitor.
25	DGND	PWR	Digital Ground. The AGND and DGND pins can be tied together on a common ground plane. DGND should be decoupled to DVDDOUT and to IOVDD with 100 nF capacitors and 10 μF capacitors.
26	ADC_SDATA/GPIO1	D_IO	ADC Serial Output Data (ADC_SDATA). General-Purpose Input/Output 1 (GPIO1).
27	DAC_SDATA/GPIO0	D_IO	DAC Serial Input Data (DAC_SDATA). General-Purpose Input/Output 0 (GPIO0).
28	BCLK/GPIO2	D_IO	Serial Data Port Bit Clock (BCLK). General-Purpose Input/Output 2 (GPIO2).
29	LRCLK/GPIO3	D_IO	Serial Data Port Frame Clock (LRCLK). General-Purpose Input/Output 3 (GPIO3).
30	ADDR1/CDATA	D_IN	I2C Address Bit 1 (ADDR1). SPI Data Input (CDATA).
31	SDA/COUT	D_IO	I2C Data (SDA). This pin is a bidirectional open-collector input/output. The line connected to this pin should have a 2 kΩ pull-up resistor. SPI Data Output (COUT). This pin is used for reading back registers and memory locations. It is three-state when an SPI read is not active.
32	SCL/CCLK	D_IN	I2C Clock (SCL). This pin is always an open-collector input when in I2C control mode. The line connected to this pin should have a 2 kΩ pull-up resistor. SPI Clock (CCLK). This pin can run continuously or be gated off between SPI transactions.
EP	Exposed Pad		Exposed Pad. The exposed pad is connected internally to the ADAU1461 grounds. For increased reliability of the solder joints and maximum thermal capability, it is recommended that the pad be soldered to the ground plane. See the Exposed Pad PCB Design section for more information.

[1] A_IN = analog input, A_OUT = analog output, D_IN = digital input, D_IO = digital input/output, PWR = power.
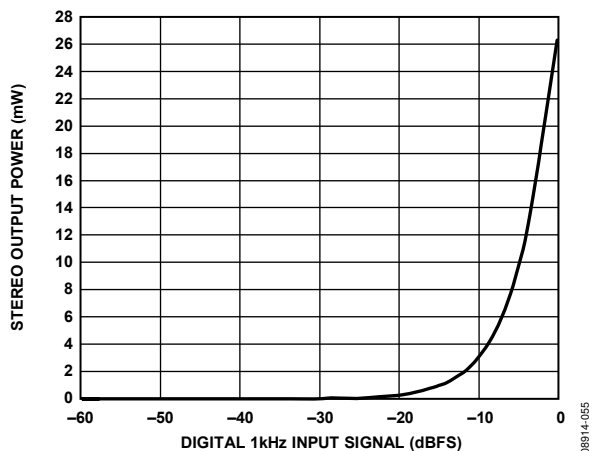
TYPICAL PERFORMANCE CHARACTERISTICS

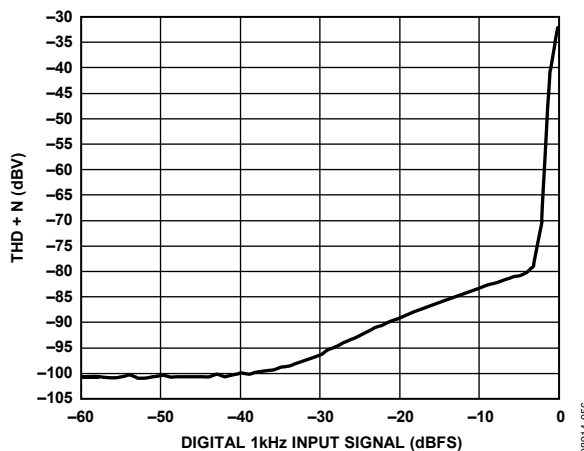Figure 8. Headphone Amplifier Power vs. Input Level, 16 Ω Load

Figure 11. Headphone Amplifier THD + N vs. Input Level, 16 Ω Load
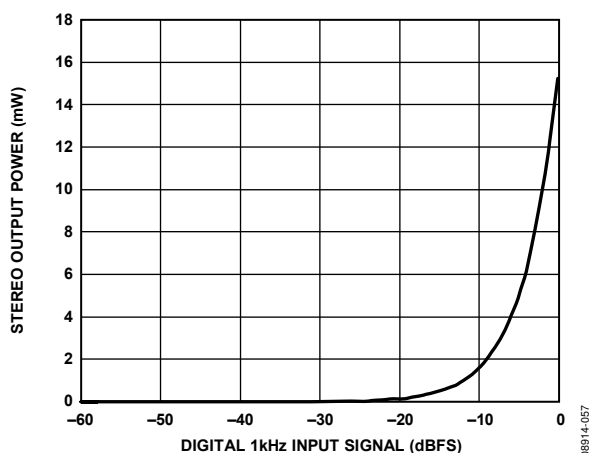

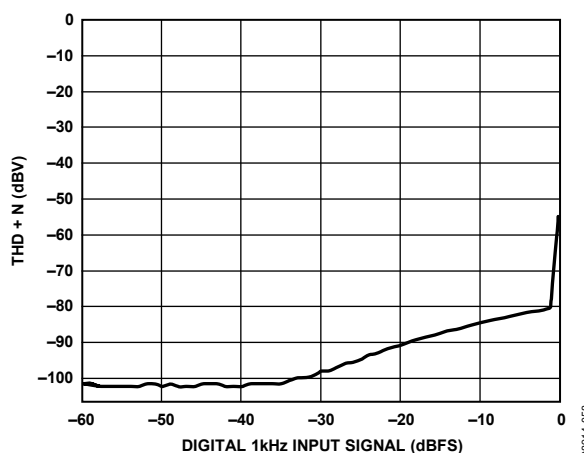Figure 9. Headphone Amplifier Power vs. Input Level, 32 Ω Load

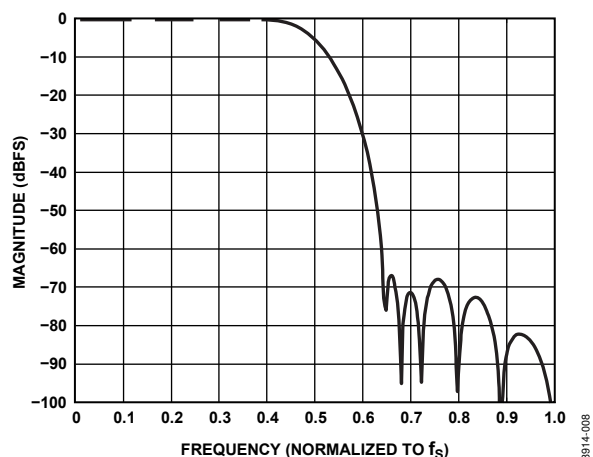Figure 12. Headphone Amplifier THD + N vs. Input Level, 32 Ω Load

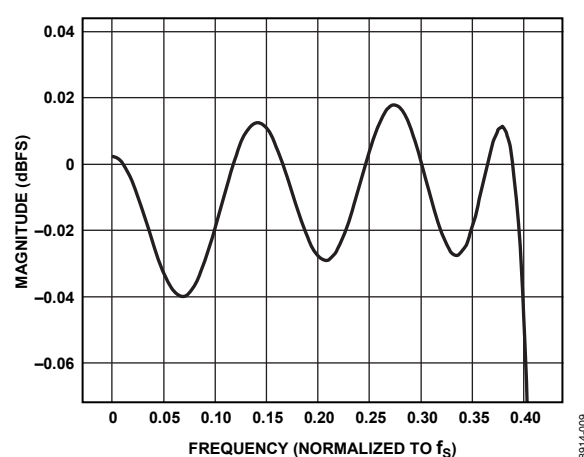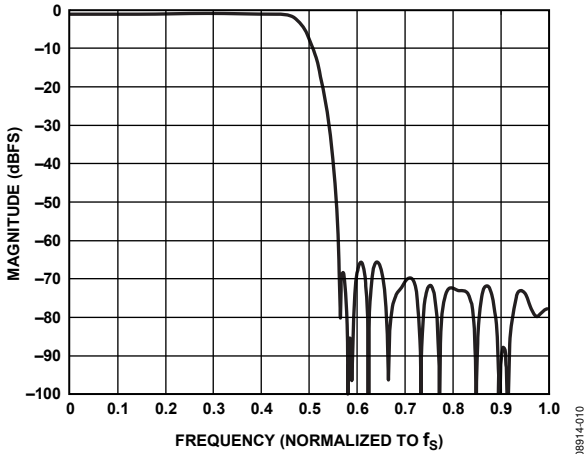Figure 10. ADC Decimation Filter, 64× Oversampling, Normalized to f_S

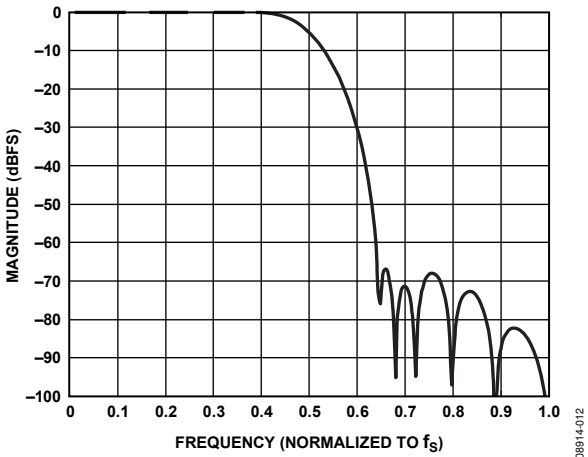Figure 13. ADC Decimation Filter Pass-Band Ripple, 64× Oversampling, Normalized to f_S

Figure 14. ADC Decimation Filter, 128× Oversampling, Normalized to f$_S$
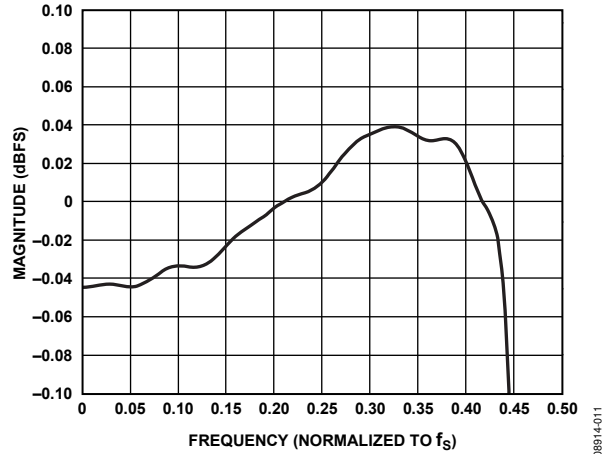
Figure 17. ADC Decimation Filter Pass-Band Ripple, 128× Oversampling, Normalized to f$_S$

Figure 15. ADC Decimation Filter, 128× Oversampling, Double-Rate Mode, Normalized to f$_S$
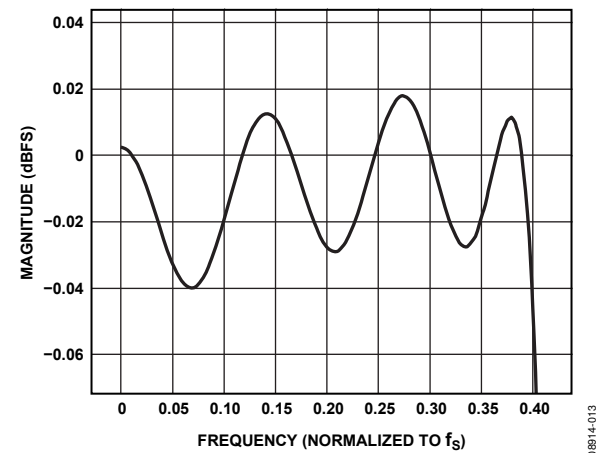
Figure 18. ADC Decimation Filter Pass-Band Ripple, 128× Oversampling, Double-Rate Mode, Normalized to f$_S$
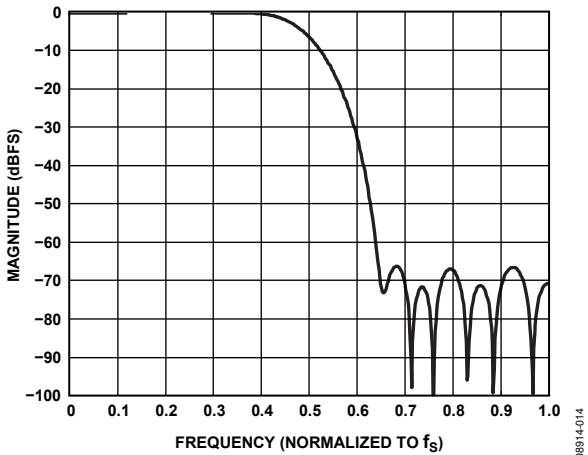
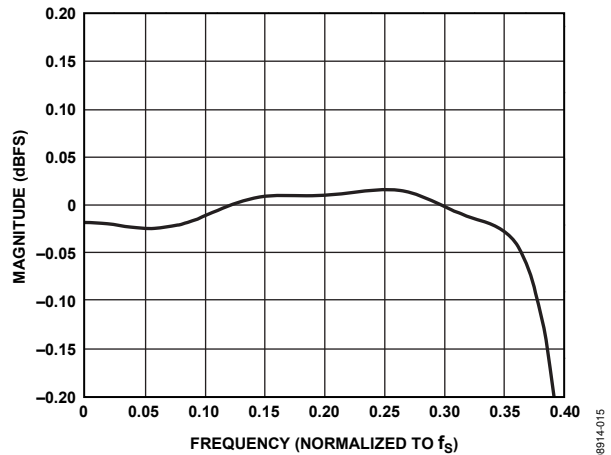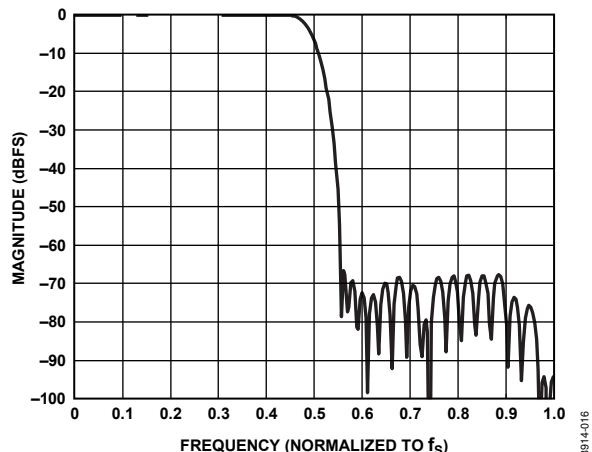Figure 16. DAC Interpolation Filter, 64× Oversampling, Double-Rate Mode, Normalized to f$_S$

Figure 19. DAC Interpolation Filter Pass-Band Ripple, 64× Oversampling, Double-Rate Mode, Normalized to f$_S$

Figure 20. DAC Interpolation Filter, 128× Oversampling, Normalized to f_S
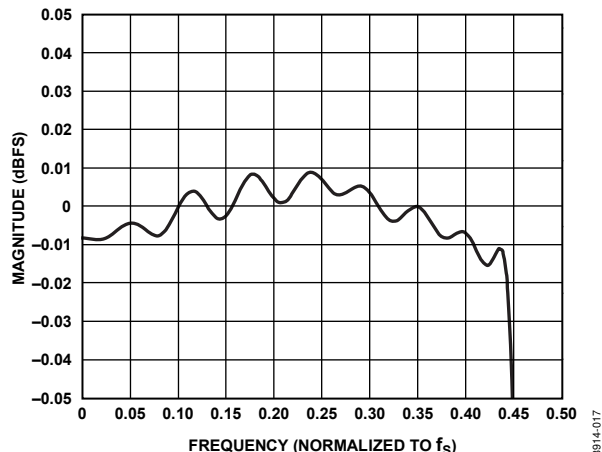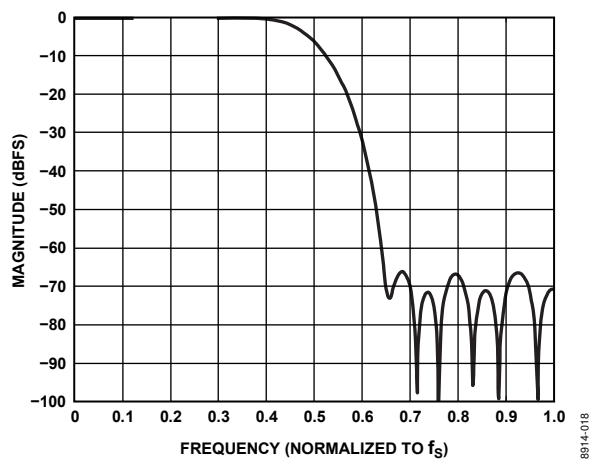

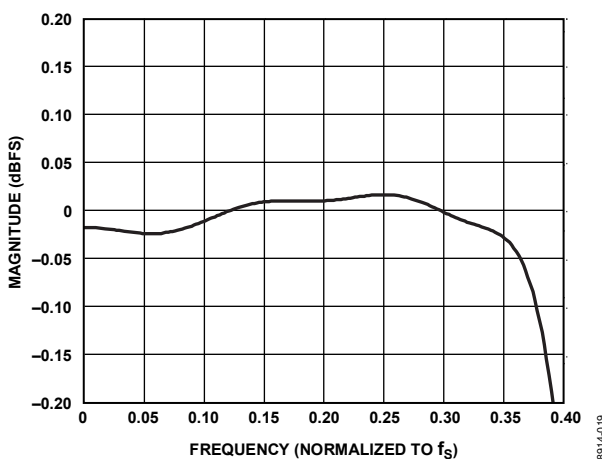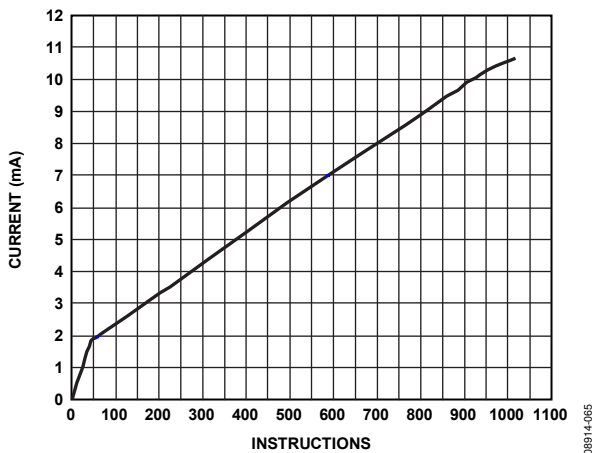Figure 21. DAC Interpolation Filter, 128× Oversampling, Double-Rate Mode, Normalized to f_S

Figure 22. Typical DSP Current Draw

Figure 23. DAC Interpolation Filter Pass-Band Ripple, 128× Oversampling, Normalized to f_S

Figure 24. DAC Interpolation Filter Pass-Band Ripple, 128× Oversampling, Double-Rate Mode, Normalized to f_S
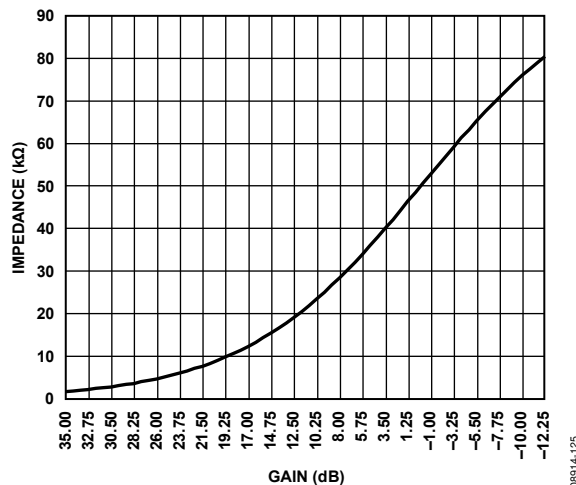

Figure 25. Input Impedance vs. Gain for Analog Inputs
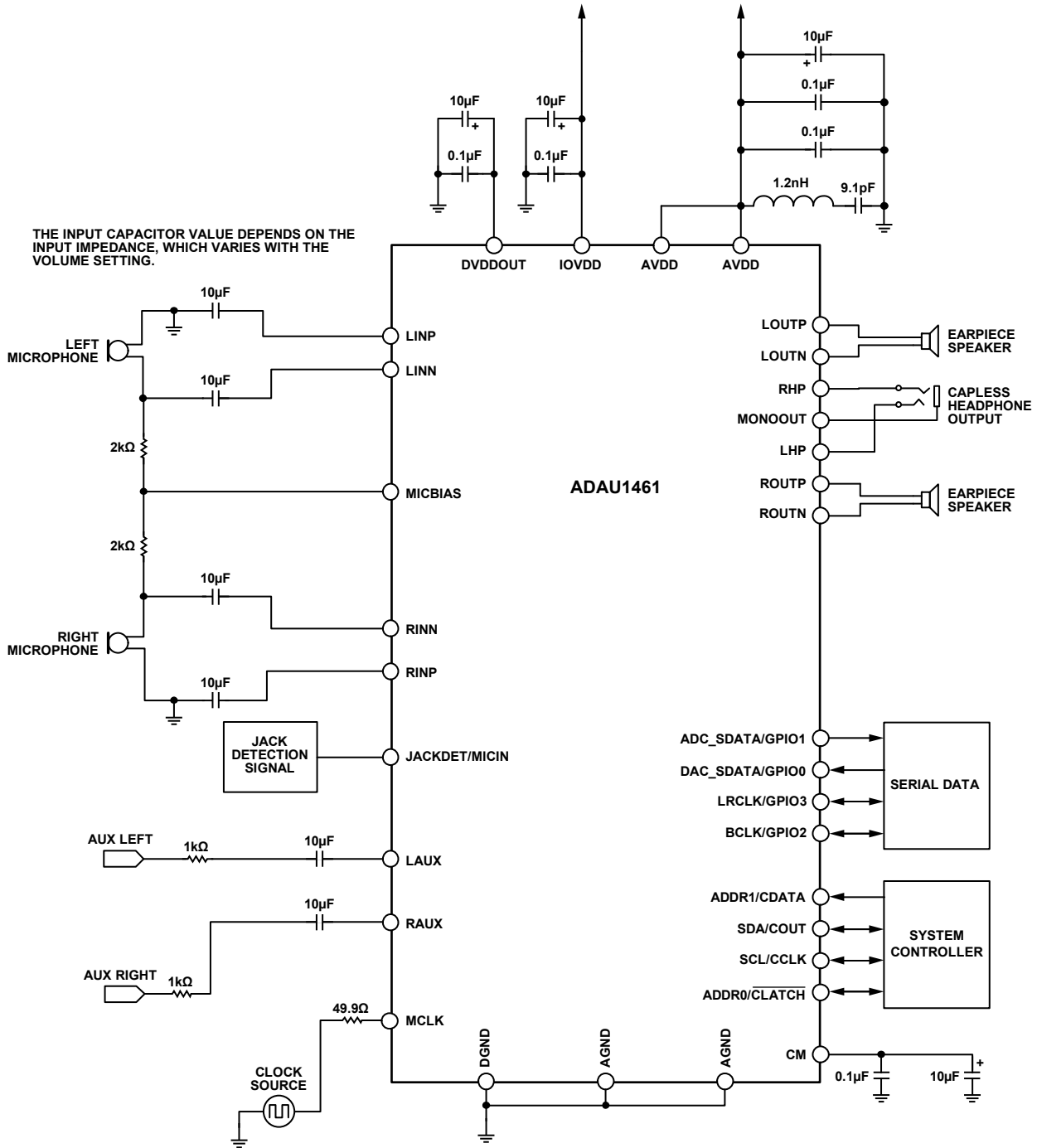
SYSTEM BLOCK DIAGRAMS

THE INPUT CAPACITOR VALUE DEPENDS ON THE
INPUT IMPEDANCE, WHICH VARIES WITH THE
VOLUME SETTING.

Figure 26. System Block Diagram

08914-045

Figure 27. System Block Diagram with Analog Microphones
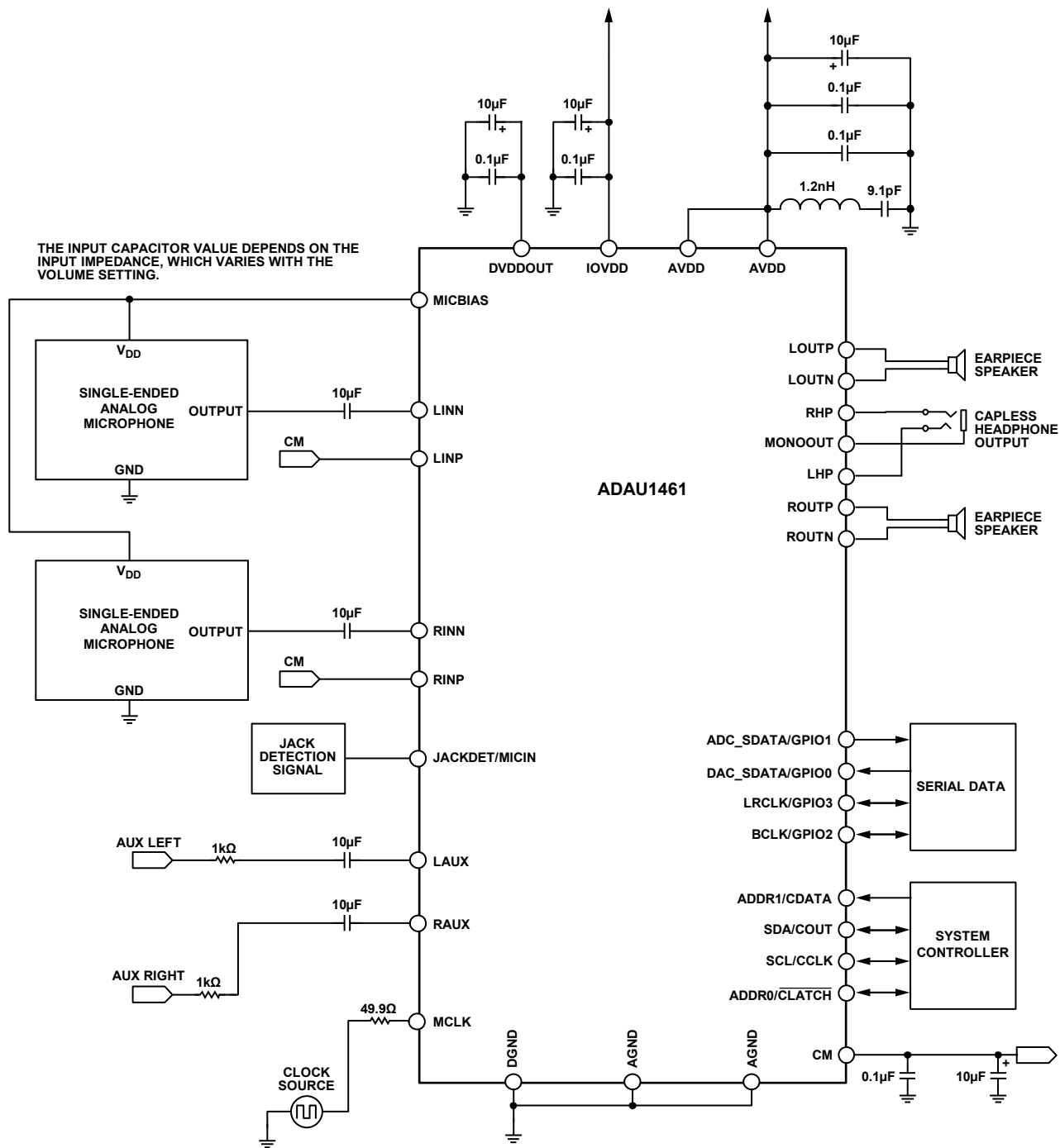
Figure 28. System Block Diagram with Digital Microphones and SSM2306 Class-D Speaker Driver

THEORY OF OPERATION
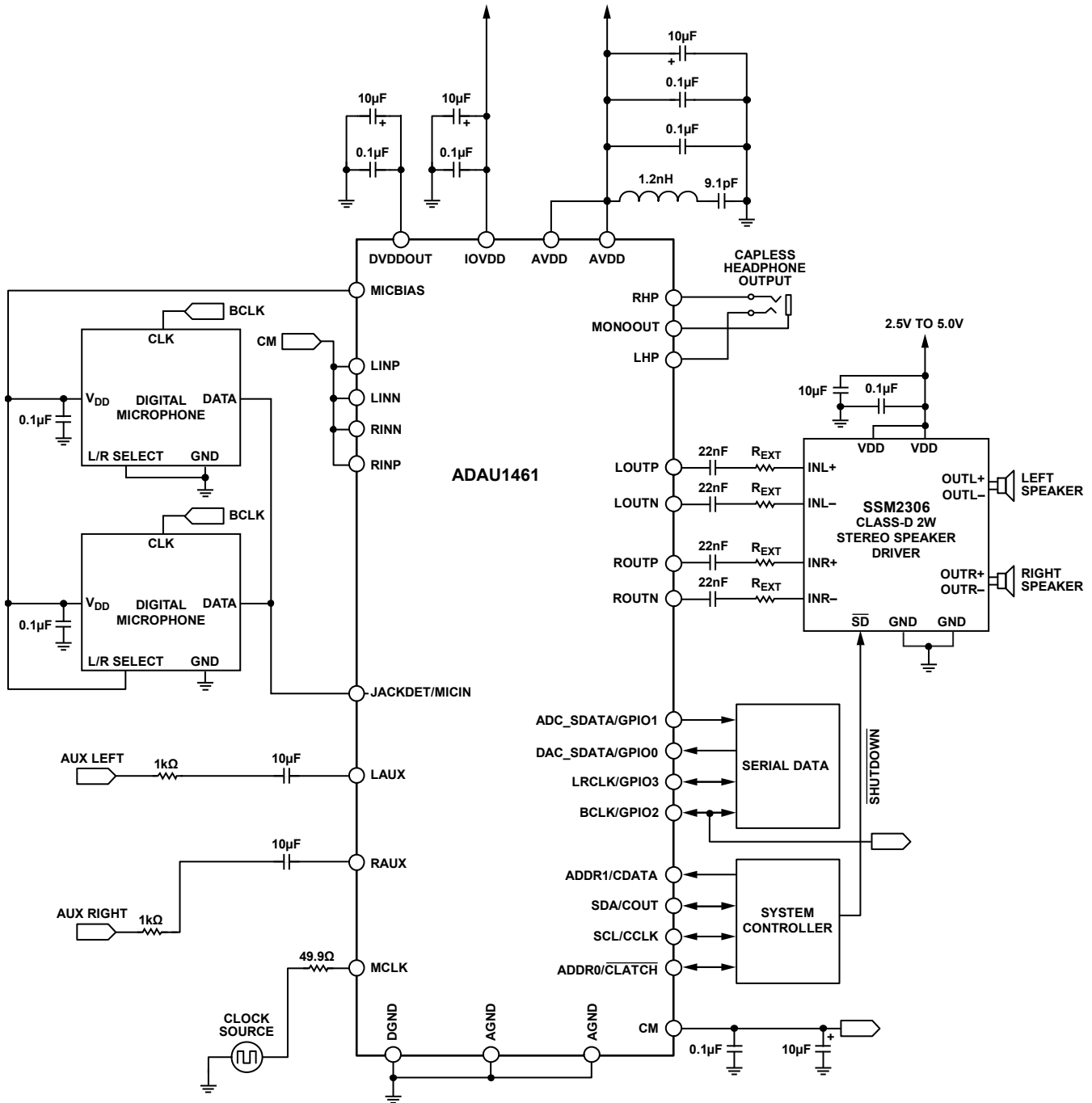
The ADAU1461 is a low power audio codec with an integrated stream-oriented DSP core, making it an all-in-one package that offers high quality audio, low power, small size, and many advanced features. The stereo ADC and stereo DAC each have an SNR of at least +98 dB and a THD + N of at least −90 dB. The serial data port is compatible with I²S, left-justified, right-justified, and TDM modes for interfacing to digital audio data. The operating voltage is 3.3 V, with an on-board regulator generating the internal digital supply voltage.

The record signal path includes very flexible input configurations that can accept differential and single-ended analog microphone inputs as well as a digital microphone input. A microphone bias pin provides seamless interfacing to electret microphones. Input configurations can accept up to six single-ended analog signals or variations of stereo differential or stereo single-ended signals with two additional auxiliary single-ended inputs. Each input signal has its own programmable gain amplifier (PGA) for volume adjustment and can be routed directly to the playback path output mixers, bypassing the ADCs. An automatic level control (ALC) can also be implemented to keep the recording volume constant.

The ADCs and DACs are high quality, 24-bit Σ-Δ converters that operate at selectable 64× or 128× oversampling ratios. The base sampling rate of the converters is set by the input clock rate and can be further scaled with the converter control register settings. The converters can operate at sampling frequencies from 8 kHz to 96 kHz. The ADCs and DACs also include very fine-step digital volume controls.

The playback path allows input signals and DAC outputs to be mixed into various output configurations. Headphone drivers are available for a stereo headphone output, and the other output pins are capable of differentially driving an earpiece speaker. Capless headphone outputs are possible with the use of the mono output as a virtual ground connection. The stereo line outputs can be used as either single-ended or differential outputs and as an optional mix-down mono output.

The DSP core introduces many features that make this codec unique and optimized for audio processing. The program and parameter RAMs can be loaded with custom audio processing signal flow built using the SigmaStudio graphical programming software from Analog Devices, Inc. The values stored in the parameter RAM control individual signal processing blocks, such as equalization filters, dynamics processors, audio delays, and mixer levels.

The SigmaStudio software is used to program and control the SigmaDSP through the control port. Along with designing and tuning a signal flow, the tools can be used to configure all of the DSP registers. The SigmaStudio graphical interface allows anyone with digital or analog audio processing knowledge to easily design DSP signal flow and port it to a target application. At the same time, it provides enough flexibility and programmability for an experienced DSP programmer to have in-depth control of the design. In SigmaStudio, the user can connect graphical blocks (such as biquad filters, dynamics processors, mixers, and delays), compile the design, and load the program and parameter files into the ADAU1461 memory through the control port. Signal processing blocks available in the provided libraries include the following:

- Enhanced stereo capture
- Single- and double-precision biquad filters
- FIR filters
- Dynamics processors with peak or rms detection for mono and multichannel dynamics
- Mixers and splitters
- Tone and noise generators
- Fixed and variable gain
- Loudness
- Delay
- Stereo enhancement
- Dynamic bass boost
- Noise and tone sources
- Level detectors

Additional processing blocks are always being developed. Analog Devices also provides proprietary and third-party algorithms for applications such as matrix decoding, bass enhancement, and surround virtualizers. Contact Analog Devices (www.analog.com) for information about licensing these algorithms.

The ADAU1461 can generate its internal clocks from a wide range of input clocks by using the on-board fractional PLL. The PLL accepts inputs from 8 MHz to 27 MHz.

The ADAU1461 is provided in a small, 32-lead, 5 mm × 5 mm LFCSP with an exposed bottom pad.

STARTUP, INITIALIZATION, AND POWER

This section describes the procedure for properly starting up the ADAU1461. The following sequence provides a high level approach to the proper initiation of the system.

1. Apply power to the ADAU1461.
2. Lock the PLL to the input clock (if using the PLL).
3. Enable the core clock.
4. Load the register settings.

See the Startup section for more information about the proper start-up sequence.

POWER-UP SEQUENCE

The ADAU1461 uses a power-on reset (POR) circuit to reset the registers upon power-up. The POR monitors the DVDDOUT pin and generates a reset signal whenever power is applied to the chip. During the reset, the ADAU1461 is set to the default values documented in the register map (see the Control Registers section). Typically, with a 10 μF capacitor on AVDD, the POR takes approximately 14 ms.
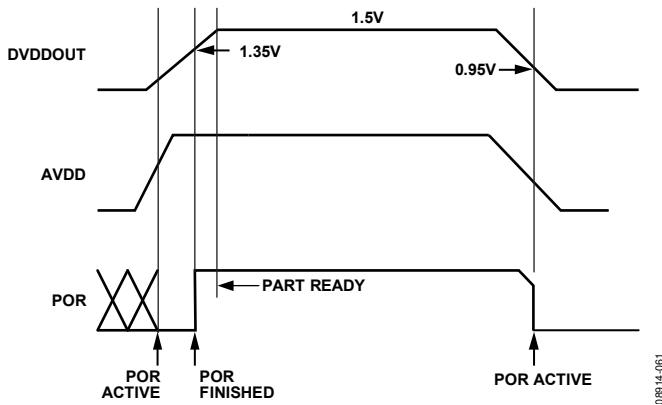

Figure 29. Power-On Reset Sequence

The PLL lock time is dependent on the MCLK rate. Typical lock times are provided in Table 10. The DSP can be enabled immediately after the PLL is locked.

Table 10. PLL Lock Times

PLL Mode	MCLK Frequency	Lock Time (Typical)
Fractional	8 MHz	3.5 ms
Fractional	12 MHz	3.0 ms
Integer	12.288 MHz	2.96 ms
Fractional	13 MHz	2.4 ms
Fractional	14.4 MHz	2.4 ms
Fractional	19.2 MHz	2.98 ms
Fractional	19.68 MHz	2.98 ms
Fractional	19.8 MHz	2.98 ms
Fractional	24 MHz	2.95 ms
Integer	24.576 MHz	2.96 ms
Fractional	26 MHz	2.4 ms
Fractional	27 MHz	2.4 ms

POWER REDUCTION MODES

Sections of the ADAU1461 chip can be turned on and off as needed to reduce power consumption. These include the ADCs, the DACs, the PLL, and the DSP core.

The digital filters of the ADCs and DACs can each be set to over-sampling ratios of 64× or 128× (default). Setting the oversampling ratios to 64× for these filters lowers power consumption with a minimal impact on performance. See the Digital Filters section for specifications; see the Typical Performance Characteristics section for graphs of these filters.

DIGITAL POWER SUPPLY

The digital power supply for the ADAU1461 is generated from an internal regulator. This regulator generates a 1.5 V supply internally. The only external connection to this regulator is the DVDDOUT bypassing point. A 100 nF capacitor and a 10 μF capacitor should be connected between this pin and DGND.

INPUT/OUTPUT POWER SUPPLY

The power for the digital output pins is supplied from IOVDD, and this pin also sets the highest input voltage that should be seen on the digital input pins. IOVDD should be set to 3.3 V; no digital input signal should be at a voltage level higher than the one on IOVDD. The current draw of this pin is variable because it depends on the loads of the digital outputs. IOVDD should be decoupled to DGND with a 100 nF capacitor and a 10 μF capacitor.

CLOCK GENERATION AND MANAGEMENT

The ADAU1461 uses a flexible clocking scheme that enables the use of many different input clock rates. The PLL can be bypassed or used, resulting in two different approaches to clock manage-ment. For more information about clocking schemes, PLL configuration, and sampling rates, see the Clocking and Sampling Rates section.

Case 1: PLL Is Bypassed

If the PLL is bypassed, the core clock is derived directly from the MCLK input. The rate of this clock must be set properly in Register R0 (clock control register, Address 0x4000) using the INFREQ[1:0] bits. When the PLL is bypassed, supported external clock rates are $256 \times f_S$, $512 \times f_S$, $768 \times f_S$, and $1024 \times f_S$, where f_S is the base sampling rate. The core clock of the chip is off until the core clock enable bit (COREN) is asserted. If a clock slower than $1024 \times f_S$ is directly input to the ADAU1461 (bypassing the PLL), the number of available SigmaDSP processing cycles is reduced, and the DSPSR bits in Register R57 (Address 0x40EB) should be adjusted accordingly.

Case 2: PLL Is Used

The core clock to the entire chip is off during the PLL lock acquisition period. The user can poll the lock bit to determine when the PLL has locked. After lock is acquired, the ADAU1461 can be started by asserting the core clock enable bit (COREN) in Register R0 (clock control register, Address 0x4000). This bit enables the core clock to all the internal blocks of the ADAU1461.

PLL Lock Acquisition

During the lock acquisition period, only Register R0 (Address 0x4000) and Register R1 (Address 0x4002) are accessible through the control port. Because all other registers require a valid master clock for reading and writing, do not attempt to access any other register. Any read or write is prohibited until the core clock enable bit (COREN) and the lock bit are both asserted.

To program the PLL during initialization or reconfiguration of the clock setting, the following procedure must be followed:

1. Power down the PLL.
2. Reset the PLL control register.
3. Start the PLL.
4. Poll the lock bit.
5. Assert the core clock enable bit after the PLL lock is acquired.

The PLL control register (Register R1, Address 0x4002) is a 48-bit register where all bits must be written with a single continuous write to the control port.
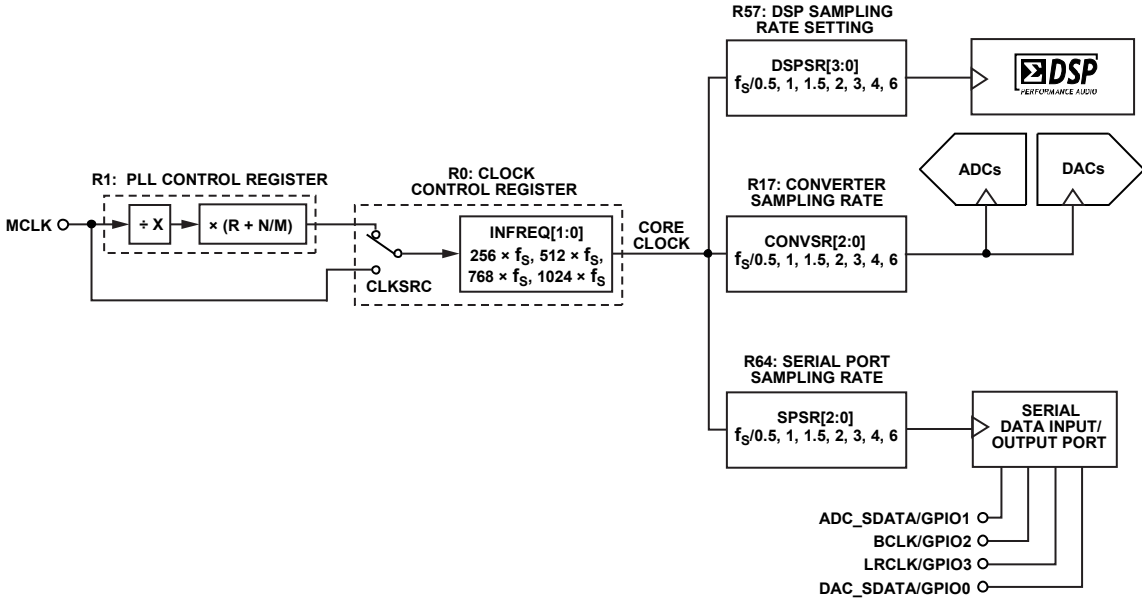
CLOCKING AND SAMPLING RATES

Figure 30. Clock Tree Diagram

CORE CLOCK

Clocks for the converters, the serial ports, and the DSP are derived from the core clock. The core clock can be derived directly from MCLK or it can be generated by the PLL. The CLKSRC bit (Bit 3 in Register R0, Address 0x4000) determines the clock source.

The INFREQ[1:0] bits should be set according to the expected input clock rate selected by CLKSRC; this value also determines the core clock rate and the base sampling frequency, f_S.

For example, if the input to CLKSRC = 49.152 MHz (from PLL), then

$INFREQ[1:0] = 1024 \times f_S$

$f_S = 49.152 \text{ MHz}/1024 = 48 \text{ kHz}$

The PLL output clock rate is always $1024 \times f_S$, and the clock control register automatically sets the INFREQ[1:0] bits to $1024 \times f_S$ when using the PLL. When using a direct clock, the INFREQ[1:0] frequency should be set according to the MCLK pin clock rate and the desired base sampling frequency.

To utilize the maximum amount of DSP instructions, the core clock should run at a rate of $1024 \times f_S$.

Table 11. Clock Control Register (Register R0, Address 0x4000)

Bits	Bit Name	Settings
3	CLKSRC	0: Direct from MCLK pin (default) 1: PLL clock
[2:1]	INFREQ[1:0]	00: $256 \times f_S$ (default) 01: $512 \times f_S$ 10: $768 \times f_S$ 11: $1024 \times f_S$
0	COREN	0: Core clock disabled (default) 1: Core clock enabled