NAU88L21

Ultra-Low Power Audio CODEC Ground-Referenced Headphone Amplifier

GENERAL DESCRIPTION

The NAU88L21 is an ultra-low power high performance audio codec that supports both analog and digital audio functions. It includes one I2S/PCM interface, one digital microphone interface, one digital mixer, two high quality DACs and ADC's, and one stereo class G headphone amplifier. The advanced on-chip signal processing engine that includes dynamic range compressor (DRC), programmable biquad filter, as well as an integrated frequency locked loop (FLL) to support various input clocks.

FEATURES

- 1 Digital I2S/PCM I/O port
- Two mono differential or one stereo differential analog microphone inputs, two single-ended microphone inputs or one stereo digital microphone input
- Cap-free Low noise Microphone bias with 7uVrms noise between 20Hz-20kHz, internal pull high resistor for microphone
- Class G Headphone Amplifier (28mW @ 32Ω, 1% THD+N)
- DAC: 105dB SNR, (A-weighted) @ 0dB gain, 1.8V and -88dB THD @ 20mW and RL= 32Ω, DAC playback to headphone output mode
- ADC: 102dB SNR (A-weighted) @ 0dB MIC gain, 1.8V, Fs = 48kHz and -91dB THD, 1.8V, MIC gain 0dB, OSR 256x
- Sampling rate from 8k to 192 kHz
- Dynamic Range Compressor (DRC)Programmable Biquad filter Integrated DSP with specific functions:Input automatic level control (ALC/AGC)/limiter
- Output dynamic-range-compressor/limiter
- Package: QFN-32
  Package is Halogen-free, RoHS-compliant and TSCA-compliant

APPLICATIONS

- Gaming controller
- Wireless Headset
- Smart Remote Controller

Block Diagram - QFN32

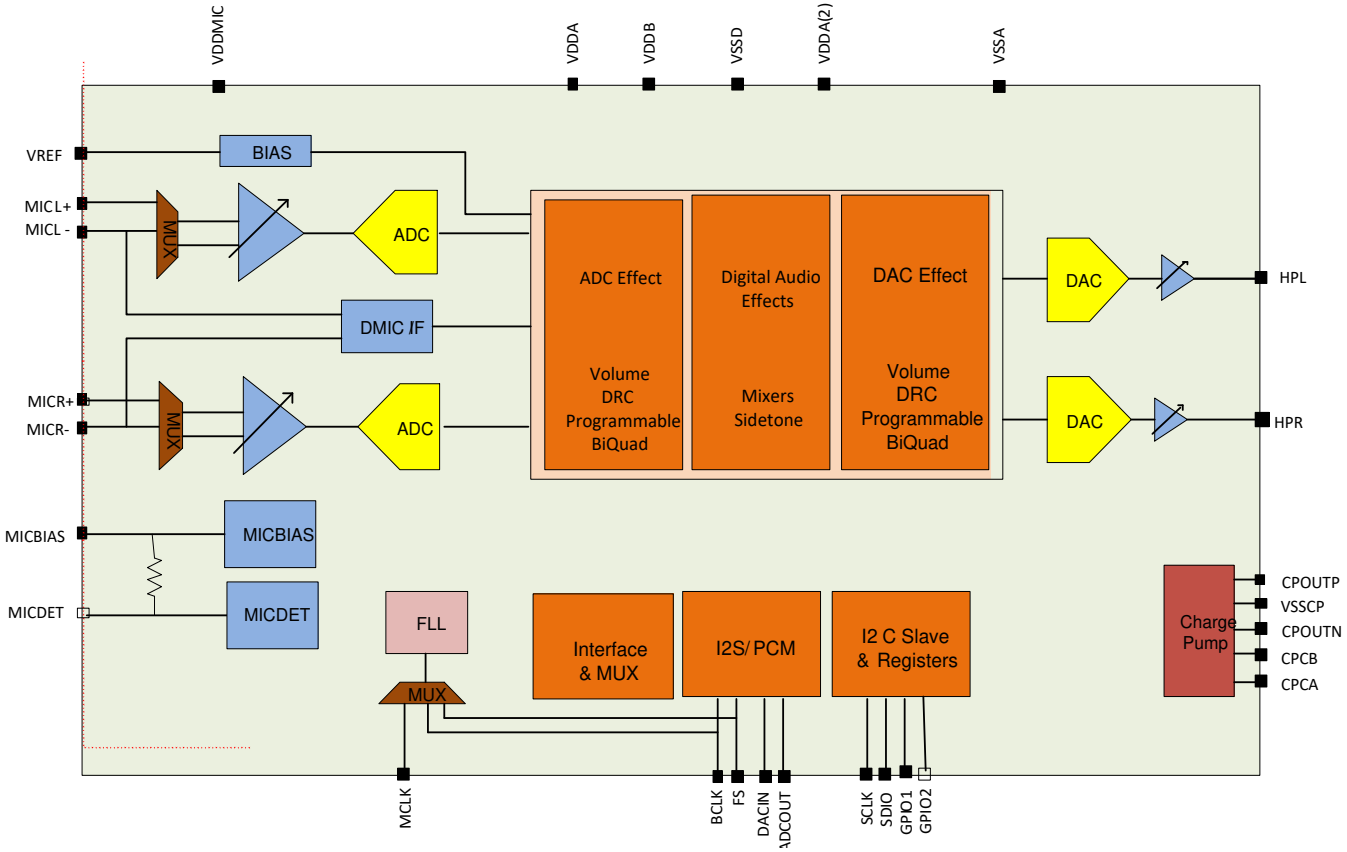
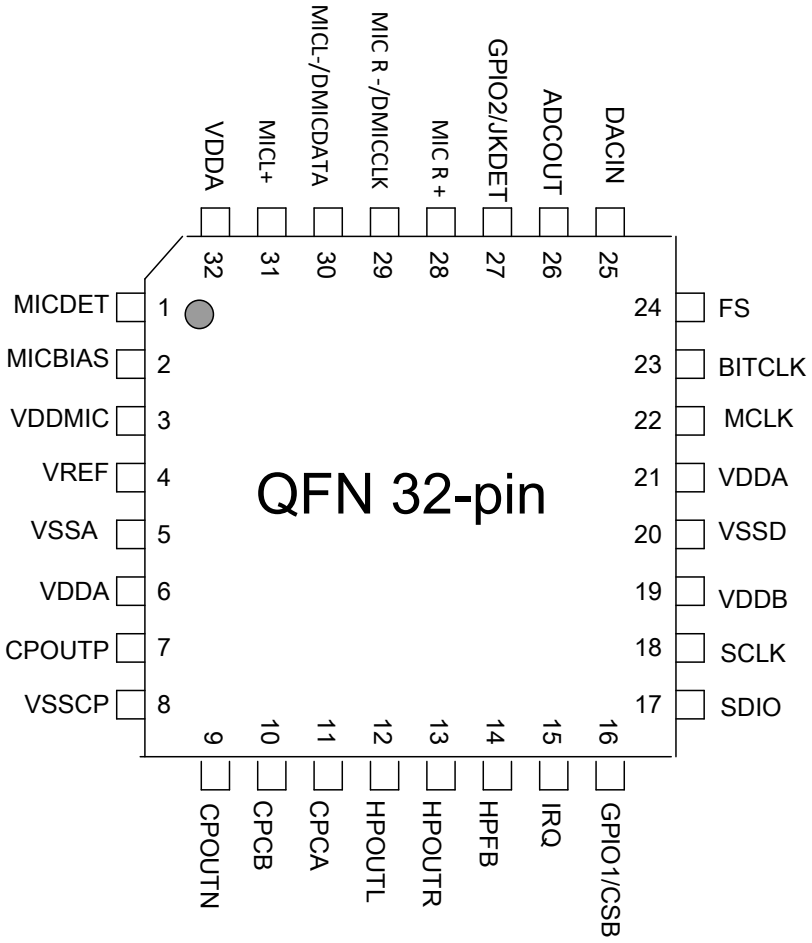


Table of Contents

Pin Diagram


QFN 32-pin
1 MICDET
2 MICBIAS
3 VDDMIC
4 VREF
5 VSSA
6 VDDA
7 CPOUTP
8 VSSCP
9 CPOUTN
10 CPCB
11 CPCA
12 HPOUTL
13 HPOUTR
14 HPFB
15 IRQ
16 GPIO1/CSB
17 SDIO
18 SCLK
19 VDDB
20 VSSD
21 VDDA
22 MCLK
23 BITCLK
24 FS
25 DACIN
26 ADCOUT
27 GPIO2/JKDET
28 MIC R +
29 MIC R -/DMICCLK
30 MICL-/DMICDATA
31 MICL+
32 VDDA

Pin Description

Pin #	Name	Type	Functionality
1	MICDET	Analog IO	Microphone/button detect, 2kOhm between Mic and Mic Bias
2	MICBIAS	Analog Output	Microphone Bias Output
3	VDDMIC	Supply	Microphone supply
4	VREF	Analog I/O	Internal DAC & ADC voltage reference decoupling I/O
5	VSSA	Ground	Analog Ground
6	VDDA	Supply	Analog Supply
7	CPOUTP	Analog I/O	Charge Pump positive voltage
8	VSSCP	Ground	Charge Pump Supply ground
9	CPOUTN	Analog I/O	Charge Pump negative voltage
10	CPCB	Analog I/O	Charge Pump switching capacitor node B
11	CPCA	Analog I/O	Charge Pump switching capacitor node A
12	JKTIP(HPL)	Analog Output	Jack Tip; Headphone left channel output
13	JKR1(HPR)	Analog Output	Jack Ring1; Headphone right channel output
14	HPFB	Ground	Headphone Ground
15	IRQ	Digital I/O	IRQ
16	GPIO1/CSB	Digital I/O	General Purpose IO/CSB
17	SDIO	Digital I/O	Serial Data for I2C
18	SCLK	Digital Input	Serial Data Clock for I2C
19	VDDB	Supply	Digital IO Supply
20	VSSD	Ground	Digital IO ground
21	VDDA	Supply	Analog supply
22	MCLK	Digital Input	CODEC Master clock input
23	BCLK	Digital I/O	Serial data bit clock input or output for I2S or PCM data
24	FS	Digital I/O	Frame Sync input or output for I2S or PCM data
25	DACIN	Digital Input	Serial Audio data input for I2S or PCM data
26	ADCOUT	Digital Output	Serial Audio data Output for I2S or PCM data
27	JKDET	Analog Input	Jack detect input
28	MICR+	Analog Input	PGA MICR+ Analog Input
29	MICR-/DMCLK	Analog/Digital Output	PGA MICR- Analog Input / Digital Microphone Clk output
30	MICL-/DMDATA	Analog Input/Digital Input	PGA MICL- Analog Input / Digital Microphone Data input
31	MICL+	Analog Input	PGA MICL+ Analog Input
32	VDDA	Supply	Analog Supply

Electrical Characteristics

Conditions: $V_{DD}A$ = $V_{DD}B$ = 1.8V; $V_{DD}MIC$= 3.6V.
R_L(Headphone) = 32 Ω, f = 1kHz, MCLK=12.288MHz, unless otherwise specified. Limits apply for T_A = 25°C

Symbol	Parameter	Conditions	Typical	Limit	Units
ISD	Shutdown Current	$V_{DD}A$	4	16	µA
		$V_{DD}B$	0.2	1	
		$V_{DD}MIC$	0.2	1	
I_{DD}	Headset Detection Standby Mode	MCLK off, Jack Insertion, IRQ enabled		10	µA
	Active Current Normal Playback Mode	fS = 48kHz, Stereo HP DAC On, HP On, POUT = 0mW. RL(HP) = 32Ω		5	mA
Headphone Amplifier					
P_O	Output Power	Stereo RL = 32Ω, DAC Input, CPVVDD = 1.8V, f=1020Hz, 22kHz BW, THD+N = 1% (QFN package)	28		mW
		Stereo RL = 16Ω, DAC Input, CPVVDD = 1.8V, f=1020Hz, 22kHz BW, THD+N = 1% (QFN Package)	33		mW
THD+N	Total Harmonic Distortion + Noise	RL = 32Ω, f=1020Hz, PO = 20mW	-88		dB
SNR	Signal to Noise Ratio	VOUT = 1VRMS, DAC Input, DAC_Gain = 0dB, HP_Gain = 0dB, Digital Zero Input, f=1020Hz, A-Weighted)	105		dB
		VOUT = 1 VRMS, DAC Input, DAC_Gain = 0dB, HP_Gain = 0dB, Digital Zero Input, f=1020Hz, A-Weighted, auto attenuate enabled	108		dB
PSRR	Power Supplys Rejection Ratio	fRIPPLE = 217Hz, VRIPPLE = 200mVP_P Input Referred, HP_GAIN = 0dB DAC Input, DAC_Gain = 0dB Ripple Applied to VDDA	90		dB
		Mono_Gain = 0dB Ripple Applied to VDDA	90		dB
		Stereo Single Ended Input Terminated, Stereo_Gain = 0dB Ripple Applied to VDDA	90		dB
X_{TALK}	Channel Crosstalk	Left Channel to Right Channel, -1dBFS, Gain = 0dB, f = 1020Hz	70		dB
	Interchannel Level Mismatch		+/- 0.1		dB
	Frequency Response	F = 20Hz ~ 20kHz	+0.1/-0.2		dB
	Pop up Noise			1	mVrms
e_{OS}	Output Noise	DAC_Gain = 0dB, HP_Gain = 0dB, fS=48kHz, OSRDAC = 128, A-Weighted	4.4		uV_{RMS}
	Out of Band Noise Level		-60dB		

Symbol	Parameter	Conditions	Typical	Limit	Units
V_{OS}	Output Offset Voltage	HP_Gain = 0dB, DAC_Gain= 0dB, DAC Input		±1	mV
	Power Consunption MP3 Mode	No Load, No Signal, Amp on fS = 48kHz, Stereo DAC On, Amp On, POUT = 0mW. RL = 32Ω	6		mW
	Fs Accuracy (44.1 / 48 kHz)		+/- 0.02%		
	Pop and Click Noise	Plug into or out of DAC to Headphone	1		mVrms
ADC					
THD+N	ADC Total Harmonic Distortion + Noise	MIC Input, MIC_GAIN = 0dB, VIN = 0.8Vrms, f=1020Hz, fs = 48KHz, Mono Differential Input	-91		dB
		MIC Input, MIC_GAIN = 30dB, Volume = 0dB, Vin=28.5mVrms, f=1020Hz, Digital Gain = 0dB, Mono Differential Input	-80		dB
SNR	Signal to Noise Ratio	Reference = VOUT(0dBFS), A-Weighted, MIC Input, MIC Gain = 0dB,fs = 48kHz, Mono Differential Input	102		dB
		Reference = VOUT(0dBFS), A-Weighted, MIC Input, MIC Gain = 6dB,fs = 48kHz, Mono Differential Input	101		dB
PSRR	Power Supply Rejection Ratio	VRIPPLE = 200mVPP applied to VDDA, fRIPPLE = 217Hz, Input Referred, MIC_GAIN = 0dB Differential Input	90		dB
CMRR	Common Mode Rejection Ratio	Differential Input 100mVrms, PGA gain = 20dB, frequency sweep from 20Hz to 20KHz	65		dB
FS_{ADC}	ADC Full Scale Input Level	$V_{DD}A$= 1.8V	1		V_{RMS}
	Minimum Input Impedance		10		kOhm
	Frequency Response	f = 20Hz ~ 20kHz	+0.1/-0.2		dB
	Pop up Noise	TBD	1		mVrms
	Power Consumption	No Signal, ADC on f_s = 44.1kHz	5		mW
MICBIAS					
V_{BIAS}	Output Voltage	Programmable 1.8V to 3.0V in 6 steps	2.5		V
I_{OUT}	Output Current			4	mA
e_{os}	Output Noise	Low noise mode, at 1kHz		47	nV/√Hz

Digital I/O

Parameter	Symbol	Comments/Conditions		Min	Max	Units
Input LOW level	V_{IL}	$V_{DD}B$ = 1.8V			0.33*$V_{DD}B$	V
		$V_{DD}B$ = 3.3V			0.37*$V_{DD}B$	
Input HIGH level	V_{IH}	$V_{DD}B$ = 1.8V		0.67*$V_{DD}B$		V
		$V_{DD}B$ = 3.3V		0.63*$V_{DD}B$		
Output HIGH level	V_{OH}	I_{Load}= 1mA	$V_{DD}B$=1.8V	0.9*$V_{DD}B$		V
			$V_{DD}B$ = 3.3V	0.95*$V_{DD}B$		

Parameter	Symbol	Comments/Conditions		Min	Max	Units
Output LOW level	V_{OL}	I_{Load}= 1mA	$V_{DD}B$ = 1.8V		0.1*$V_{DD}B$	V
			$V_{DD}B$=3.3V		0.05*$V_{DD}B$	

Recommended Operating Conditions

Condition	Symbol	Min	Typical	Max	Units
Digital I/O Supply Range	$V_{DD}B$	1.62	3.3	3.6	V
Analog Supply Range	$V_{DD}A$	1.62	1.8	1.98	V
Headphone Supply Range	$V_{DD}A$	1.62	1.8	1.98	V
Microphone Bias Supply Voltage	$V_{DD}MIC$	3.0	3.3	3.6	V
Temperature Range	T_A	-40		+85	°C

Absolute Maximum Ratings

Parameter	Min	Max	Units
Digital Supply Range	-0.3	2.2	V
Digital I/O Supply Range	-0.3	4.0	V
Analog Supply Range	-0.3	2.2	V
Headphone Supply Range	-0.3	2.2	V
Microphone Bias Supply Voltage	-0.3	4.0	V
Voltage Input Digital Range	DGND - 0.3	V_{DD} + 0.3	V
Voltage Input Analog Range	AGND - 0.3	V_{DD} + 0.3	V
Junction Temperature, T_J	-40	+150	°C
Storage Temperature	-65	+150	°C

CAUTION: Do not operate at or near the maximum ratings listed for extended periods. Exposure to such conditions may adversely influence product reliability and result in failures not covered by warranty.

1. General Description

NAU88L21 is an ultra-low power CODECs that has both analog and digital blocks operating at 1.8V. This CODEC includes DSP functions including DRCs (Dynamic Range Compression) and programmable biquad filters. Mic bias supply is upgraded to support voltages up to 3V.

1.1 Inputs

The NAU88L21 provides analog inputs to acquire and process audio signals from microphones with high fidelity and flexibility. There is a stereo input path that can be used to capture signals from single-ended or differential sources. The channel has a fully differential programmable gain amplifier (PGA). The outputs of the PGA connect to the ADC.

The NAU88L21 also has an input for one digital microphone. The NAU88L21 provides a DMCLK, the clock signal for the digital microphones.

The analog and the digital microphone inputs cannot be used simultaneously.

1.2 Outputs

NAU88L21 has one pair of ground-referenced Class G headphone outputs that are fed by two DACs. The headphone amplifier has a gain range of -9dB to 0dB.

The Class G headphone amplifier is powered by the charge pump output voltages CPOUTP and CPOUTN. When there is no loading the CPOUTP is equal to VDDA, and CPOUTN is equal to –VDDA.
This headphone output can also be used as a lineout.

1.3 ADC, DAC and Digital Signal Processing

The NAU88L21 has two independent high quality ADC's and DACs. These are high performance 24-bit sigma-delta converters, which are suitable for a very wide range of applications.

The ADCs and DACs have functions that individually support digital mixing and routing. The ADCs and DACs blocks also support advanced digital signal processing subsystems that enable a very wide range of programmable signal conditioning and signal optimizing functions. All digital processing is done with 24-bit precision to minimize processing artifacts and maximize the audio dynamic range supported by the NAU88L21.

The ADCs and DACs digital signal process can support two-point dynamic range compressors (DRCs), programmable biquad filters configurable for low pass filters, high pass filters, Notch filter, Bell, low shelf, and high shelf filters with various gain, Q, and frequency controls. Two-point DRCs can be programmed to limit the maximum output level and/or boost a low output level. The biquad filters can be configured as high pass filters intended for DC-blocking or low frequency noise reduction, such as reducing unwanted ambient noise or "wind noise" on a microphone inputs.

1.4 Digital Interfaces

Command and control of the device is accomplished by using the I2C interface.

The digital audio I/O data streams transfer separately from command and control using either I2S or PCM audio data protocols

These simple but highly flexible interface protocols are compatible with most commonly used serial data protocols, host drivers, and industry standard I2S and PCM devices.

2. Power Supply

This NAU88L21 has been designed to operate reliably using a wide range of power supply conditions and power-on/power-off sequences. Because of this, there are no special requirements for the sequence or rate at which the various power supply pins change. Any supply can rise or fall at any time without harming the device. However, pops and clicks may result from some sequences.

2.1 Power on and off reset

The NAU88L21 includes a power on reset circuit on chip. The circuit resets the internal logic control at VDDA supply power up and this reset function is automatically generated internally when power supplies are too low for reliable operation. The reset threshold is approximately 0.55Vdc and 1.0Vdc for VDDA. It should be noted that these values are much lower than the required voltage for normal operation of the chip.

The reset is held on while the power levels for VDDA are below their respective thresholds. Once the power levels rise above their thresholds, the reset is released. Once the reset is released, the registers are ready to be written to. It is also important to note that all the registers should be kept in their reset state for at least 6µs.

An additional internal RC filter based circuit is added which helps the circuit respond for fast ramp rates (~10µs) and generate the desired reset period width (~10µs at typical corner). This filter is also used to eliminate supply glitches which can generate a false reset condition, typically 50ns.
For reliable operation, it is recommended to write any value to register upon power up. This will reset all registers to the known default state.

Note that when VDDA are below the power on reset threshold, then the digital IO pins will go into a tri-state condition.

2.2 Power up and Start Sequence

The power up sequence to bring up the analog blocks smoothly is illustrated below and involves three different time segments (T1 – T4). The power supply ramp rate depends on a number of factors such as the power source drive strength, board parasitics and the decopling capacitor size on the supply line. Typically, a power supply ramp time can be as fast as 5mS or as slow as 200mS.

During time T1, the power supply ramps-up. The internal PORB reset is generated when V_{DDA} is lower than 1.1V for reliable maintenance of internal logic circuits. While PORB signal is low, it clears internal digital flops. Most of the flops will be cleared to ‘0’ while some flops can be set to ‘1’ during the PORB pulse depending on the required default state of the register.

After time T1, wait another time 1mS so that the power supply is stable before writing to the registers. During time T2, the chip is in stand-by mode and all registers are in a default state. In stand-by, the chip only consumes leakage current and all analog blocks are turned-off. At time T3, the user can start to write data into the registers via the I2C serial bus to setup the chip for their application.

When I2C finished loading register in T3 period, waiting for T4, 1ms period, I2S clocks could input to device.

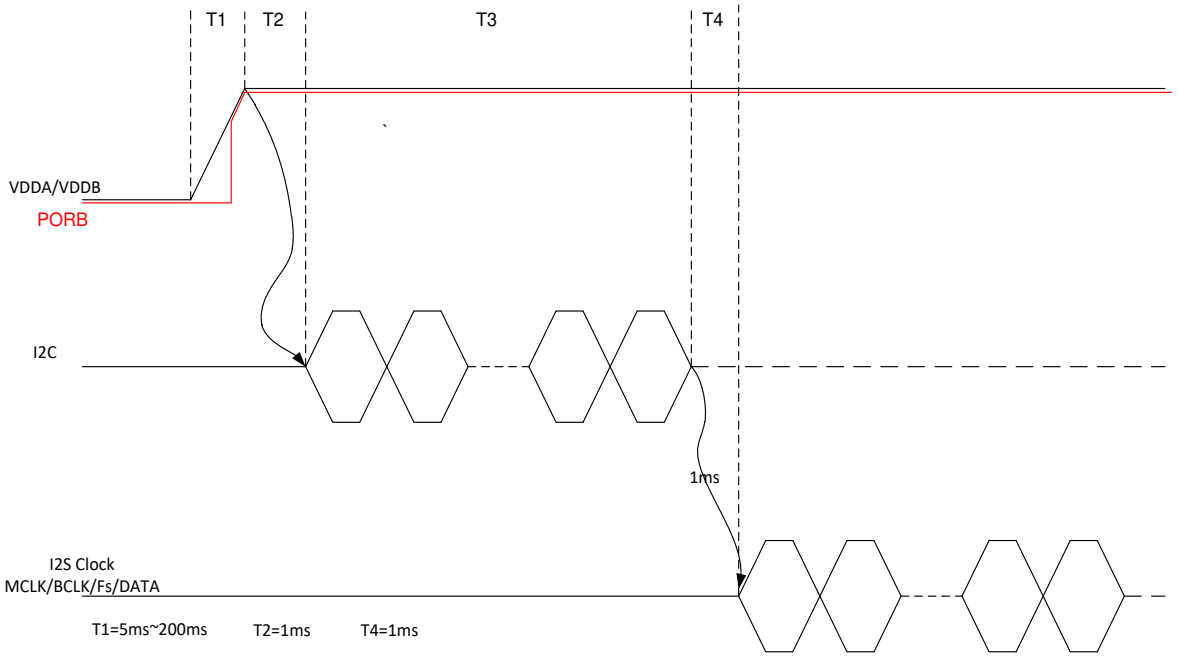


Figure 1: Power up sequence

3. Input Path Detailed Descriptions

NAU88L21 has two low noise, high common mode rejection ratio analog microphone differential input. The microphone inputs MICL+/- & MICR+/- which are followed by -1dB to 36dB PGA gain stages that have a fixed 12kOhm input impedance.

Inputs are maintained at a DC bias of approximately ½ of the VDDA supply voltage. Connections to these inputs should be AC-coupled by means of external DC blocking capacitors suitable for the device application.

The differential microphone input structure is essential in noisy digital systems where amplification of low-amplitude analog signals is necessary such as in portable digital media devices and cell phones. Differential inputs are also very useful to reduce ground noise in systems in which there are ground voltage differences between different chips and components. When properly implemented, the differential input architecture offers an improved power-supply rejection ratio (PSRR) and higher ground noise immunity.

3.1 Analog Microphone Inputs

The analog microphone inputs are routed to the FEPGA (Front End Programmable Gain Amplifier). The input stage can be configured in different modes. The FEPGA gain can be varied from -1dB to 36dB in 1dB steps. The gain stage has a fixed 12kOhm input impedance and can be individually enabled or disabled by using register.

As shown in Figure 1,
For left channel input path
SL1, it is controlled by 0x76[11]DISCHRG and 0x77[14]ACDC_CTRL[0],
SL2, it is controlled by 0x76[11]DISCHRG and 0x77[15]ACDC_CTRL[1],
SL3 and SL4, they are controlled by 0x77[5]FEPGA_MODEL[1],
SL5 and SL6, they are controlled by 0x77[7]FEPGA_MODEL[3],
SL7, it is controlled by 0x6B[3],
SL8, it is controlled by 0x6B[5].

For right channel input path
SR1, it is controlled by 0x76[11]DISCHRG and 0x77[14]ACDC_CTRL[0],
SR2, it is controlled by 0x76[11]DISCHRG and 0x77[15]ACDC_CTRL[1],
SR3 and SR4, they are controlled by 0x77[1]FEPGA_MODER[1],
SR5 and SR6, they are controlled by 0x77[3]FEPGA_MODER[3],
SR7, it is controlled by 0x6B[2],
SR8, it is controlled by 0x6B[4].

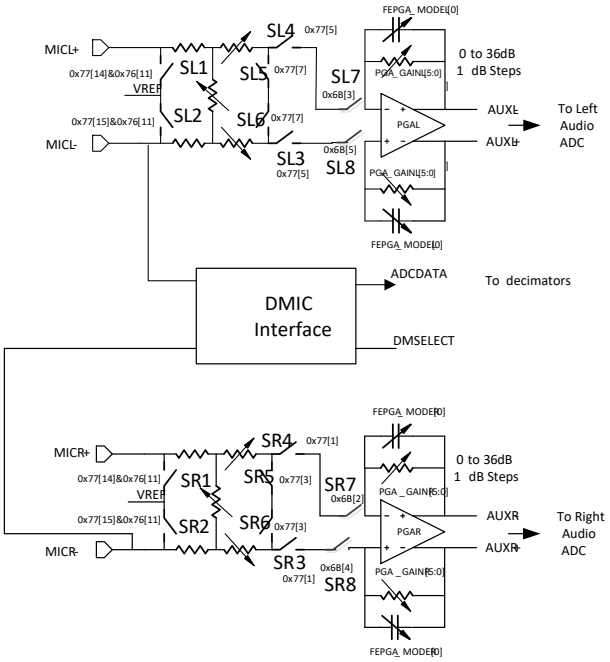


Figure 2: Microphone Input Block Diagram with Registers

3.2 Digital Microphone Input

The MICL- and MICR- pins can be used for the digital microphone input. MICR- is the clock for the digital microphones and the MICL- is the data in.

3.3 VREF

The NAU88L21 includes a mid-supply reference circuit that produces a voltage close to VDDA/2. This “VREF” pin should be decoupled to VSS through an external bypass capacitor. Because VREF is used as a reference voltage inside the NAU88L21, a large capacitance is required to achieve good power supply rejection at low frequency. Typically, a value of 4.7µF should be used. This larger capacitance may introduce longer rise time of VREF and delay the line output signal. However, a pre-charge circuit can be supported to help reduce the rise time. Due to the high

impedance of the VREF pin, it is important to use a low leakage capacitor. A pre-charge circuit has been implemented to reduce the VREF rise time. Once charged, this can be disabled using to save power or prevent rapid changes in level due to fluctuations in VDDA. The below Table 1 shows the VREF tie-off resister selection.

VMIDSEL	VREF Resistor Selection	VREF Impedance
00	Open, no resistor selected	Open, no impedance installed
01	50kOhm	25kOhm
10	250kOhm	125kOhm
11	5kOhm	2.5kOhm

Table 1: VREF Impedance Selection

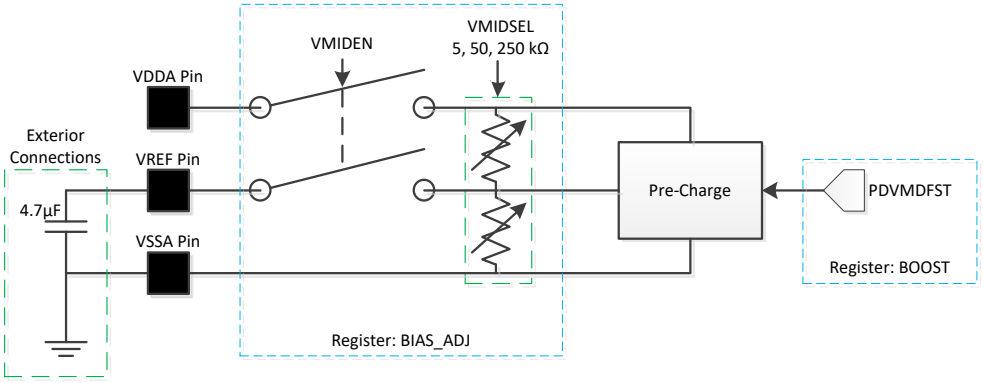


Figure 3: V_{REF} Circuitry

3.4 MIC Bias

The NAU88L21 provides one MIC bias pin, which can be used to power various microphones. The output level of MIC Bias can be set between VDDA and 1.53 X VDDA using register settings.

It is recommended that the microphones do not draw more than 4mA from the MICBIAS pin. There are options for connecting internal 2 Kohm resistor to the microphone and for low noise or low power mode. If MICBIAS is used in low power mode, typically 100nF or 200nF capacitor can be used along with MIC Bias level at VDDA. In the low noise mode, external 1uF or 4.7uF capacitor can be omitted by register settings when MIC Bias is used to power analog microphones.

3.5 MIC detect

The MIC detect block can detect whether a microphone is connected between the MICBIAS output and the MICDET pin. Either the internal 2kOhm resistor or an external 2kOhm resistor can be used to connect the microphone to the MICDET pin and MICBIAS. See Figure 4, where the internal hookup of the MICDET and MICBIAS blocks is shown.

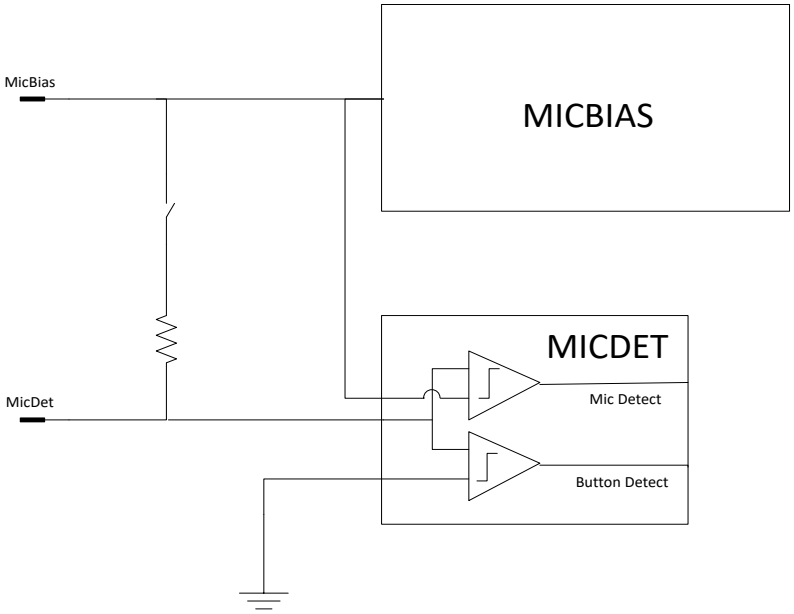


Figure 4. Mic Detect and MICBIAS blocks

Application note: Adding a simple RC on the MICDET pin can help reduce noise coupling. These may be board level related, or component related effects.

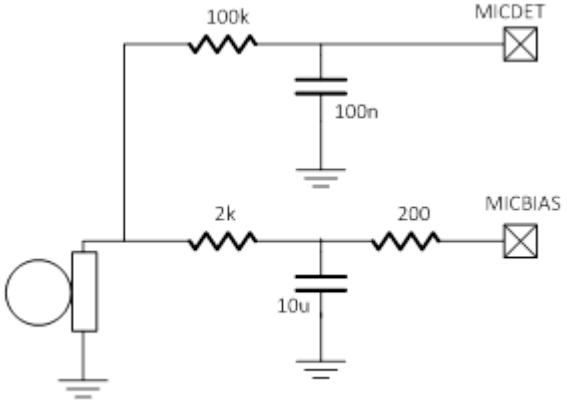


Figure 5. Reducing noise coupling effects

If the optional external 2KOhm resistor is used, then the internal 2K Ohm resistor (Between MICBIAS and MICDET) should be disabled.

3.5.1 Key Release

This feature detects the edge case where the key press interrupt is not followed by a release interrupt until later on in the sequence and clears the x11 register to prepare for further interrupts.

Figure 6. Key Release Flowchart

Note:

1. Mic detect current threshold ~ 12.5uA, and voltage threshold = MICBIAS - 26mV. Either condition trigger mic detectons
2. Button (key) detection current throushold larger than 800uA and voltage threshold is GND + 85mA. Either condition trigger button detection.

4. ADC Digital Block

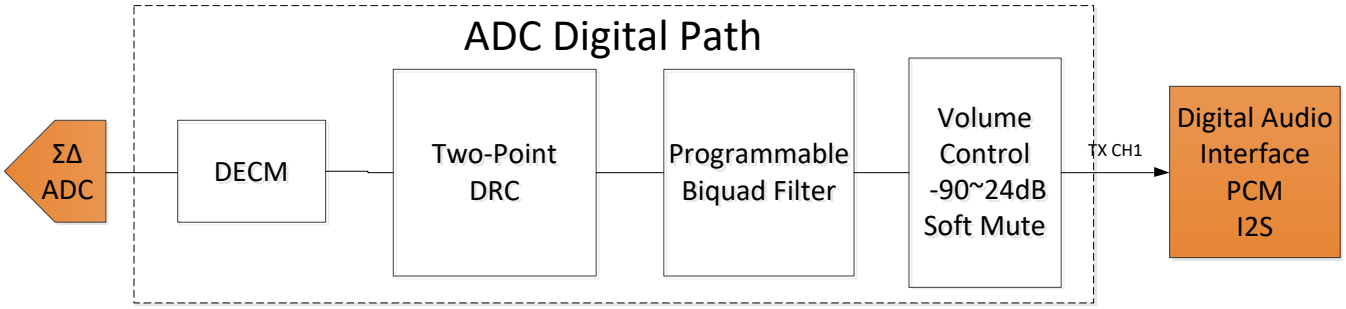


Figure 7: ADC Digital Path

The ADC digital block takes the output of the 24-bit Analog-to-Digital converter and performs signal processing aimed at producing a high quality audio sample stream to the audio path digital interface. The Figure 7 shows the various steps associated with the ADC digital path.

Oversampling is used to improve noise and distortion performance; however this does not affect the final audio sample rate. The oversampling rate configured between 32X and 256X using register settings.
The polarity of either ADC output signal can be changed independently on either ADC logic output as a feature sometimes useful in management of the audio phase. This feature can help minimize any audio processing that may be otherwise required as the data is passed to other stages in the system.

The full-scale input level is proportional to VDDA. For example, with a 1.8V supply voltage, the full-scale level is 1.0VRMS.

4.1 ADC Dynamic Range Compressors (DRC)

The ADC's in the digital signal path each support a two-point dynamic range compressor (DRC) for advanced signal processing. Each DRC can be programmed to limit the maximum output level and/or boost a low output level signal. The DRC's function consists of level estimation and static curve control.

4.1.1 Level Estimation

The NAU88L21 uses Peak level estimation that depends on the attack and decay time settings, which can be programmable by register settings as shown in the Table 2.

BITS	DRC_PK_COEF1_ADC	DRC_PK_COEF2_ADC
0000	Ts	63*Ts
0001	3*Ts	127*Ts
0010	7*Ts	255*Ts
0011	15*Ts	511*Ts
0100	31*Ts	1023*Ts
0101	63*Ts	2047*Ts
0110	127*Ts	4095*Ts
0111	255*Ts	8191*Ts

Table 2: ADC Level Estimation - Attack and Decay Time Register Settings

Please note that Ts is the sampling time given by 1/(Sampling Frequency)

4.1.2 Static Curve

The DRC static curve supports up to five programmable sections as shown in the Figure 8.

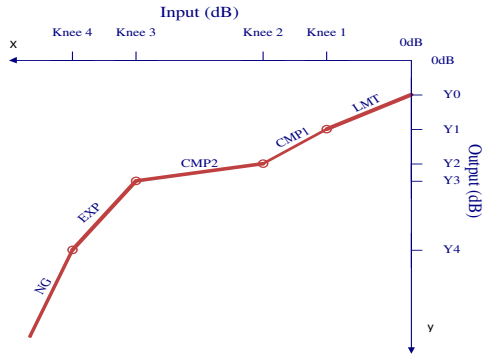


Figure 8: DRC Static Characteristic

Each section on the characteristic (labeled NG, EXP, CMP2, CMP1, and LMT) can be controlled by setting the slope and knee point values, in their respective registers. The table below provides the corresponding register locations.

Static Curve Section	Slope	Knee Point
LMT	0, 1/2, 1/4, 1/8, 1/16, 1/32, 1/64, 1	
CMP1	0, 1/2, 1/4, 1/8, 1/16, 1	0 to -31dB with -1dB step
CMP2	0, 1/2, 1/4, 1/8, 1/16, 1	0 to -63dB with -1dB step
EXP	1, 2, 4	-18 to -81dB with -1dB step
NG	1, 2, 4, 8	-35 to -98dB with -1dB step

Table 3: ADC DRC Static Curve control registers

The output Y values can be determined based on the slopes and knee points selected. Y1 is always equal to Knee 1, as an initial and default condition.

Y1 = Knee 1
Y0 = Y1 - (Knee 1) * (LMT Slope)
Y2 = (Knee 2 - Knee 1) * (CMP1 Slope) + Y1
Y3 = (Knee 3 - Knee 2) * (CMP2 Slope) + Y2
Y4 = (Knee 4 - Knee 3) * (EXP Slope) + Y3

The attack time and decay time is programmable as shown in the Table 4. And the smooth knee filter can be also enabled by register setting.

BITS	DRC_ATK_ADC_CH##	DRC_DCY_ADC_CH##
0000	Ts	63*Ts
0001	3*Ts	127*Ts
0010	7*Ts	255*Ts
0011	15*Ts	511*Ts
0100	31*Ts	1023*Ts
0101	63*Ts	2047*Ts
0110	127*Ts	4905*Ts
0111	255*Ts	8191*Ts
1000	511*Ts	16383*Ts
1001	1023*Ts	32757*Ts
1010	2047*Ts	65535*Ts
1011	4095*Ts	
1100	8191*Ts	

Table 4: ADC Attack and Decay Time Register Settings

4.1.3 Limitation

Due to DC offsets from the ADC block, DRC performance may have limitation. Especially when DRC is designed with 3 or 4 knee points, output level variation caused by DC offsets should be taken into consideration.
For DRC design with 2 knee points by setting knee2, knee3, knee4 together, DRC output curves of left and right channels typically share the same track as shown below:

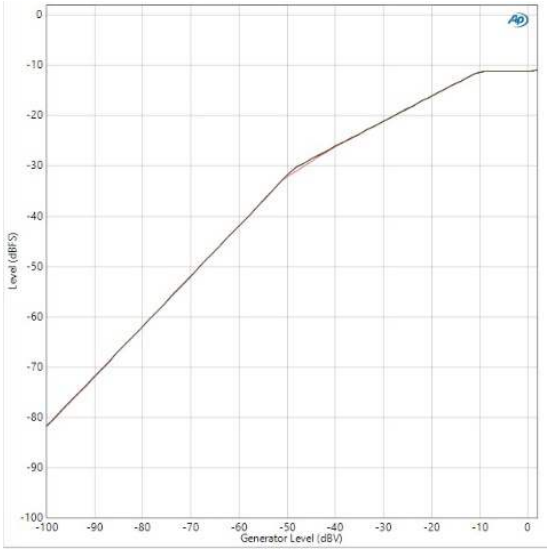


Figure 9: DRC Output Curve with 2 Knee Points

For DRC with 3 or 4 knee points, systematic DC offsets on both channels will become unnegligible. Especially for input level at lower than -50dB, DRC output curves of left and right channels will split into two traces. At -60dB input level, the output level variation may be up to 15dB as shown below:

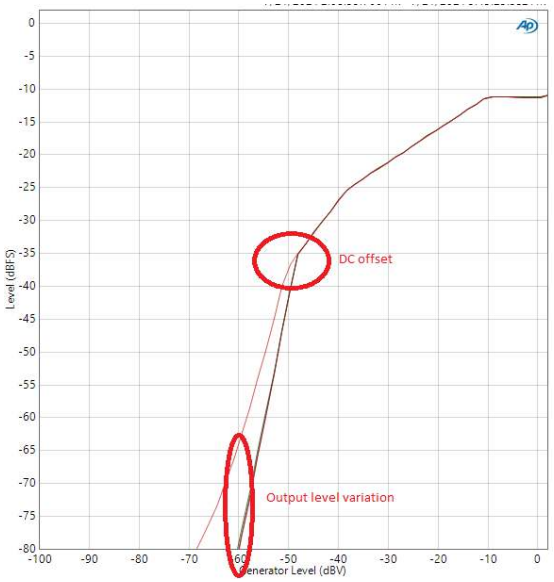


Figure 10: DRC Output Curve with 3 Knee Points

4.2 ADC Digital Volume Control

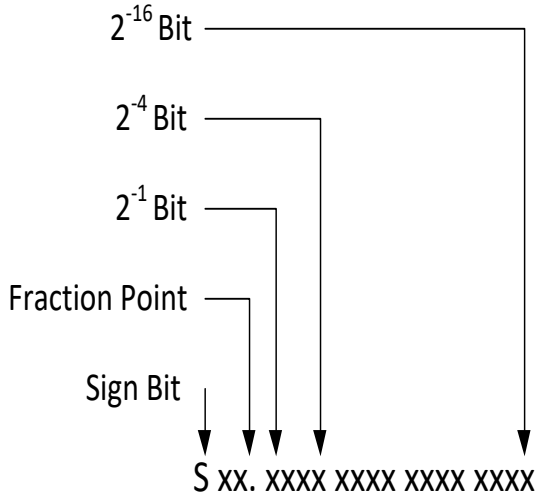
The digital volume control feature allows adjustment of the audio volume coming from ADC using a two-stage volume control. This allows the gain to be adjusted from -103dB to +24dB. Also included is a mute value that will reduce the output signal of the ADCs to zero.

4.3 ADC Programmable Biquad Filter

The NAU88L21 has 4 dedicated digital biquad filters. Two for the ADC path, and two for the DAC path. The biquad filter is a second-order recursive linear filter with two poles and two zeros. Its transfer function is the Z-domain consists of two quadratic functions:

$$H(z) = \frac{B_0 + B_1 Z^{-1} + B_2 Z^{-2}}{1 + A_1 Z^{-1} + A_2 Z^{-2}}$$

The coefficients A_1, A_2, B_0, B_1, B_2 are represented in the 3.16 format described below



Each Biquad Coefficient has 19 bits in Sxx.16 format where

- S is the sign bit (1 bit),
- xx are integers (2bits)
- 16 fractional bits (16 bits)

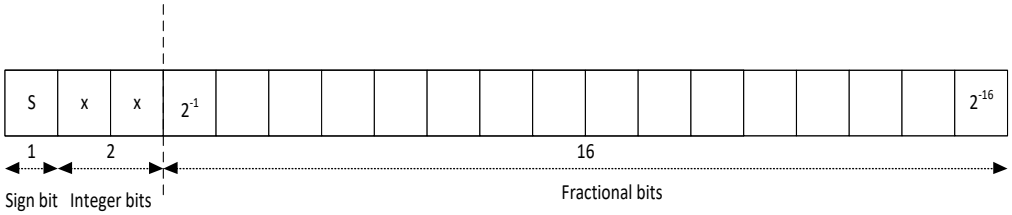


Figure 11: Number format description for biquad filters coefficients

4.4 Companding

Companding is used in digital communication systems to optimize signal-to-noise ratios with reduced data bit rates using non-linear algorithms. The NAU88L21 supports the two main telecommunications companding standards on both transmit and receive sides: A-law and μ-law. The A-law algorithm is primarily used in European communication systems and the μ-law algorithm is primarily used by North America, Japan, and Australia.

4.5 Additional ADC Application Notes

The ADC clock polarity can be inverted if necessary by register setting. It is recommend to match ADC oversampling rate with ADC clock rate as shown in the Table 5.

ADC_RATE	CLK_ADC_SRC
00(OSR=32)	11(CODEC 1/8)
01(OSR=64)	10(CODEC1/4)
10(OSR=128)	01(CODEC 1/2)
11(OSR=256)	00(CODEC CLK)

Table 5: ADC_RATE and CLK_ADC_SRC Pairs

5. DAC Digital Block

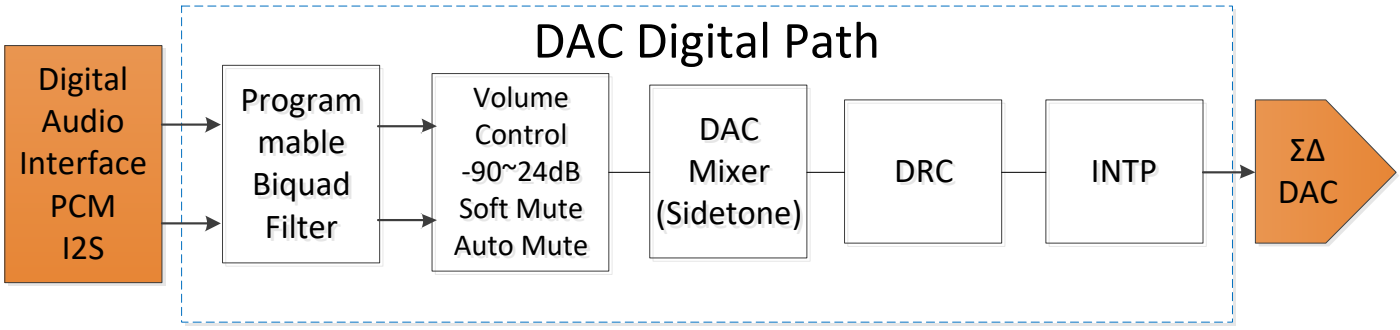


Figure 12: DAC Digital Path

The DAC digital block uses 24-bit signal processing to generate analog audio with a 16-bit digital sample stream input. This block consists of a sigma-delta modulator, digital decimator/filter, programmable biquad filter, and a DRC. The full-scale output level is proportional to VDDA. For example, with a 1.8V supply voltage, the full-scale level is 1.0 VRMS. The oversampling rate of the DAC can be changed from 32x to 256x for improved audio performance at higher power consumption. The DAC output signal polarity can be changed using register setting. This can help minimize any audio processing that may be required as the data is passed from other stages of the system.

5.1 DAC Dynamic Range Control (DRC)

The DAC DRC functions in the same way as the ADC DRC explained in Section 4.1. However, different control registers are used.

5.1.1 Level Estimation

The Table 6 shows the attack and decay times for the peak level estimation. And, the time constant Ts is the the sampling time given by 1/(Sampling Frequency).

BITS	DRC_PK_COEF1_ADC	DRC_PK_COEF2_ADC
0000	Ts	63*Ts
0001	3*Ts	127*Ts
0010	7*Ts	255*Ts
0011	15*Ts	511*Ts
0100	31*Ts	1023*Ts
0101	63*Ts	2047*Ts
0110	127*Ts	4095*Ts
0111	255*Ts	8191*Ts

Table 6: DAC Level Estimation Attack and Decay Time Register Settings

5.1.2 Static Curve

The DRC static curve supports five programmable sections, and slope and knee points can be configured as shown in the Table 7.

Static Curve Section	Slope	Knee Point
LMT	0, 1/2, 1/4, 1/8, 1/16, 1/32, 1/64, 1	
CMP1	0, 1/2, 1/4, 1/8, 1/16, 1	0 to -31dB with -1dB step
CMP2	0, 1/2, 1/4, 1/8, 1/16, 1	0 to -63dB with -1dB step
EXP	1, 2, 4, 8	-18 to -81dB with -1dB step
NG	1, 2, 4, 8	-35 to -98dB with -1dB step

Table 7: DAC DRC Static Curve Control Registers

The Table 8 shows the attack and decay time for DRC. And, it needs to be carefully used combination with cross talk function because DRC is the last blocks in the path after mixer. Small cross-talk signal might be filtered out by DRC. The smooth knee function can be also enabled by register setting.

BITS	DRC_ATK_DAC	DRC_DCY_DAC
0000	Ts	63*Ts
0001	3*Ts	127*Ts
0010	7*Ts	255*Ts
0011	15*Ts	511*Ts
0100	31*Ts	1023*Ts
0101	63*Ts	2047*Ts
0110	127*Ts	4095*Ts
0111	255*Ts	8191*Ts
1000	511*Ts	16383*Ts
1001	1023*Ts	32757*Ts
1010	2047*Ts	65535*Ts
1011	4095*Ts	
1100	8191*Ts	

Table 8: DAC Static Curve Attack and Delay Time Register Settings

5.2 DAC Digital Volume Control, Mute and Channel selection

DACL and DACR both have separate digital volume controls that allow the user to adjust the gain from -103dB to +24dB in 0.5dB steps as well as mutes. Left and Right channels can be adjusted separately and control is accessed through register settings.

5.3 DAC Soft Mute

The soft mute function ramps the DAC digital volume down to zero when enabled. When disabled, the volume increases to the register specified volume level for each channel. This feature provides a tool that is useful for using the DAC without introducing pop and click sounds.

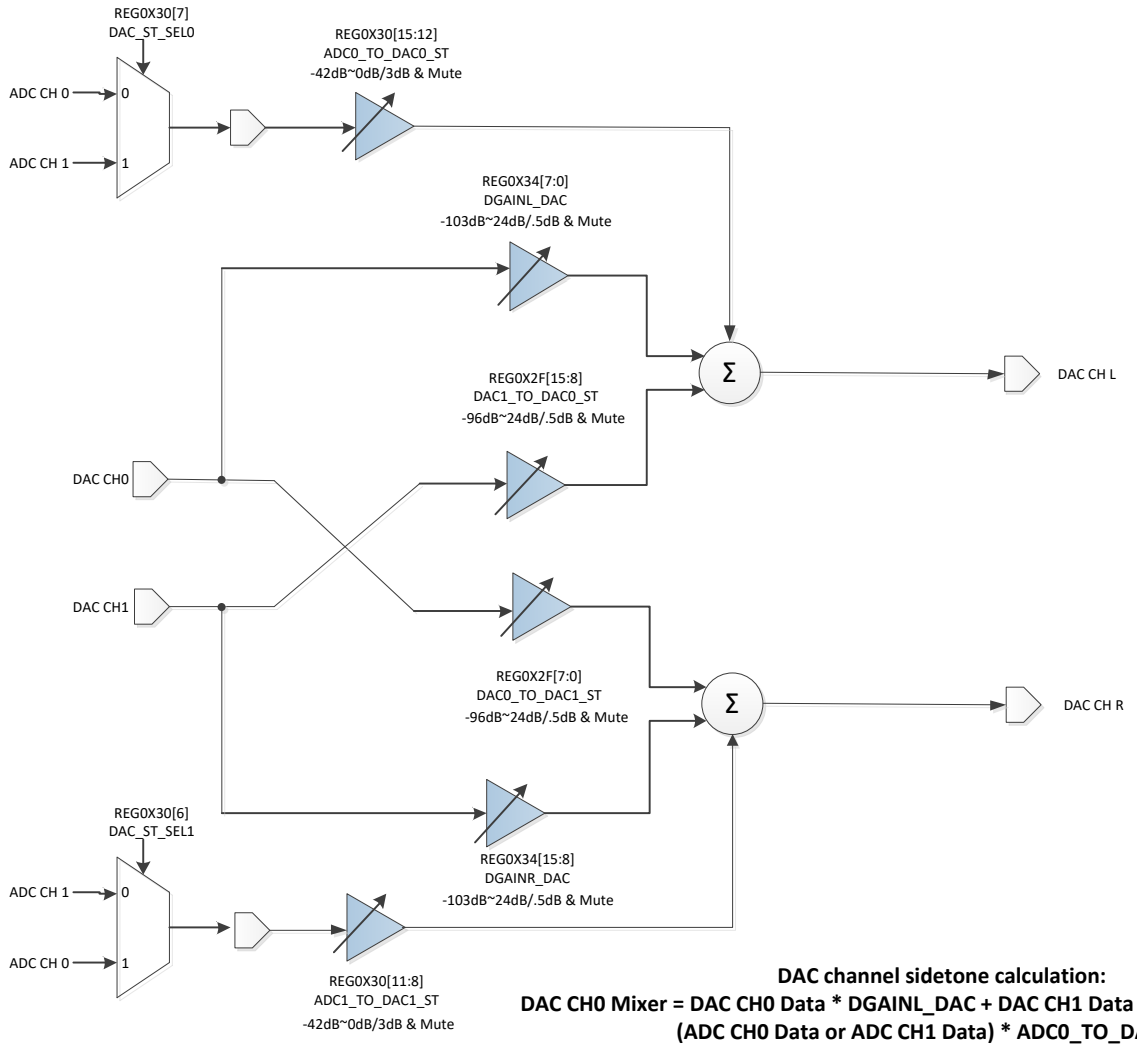
5.4 DAC Auto Attenuate

Auto-attenuate can greatly increase the perceived SNR during playback of silence. The last analog output stage is attenuated such that the noise contribution of the preceding stages is eliminated. The use of auto-attenuate by attenuating the analog output on a DAC path when the digital input represents a zero signal needs to be done gradually in order to avoid audible pops due to sudden offset changes. It is desirable to slowly ramp down the gain of the analog output stage to the maximum attenuation level. This function will be referred to as auto-attenuate. The auto-attenuate feature is used to increase the Signal to Noise Ratio. In addition, the auto attenuate logic can be used to attenuate the analog output manually,saving some software routines and allowing pop-less ramp up and down of the analog outputs with little register writes.

The auto-attenuate function can be enabled manually or automatically. In the automatic mode, if both the left and right channel receive 1024 consecutive samples of „0", then it will read and store the value of the headset driver volume control into internal temporary registers and then attenuate the headset driver output by 1dB for every 128 samples, until -54dB is reached (54 steps maximum). If , at any time, the I2S DACIN signal receives non-zero signal samples, the headset output driver gain is increased by 1dB per step and in 1, 16, 32 or 128 samples per step (programmable by register) until the gain will be stepped up until the original gain setting is reached. In the manual mode, once enabled, it will immediately start saving the volume control into temporary registers and attenuate signals by 1dB for every 128 samples untile -54dB is reached. If the manual attenuate is disabled, the gain will be fully recovered by 1dB step in 1, 16, 32, or 128 samples per step.

5.5 DAC Path Digital Mixer with Side tone

The NAU88L21 implements a channel based digital mixer architecture. Each DAC outputs can be selected between the different inputs. The ADC input channels, I2S channels are capable of being mixed into either output of the DAC. The figure below shows a block diagram of how the mixer works along with the related registers.



DAC channel sidetone calculation:

DAC CH0 Mixer = DAC CH0 Data * DGAINL_DAC + DAC CH1 Data * DAC1_TO_DAC0_ST + (ADC CH0 Data or ADC CH1 Data) * ADC0_TO_DAC0_ST

DAC CH1 Mixer = DAC CH1 Data * DGAINR_DAC + DAC CH0 Data * DAC0_TO_DAC1_ST + (ADC CH1 Data or ADC CH0 Data) * ADC1_TO_DAC1_ST

Figure 13: DAC Path Digital Mixer with Side tone.

5.6 Companding

Companding is used in digital communication systems to optimize signal-to-noise ratios with reduced data bit rates using non-linear algorithms. The NAU88L21 supports the two main telecommunications companding standards on both transmit and receive sides: A-law and μ-law. The A-law algorithm is primarily used in European communication systems and the μ-law algorithm is primarily used by North America, Japan, and Australia.

Companding converts 14 bits (μ-law) or 13 bits (A-law) to 8 bits using non-linear quantization resulting in 1 sign bit, 3 exponent bits and 4 mantissa bits. When the companding mode is enabled, 8 bit word operation must be enabled.

Sections 5.6.1 and 5.6.2 contain the compression equations set by the ITU-T G.711 standard and implemented in the NAU88L21.

5.6.1 μ-law

$$F(x) = \frac{\ln(1+\mu\times|x|)}{\ln(1+\mu)}, \quad -1 < x < 1$$

$$\mu = 255$$

5.6.2 A-law

$$F(x) = \frac{A\times|x|}{(1+\ln(A))}, \quad 0 < x < \frac{1}{A}$$

$$F(x) = \frac{(1+\ln(A\times|x|)}{(1+\ln(A))}, \quad \frac{1}{A} \le x \le 1$$

$$A = 87.6$$

6. Clocking and Sample Rates

The internal clocks for the NAU88L21 are derived from a common internal clock source. This master system clock can set directly by the MCLK pin input or it can be generated from a Frequency Locked Loop (FLL) using the MCLK_PIN, BCLK or FS as a reference. While most of the common audio sample rates can be derived directly from typical MCLK frequencies, the FLL provides additional flexibility for a wide range of MCLK inputs or as a free running clock in the absence of an external reference.

The figures below is a block diagram illustrating how the various register settings can be used to adjust/select the MCLK, BCLK, FS, and ADC_CLK clock frequency.

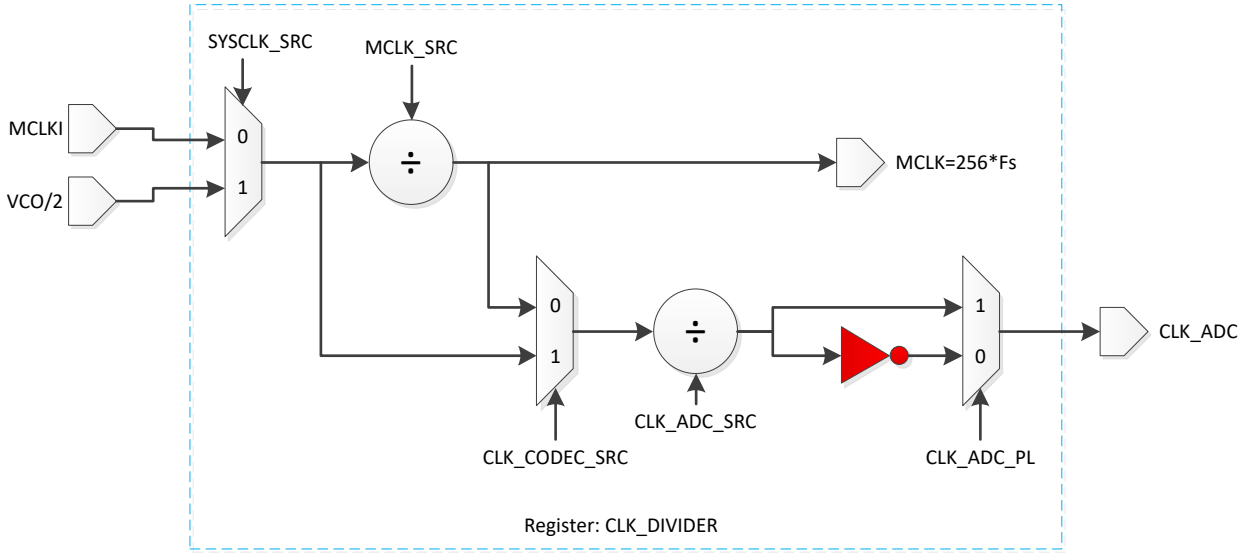


Figure 14: MCLK and ADC_CLK Frequency Selection

Bits	MCLK_SRC
0000	Divide by 1
0001	Invert
0010	Divide by 2
0011	Divide by 4
0100	Divide by 8
0101	Divide by 16
0110	Divide by 32
0111	Divide by 3
1001	Invert
1010	Divide by 6
1011	Divide by 12
1100	Divide by 24
1101	Divide by 48
1110	Divide by 96
1111	Divide by 5

Table 9: Register Settings

Bits	CLK_ADC_SRC
00	Divide by 1
01	Divide by 2
10	Divide by 4
11	Divide by 8

Table 10: Register Settings

The internal clock frequency MCLK must be running at 256*Fs (Fs = sample rate in Hz) in order to achieve the best performance. The internal clock frequency MCLK can also run at 400*Fs, which may give a slightly lower performance. For example, when targeting 48 kHz sample rate audio, the MCLK must be set to 256*48k = 12.288MHz, 400*48k = 19.2MHz. When the input clock MCLKI is higher than this speed, register **CLK_DIVIDER.MCLK_SRC_REG0X03[3:0]** provides a flexible divider selection to meet this requirement. The FLL can also be used to generate an MCLK that meets this requirement.
The OSR (over sampling rate) is defined as CLK_ADC frequency divided by the audio sample rate.

$$OSR = \frac{CLK\_ADC}{Fs}$$

Available over-sampling rates are 32, 64, 128 or 256 as set in the **ADC_RATE.ADC_RATE REG0X2B[1:0]** register. CLK_ADC frequency is set by **CLK_DIVIDER.CLK_CODEC_SRC REG0X03[13]** and **CLK_DIVIDER.CLK_ADC_SRC REG0X03[7:6]** registers.
It should be noted that the OSR and Fs must be selected so that the max frequency of CLK_ADC is less than or equal to 6.144MHz. When CLK_ADC is determined, **ADC_RATE.ADC_RATE REG0X2B[1:0]** should be set to provide appropriate down sampling through digital filters.
There are two special cases in which the OSR will be 100. If MCLK is 400 times the input sample rate of the DAC or the output sample rate of the ADC, the OSR will be 100. In the first case, set **CLK_DIVIDER.CLK_ADC_SRC_REG0X3[7:6]**=2'b10 (1/4) for ADC path, and DAC path need to set **CLK_DIVIDER.CLK_DAC_SRC_REG0X3[5:4]**=2'b10 (1/4) and **DAC_RATE.DAC_CTRL1_REG0X2C[2:0]**=3b'000 , in the second case the clock to the ADC and DAC will be adjusted automatically.

Example 1:
To configure Fs = 48 kHz, MCLK = (256*Fs) = 12.288MHz, and CLK_ADC = 6.144MHZ
Set:

- SYSCLK_SRC = MCLK
- CLK_ADC_SRC = 1/2
- ADC OSR = 128

Example 2:
To configure Fs = 16 kHz, MCLKI = 12.288MHz, and CLK_ADC = 4.096MHz
Set:

- SYSCLK_SRC = MCLK
- MCLK_SRC = 1/3
- CLK_ADC_SRC = 1
- ADC OSR = 256

6.1 I2S/PCM Clock Generation

In master mode, BCLK can be derived from MCLK via a programmable divider, and the FS can be derived from BCLK via another programmable divider.

To select specific Fs values, both dividers must be set according to the block diagram and the equation below.

$$BCLK = Fs \times data\ length \times channels$$

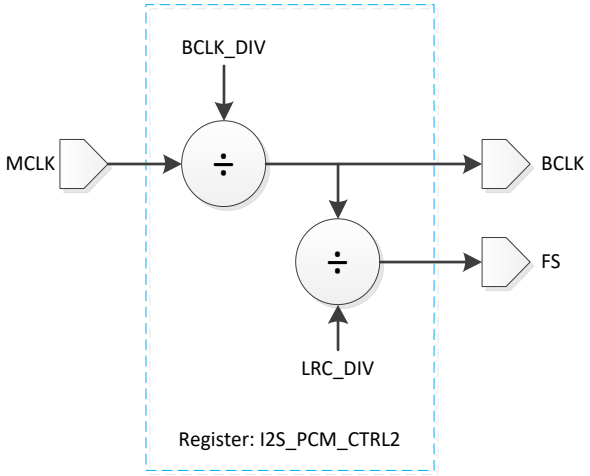


Figure 15: BCLK and FS Frequency Selection

Bits	BCLK_DIV
000	Divided by 1
001	Divided by 2
010	Divided by 4
011	Divided by 8
100	Divided by 16
101	Divided by 32

Table 11: Register Settings

Bits	LRC_DIV
00	Divided by 256
01	Divided by 128
10	Divided by 64
11	Divided by 32

Table 12: Register Settings

Example 1:
If we want an Fs of 48 kHz and 16 bit data is to be sent to the I2S bus (2 channel)

- BCLK = 48000*16*2 = 1.536MHz and MCLK = 48000*256 = 12.288MHz
- Set BCLK_DIV = 1/8
- Set LRC_DIV = 1/32

Or 32 bit data is to be sent

- BCLK = 48000*32*2 = 3.073MHz and MCLK = 48000*256 = 12.288MHz
- Set BCLK_DIV = 1/4
- Set LRC_DIV = 1/64

Example 2:
If we want an Fs of 16 kHz and 16 bit data is to be sent to the I2S bus (2 channel)

- BCLK = 16000*16*2 = 512kHz and MCLK = 16000*256 = 4.096MHz
- Set BCLK_DIV = 1/8
- Set LRC_DIV = 1/32

32 bit data is to be sent,

- BCLK = 16000*32*2 = 1.024MHz and MCLK = 16000*256 = 4.096MHz

- Set BCLK_DIV = 1/4
- Set LRC_DIV = 1/64
-

Example 3:
If we want an Fs of 16 kHz and 32 bit data is to be sent to the I2S TDM bus (4 channels)

- BCLK = 16000*32*4 = 2.048MHz and MCLK = 16000*256 = 4.096MHz
- Set BCLK_DIV = 1/2
- Set LRC_DIV = 1/128

6.2 Frequency Locked Loop(FLL)

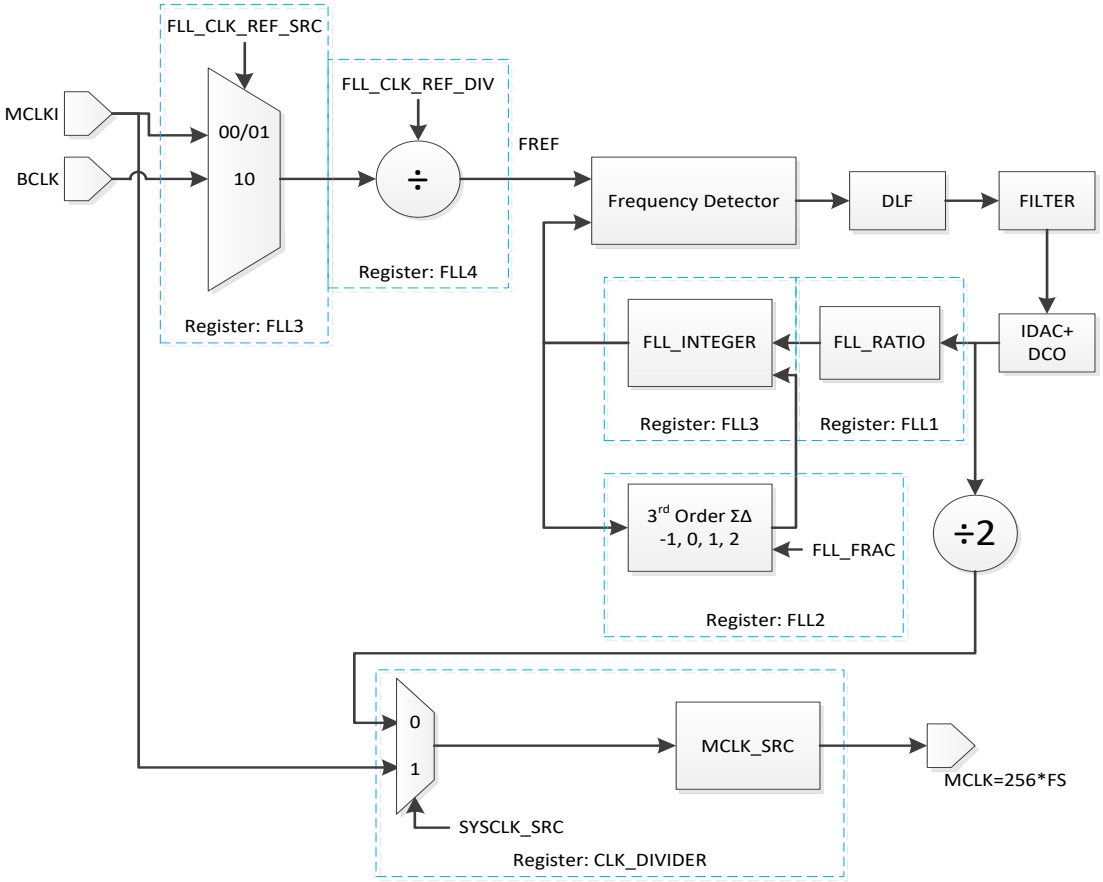


Figure 16: FLL Block diagram

The integrated FLL can be used to generate a SYSMCLK from a wide variety of reference sources such as, MCLK, BCLK, and FS or as a free running clock in the absence of an external reference. It can also create a stable SYSMCLK from less stable sources due to its tolerance of jitter.
The FLL output frequency is determined by the following parameters.

- FLL_RATIO based on input clock frequency
- MCLK_SRC Divider
- FLL_INTEGER: 10 bit Integer Input
- FLL_FRAC: 16 bit Fractional Input
- FLL_CLK_REF_DIV Divider

To determine these settings, the following output frequency equations are used.

1. FDCO = (FREF / FLL_CLK_REF_DIV) X FLL_INTEGER.FLL_FRAC X FLL_RATIO
2. MCLK = (FDCO X MCLK_SRC) / 2

Where FREF is the reference clock frequency for FLL, MCLK is the desired system frequency, and FDCO is the frequency of DCO in decimal.

Example:
If the reference frequency (FREF) is 12MHz, the desired sampling rate (Fs) is 48 kHz, and SYSCLK = 256Fs, what are the output frequency parameters?

Using these requirements, the following can be determined.

- MCLK = 256 × 48kHz = 12.288MHz
- Using Equation 2:
  - FDCO = 2 X MCLK / MCLK_SRC = 2 X 12.288MHz X MCLK_SRC
    - For FDCO to remain between 90MHz – 100MHz, MCLK_SRC must be chosen to be 1/4. This and other values for MCLK_SRC can be seen on the register tables.
  - FDCO = (2 × 12.288MHz) / (1/4) = 98.304MHz
- Using Equation 1:
  - FLL_INTEGER.FLL_FRAC = FDCO X FLL_CLK_REF_DIV / (FREF X FLL_RATIO)
    - FDCO = 98.304MHz
    - FLL_RATIO = 1 because of FREF ≥ 512 kHz.
    - FLL_CLK_REF_DIV = 1 since FREF = MCLKI (12MHz)
  - FLL_INTEGER.FLL_FRAC = 98.304MHz X 1 / (12MHz X 1) = 8.192
- Now retrieve or convert the parameter values into their corresponding HEX values
  - FLL_RATIO = 1 (for input clock frequency ≥ 512Khz)
  - MCLK_SRC = 1/4
  - FLL_INTEGER = 8
  - FLL_FRAC = 0.192 = 3,221,225 (0.192 x 2^24) = 24'h3126E9

Please Note:

- FLL_CLK_REF_DIV can be used to reduce the reference frequency for SYSMCLK by dividing the input by 1, 2, 4, or 8. Use this to ensure the reference clock frequency is less than or equal to 13.5MHz.
- FDCO must be within the 90MHz – 100MHz or the FLL cannot be guaranteed across the full range of operation.
- FLL_FRAC must be set to 0 for low power mode.

- FLL6.SDM_EN REG0X09[14] to create decimal part of frequency, if (DCO frequency)/(FLL input reference frequency) is not a integer . If the ratio is integer, it still can be on for lower noise output but higher power consumption.
- When FLL uses free running mode, NAU88L21 needs to be set as a master in I2S_PCM_CTRL2.MS0 REG0X1D[3]=1
- Set FLL6.CHB_FILTER_EN REG0X08[14] = '1' to enable FLL Loop Filter. Select filter clock source by FLL6.CHB_FILTER¬_EN REG0X08[13]. Select DCO input by FLL6.FILTER_SW REG0X08[12]. FLL6.CUTOFF500 REG0X09[13] & FLL6.CUTOFF600 REG0X09[12] can be used to define FLL cuttoff frequency at 500KHz or 600KHz. 500KHz will provide the best FLL performance but consume more power.
- set FLL6.FLL_FLTR_DITHER_SEL REG0X09[7:6] = '01' or '10' or '11' as 1LSB / 2LSB / 3LSB random bits to Randomize the number of Filter Output Bits to average out output noise. If '00', there is no dither.

7. Control Interfaces

The NAU88L21 includes a serial control bus that provides access to all the device control registers, it may be configured as a 2-wire interface that conforms to industry standard implementations of the I²C serial bus protocol.

7.1 2-Wire-Serial Control Mode (I²C Style Interface)

The 2-wire bus is a bidirectional serial bus protocol. This protocol defines any device that sends data onto the bus as a transmitter (or master), and any device receiving data as the receiver (or slave). The NAU88L21 can function only as a slave when in the 2-wire interface configuration.

To enable 2-wire I2C Style interface,

(1) Set register 0x1A &= 0xFF00 (Set 0x1A[7:0] = 8'b00000000)

(2) Externally pull GPIO1/CSB pin = LOW (0V) , I2C device address = 0x1B; Or externally pull GPIO1/CSB pin = HIGH (3.3V) , I2C device address = 0x54

7.2 2-Wire Protocol Convention

All 2-Wire interface operations must begin with a START condition, which is a HIGH-to-LOW transition of SDIO while SCLK is HIGH. All 2-Wire interface operations are terminated by a STOP condition, which is a LOW to HIGH transition of SDIO while SCLK is HIGH. A STOP condition at the end of a read or write operation places the device in a standby mode.

An acknowledge (ACK), is a software convention used to indicate a successful data transfer. To allow for the ACK response, the transmitting device releases the SDIO bus after transmitting eight bits. During the ninth clock cycle, the receiver pulls the SDIO line LOW to acknowledge the reception of the eight bits of data.

Following a START condition, the master must output a device address byte. This consists of a 7-bit device address, and the LSB of the device address byte is the R/W (Read/Write) control bit. When R/W=1, this indicates the master is initiating a read operation from the slave device, and when R/W=0, the master is initiating a write operation to the slave device. If the device address matches the address of the slave device, the slave will output an ACK during the period when the master allows for the ACK signal.

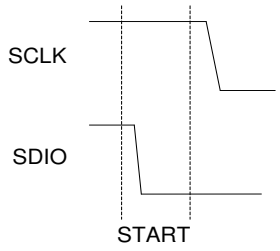


Figure 17: Valid START Condition

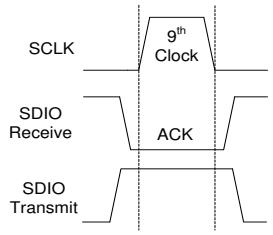


Figure 18: Valid Acknowledge

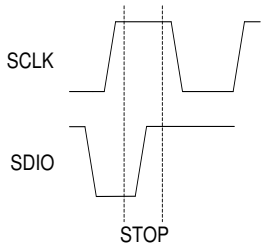


Figure 19: Valid STOP Condition

Please Note:

- Sometimes, I2C needs to use level shifter between different supplies domains. During Acknowledge, receiver side (CODEC) will pull low, and transmit side (MCU) is disable and pull high by pull high resistor. Because NAU88L21 SDIO can sink 2mA by default setting (maximum up to 8mA,) shown as below Figure 20, R_{PU1} and R_{PU2} need to be select such that total current VDDB/R_{PU1}+ VDD_MCU/R_{PU2} during Acknowledge should not be too large to exceed SDIO sinking capability.

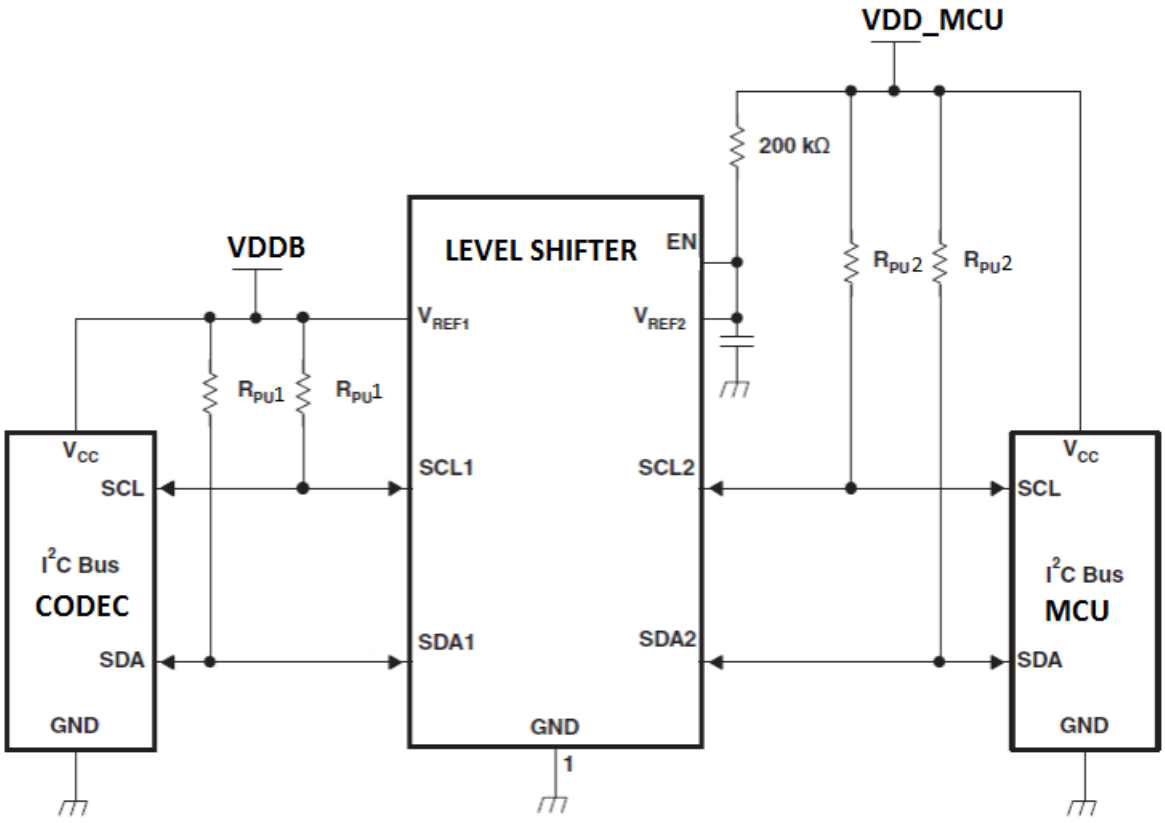


Figure 20: Typical I2C level shifter circuit

7.3 2-Wire Write Operation

A Write operation consists of a three-byte instruction followed by one or more Data Bytes. A Write operation requires a START condition, followed by a valid device address byte with R/W=0, a valid control address byte, data byte(s), and a STOP condition.

The Device Address of the NAU88L21 is either 0x1B (CSB=0) or 0x54 (CSB=1). If the Device Address matches this value, the NAU88L21 will respond with the expected ACK signaling as it accepts the data being transmitted to it.

0	0	1	1	0	1	1	R/W	Device Address Byte
A15	A14	A13	A12	A11	A10	A9	A8	Control Address Bytes
A7	A6	A5	A4	A3	A2	A1	A0	
D15	D14	D13	D12	D11	D10	D9	D8	Data Bytes
D7	D6	D5	D4	D3	D2	D1	D0	

Figure 21: Slave Address Byte, Control Address Byte, and Data Byte

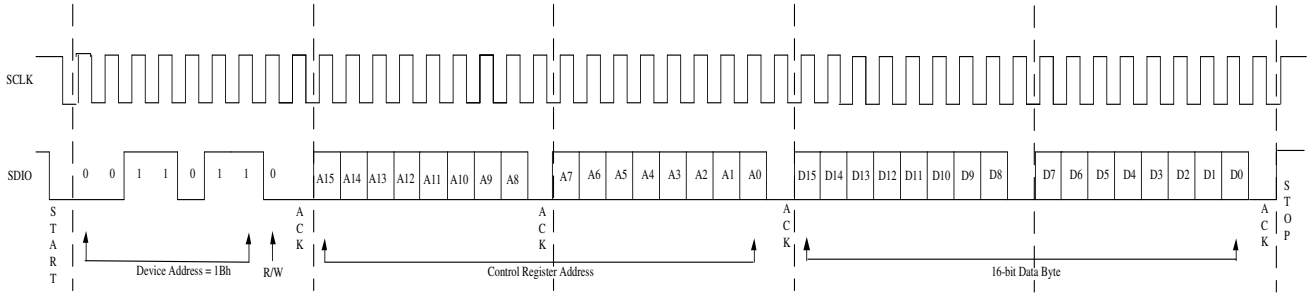


Figure 22:2-Wire Write Sequence

7.4 2-Wire Read Operation

A Read operation consists of a three-byte Write instruction followed by a Read instruction of one or more data bytes. The bus master initiates the operation issuing the following sequence: a START condition, device address byte with the R/W bit set to “0”, and a Control Register Address byte. This indicates to the slave device which of its control registers is to be accessed.

If the device address matches this value, the NAU88L21 will respond with the expected ACK signaling as it accepts the Control Register Address being transmitted into it. After this, the master transmits a second START condition, and a second instantiation of the same device address, but now with R/W=1.

After again recognizing its device address, the NAU88L21 transmits an ACK, followed by a two byte value containing the 16 bits of data from the selected control register inside the NAU88L21.

During this phase, the master generates the ACK signaling with each byte transferred from the NAU85L40. If there is no STOP signal from the master, the NAU88L21 will internally auto-increment the target Control Register Address and then output the two data bytes for this next register in the sequence.

This process will continue as long as the master continues to issue ACK signaling. If the Control Register Address being indexed inside the NAU88L21 reaches the value 0xFFFF (hexadecimal) and the value for this register is output, the index will roll over to 0x0000. The data bytes will continue to be output until the master terminates the read operation by issuing a STOP condition.

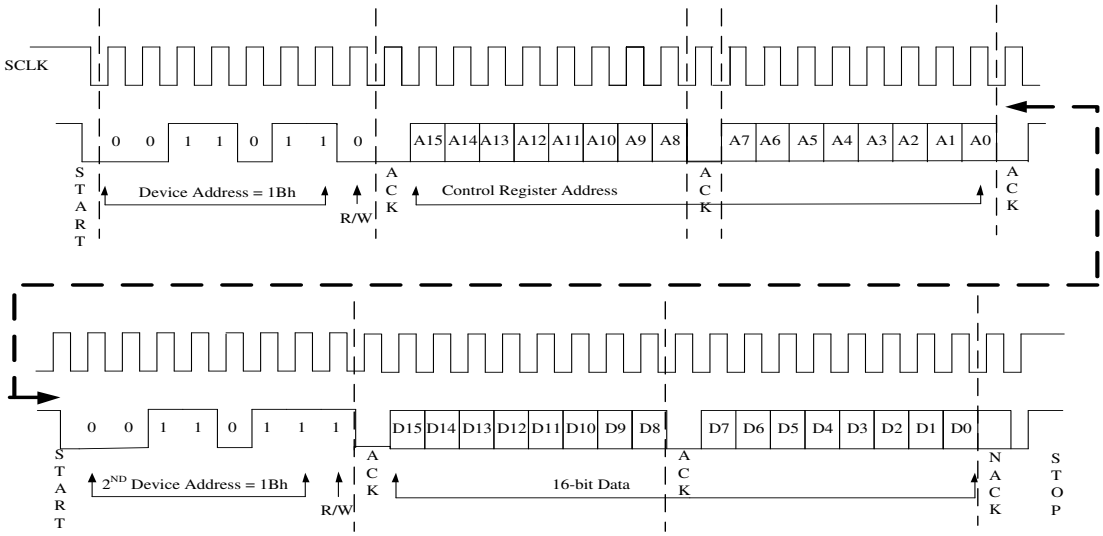


Figure 23:2-Wire Read Sequence

7.5 Digital Serial Interface Timing

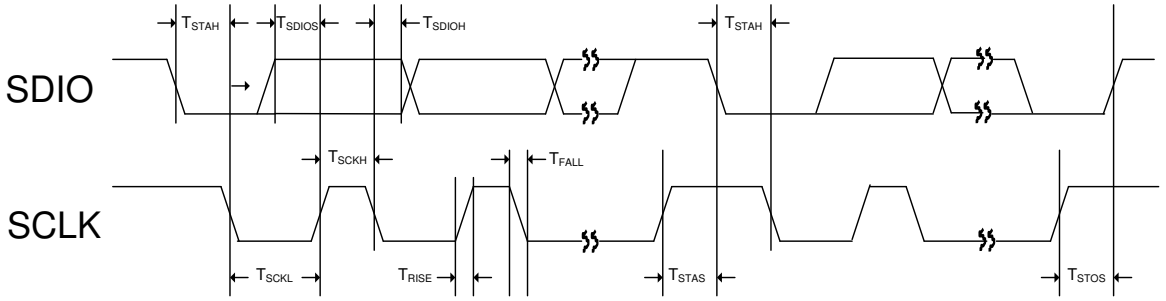


Figure 24: Two-wire Control Mode Timing

SYMBOL	DESCRIPTION	MIN	TYP	MAX	UNIT
T_{STAH}	SDIO falling edge to SCLK falling edge hold timing in START / Repeat START condition	600	-	-	ns
T_{STAS}	SCLK rising edge to SDIO falling edge setup timing in Repeat START condition	600	-	-	ns
T_{STOS}	SCLK rising edge to SDIO rising edge setup timing in STOP condition	600	-	-	ns
T_{SCKH}	SCLK High Pulse Width	600	-	-	ns
T_{SCKL}	SCLK Low Pulse Width	1,300	-	-	ns
T_{RISE}	Rise Time for all 2-wire Mode Signals	-	-	300	ns
T_{FALL}	Fall Time for all 2-wire Mode Signals	-	-	300	ns
T_{SDIOS}	SDIO to SCLK Rising Edge DATA Setup Time	100	-	-	ns
T_{SDIOH}	SCLK falling Edge to SDIO DATA Hold Time	0	-	600	ns

Table 13 Digital Serial Interface Timing Parameters

7.6 Software Reset

The NAU88L21 and all of its control registers can be reset to “default”, initial conditions by writing any value to REG0X00 using the two-wire interface mode.

8. Digital Audio Interfaces

The NAU88L21 can be configured as either the master or the slave, and the Slave mode is the default if this bit is not written. In master mode, NAU88L21 outputs both Frame Sync (FS) and the audio data bit clock (BCLK) and has full control of the data transfer. In the slave mode, an external controller supplies BCLK and FS. Data is latched on the rising edge of BCLK; SDO clocks out ADC data, while SDI clocks in data for the DACs.

When not transmitting data, SDO pulls LOW in the default state. Depending on the application, the output can be configured to pull up or pull down. When the time slot function is enabled (see below), there are additional output state modes including controlled tristate capability.
NAU88L21 supports six audio formats; right justified, left justified, I2S, PCMA, PCMB, and PCM Time Slot.

8.1 Digital Audio Interface

8.1.1 Right-Justified Audio Data

In right-justified mode, the LSB is clocked on the last BCLK rising edge before FS transitions. When FS is HIGH, channel_0 data is transmitted and when FS is LOW, channel_1 data is transmitted. This can be seen in the image below.

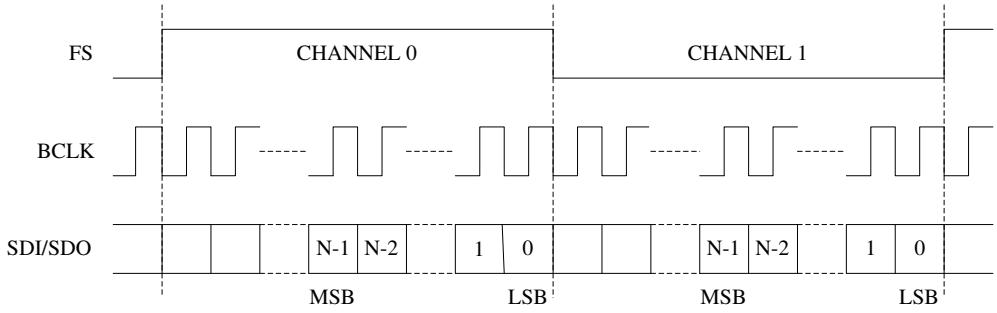


Figure 25: Right-Justified Audio Interface

8.1.2 Left-Justified Audio Data

In left-justified mode, the MSB is clocked on the first BCLK rising edge after FS transitions. When FS is HIGH, channel_0 data is transmitted and when FS is LOW, channel_1 data is transmitted. This can be seen in the figure below.

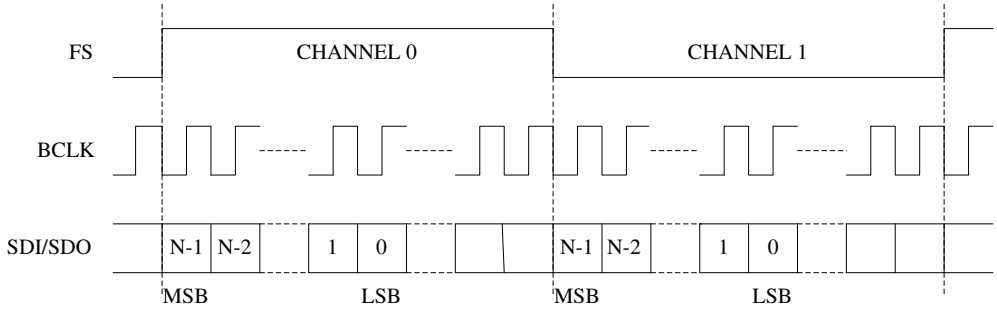


Figure 26: Left-Justified Audio Interface

8.1.3 I2S Audio Data

In I²S mode, the MSB is clocked on the second BCLK rising edge after FS transitions. When FS is LOW, left channel data is transmitted and when FS is HIGH, right channel data is transmitted. This can be seen in the figure below.

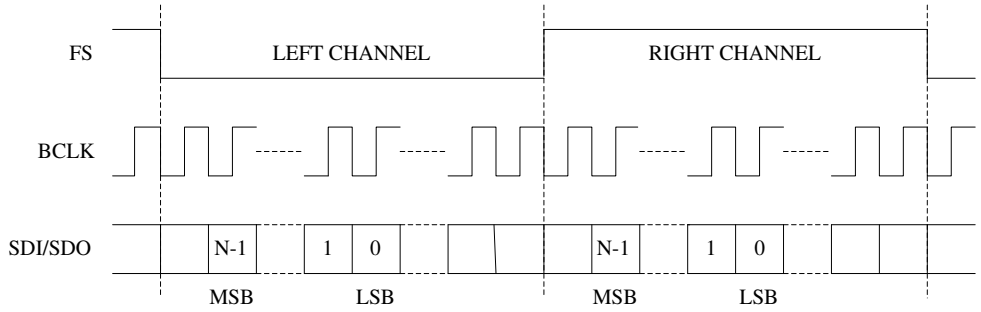


Figure 27: I2S Audio Interface

8.1.4 PCMA Audio Data

In the PCM A mode, channel 0 data is transmitted first followed immediately by channel 1 data. The channel 0 MSB is clocked on the second BCLK rising edge after the FS pulse rising edge, and channel 1 MSB is clocked on the next BCLK after the left channel LSB. This can be seen in the figure below.

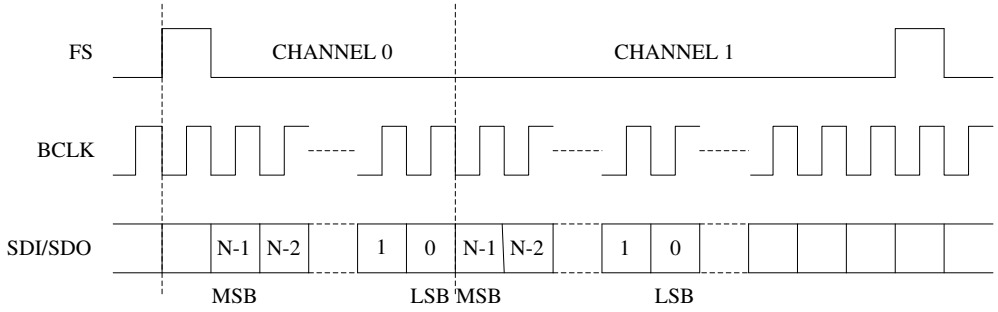


Figure 28: PCMA Audio Interface

8.1.5 PCMB Audio Data

In the PCMB mode, channel_0 data is transmitted first followed immediately by channel_1 data. Channel 0 MSB is clocked on the first BCLK rising edge after the FS pulse rising edge, and channel_1 MSB is clocked on the next BCLK after channel_0 LSB. This can be seen in the figure below.

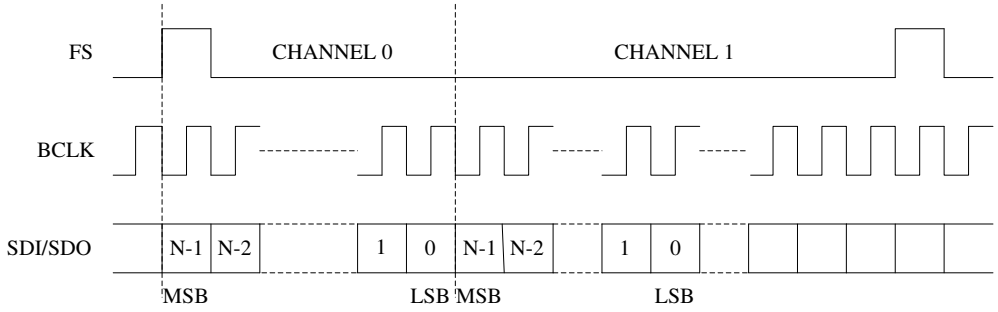


Figure 29: PCMB Audio Interface

8.1.6 PCM Time Slot Audio Data

The PCM time slot mode is used to allocate different time slots for ADC and DAC data. This can be useful when multiple NAU88L21 chips or other devices are sharing the same audio bus. This will allow each chip"s audio to be delayed around each other without interference.

Normally, the DAC and ADC data are clocked immediately after the Frame Sync (FS), however, in the PCM time slot mode; the audio data can be delayed by left / right channel PCM time slot start value in the registers.
These delays can be seen before the MSB in the figure below.

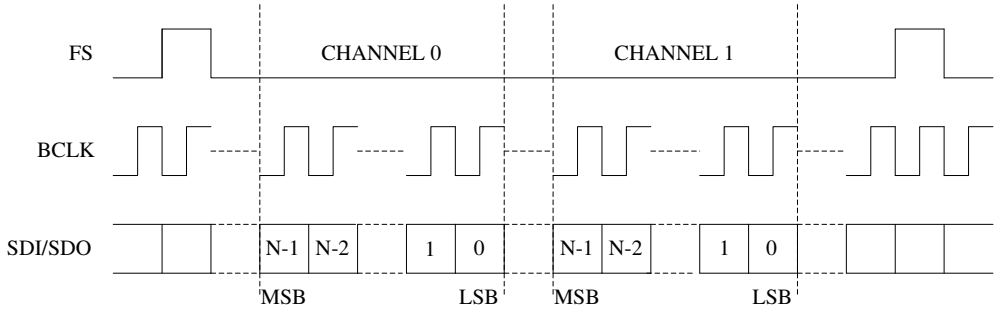


Figure 30: PCM Time Slot Audio Interface

The PMC time slot mode can be also used to swap channel 0 and channel 1 audio or cause both channels to use the same data. When using the NAU88L21 with other driver chips, the SDO pin can be set to pull up or pull down or high impedance during no transmission. Tri-stating on the negative edge allows the transmission of data by multiple sources in adjacent timeslots with reduced risk of bus driver contention.

8.1.7 TDM I2S Audio Data

In I2S mode, the MSB is clocked on the second BCLK rising edge after FS transitions. When FS is LOW, channel_0 then channel_2 data is transmitted and when FS is HIGH, channel_1 then channel_3 data is transmitted. This is shown in the figure below.

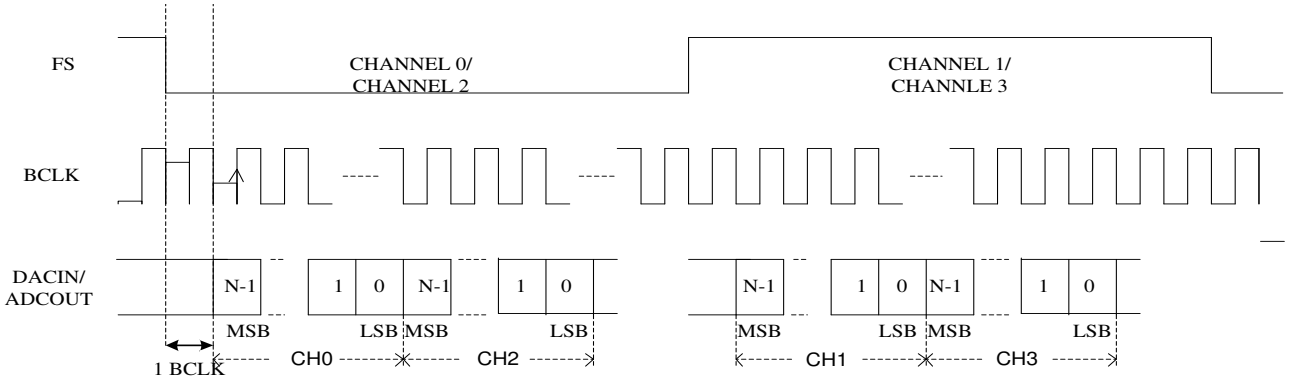


Figure 31: TDM I2S Audio Format

8.1.8 TDM PCMA Audio Data

In the PCMA mode, channel_0 data is transmitted first followed sequentially by channel_1, 2, and 3 immediately after. The channel_0 MSB is clocked on the second BCLK rising edge after the FS pulse rising edge, and the subsequent channel's MSB is clocked on the next BCLK after the previous channel's LSB. This is shown in the figure below.

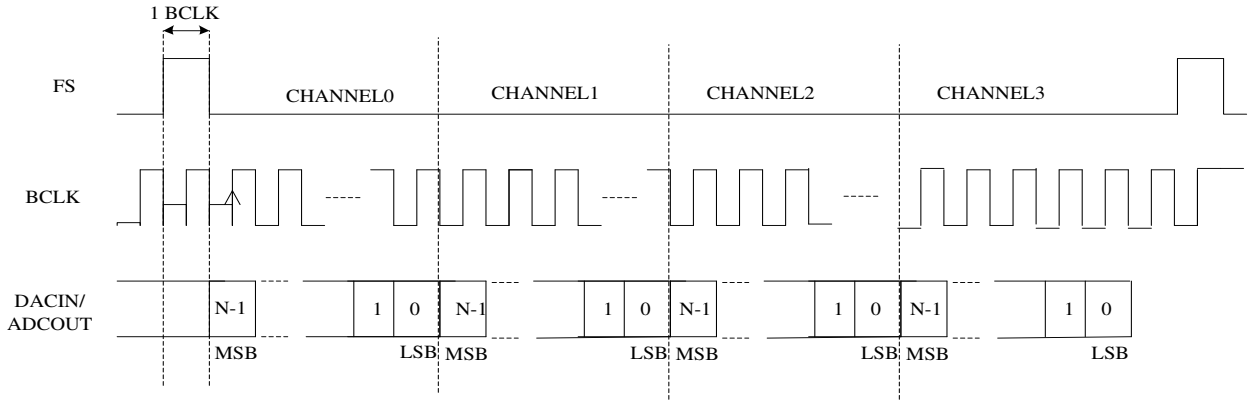


Figure 32: TDM PCMA Audio Format

8.1.9 TDM PCMB Audio Data

In TDM PCMB mode, channel_0 data is transmitted first followed immediately by channel_1 data. The channel_0 MSB is clocked on the first BCLK rising edge after the FS pulse rising edge, and channel_1 MSB is clocked on the next SCLK after channel_0 LSB.

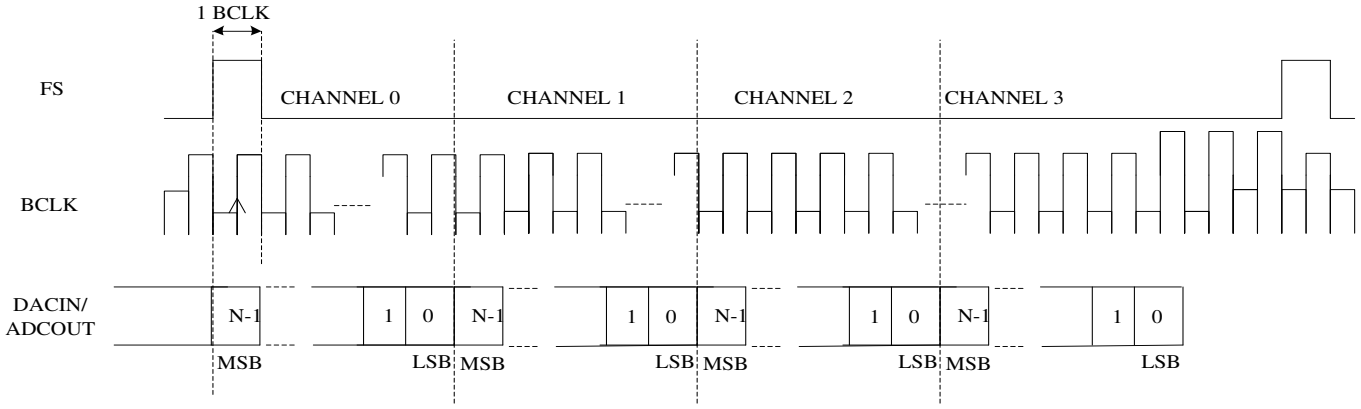


Figure 33: TDM PCMB Audio Format

8.1.10 TDM PCM Offset Audio Data

The PCM offset mode is used to delay the time at which DAC data is clocked. This increases the flexibility of the NAU88L21 to be used in a wide range of system designs. One key application of this feature is to enable multiple NAU88L21 or other devices to share the audio data bus, thus enabling more than four channels of audio. This feature may also be used to swap channel data, or to cause multiple channels to use the same data.

Normally, the DAC data are clocked immediately after the Frame Sync (FS). In this mode audio data is delayed by a delay count specified in the device control registers. The channel 0 MSB is clocked on the BCLK rising edge defined by the delay count set in .This can be seen in the figure below.

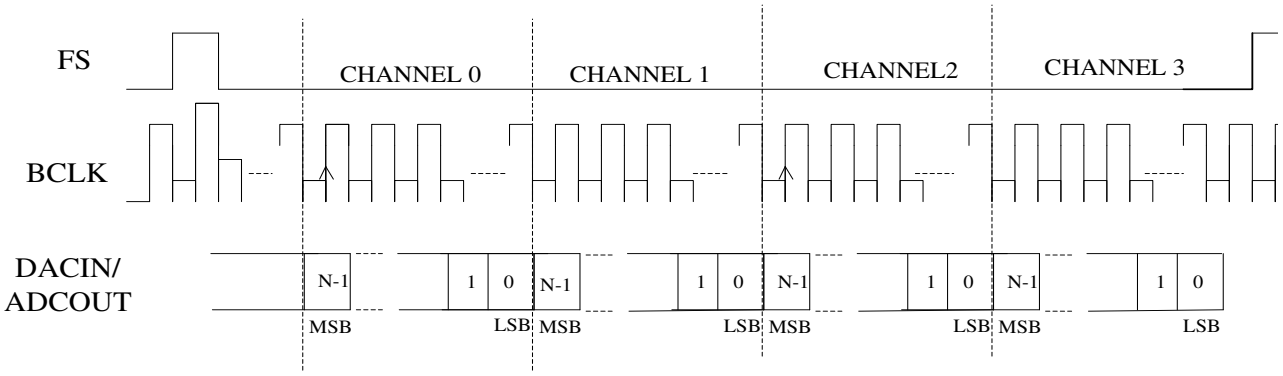


Figure 34: TDM PCM Offset Audio Format

8.2 Digital Audio Interface Timing Diagrams

8.2.1 Digital Audio Interface Slave Mode

Figure 20 provides the timing for Audio Interface Slave Mode.

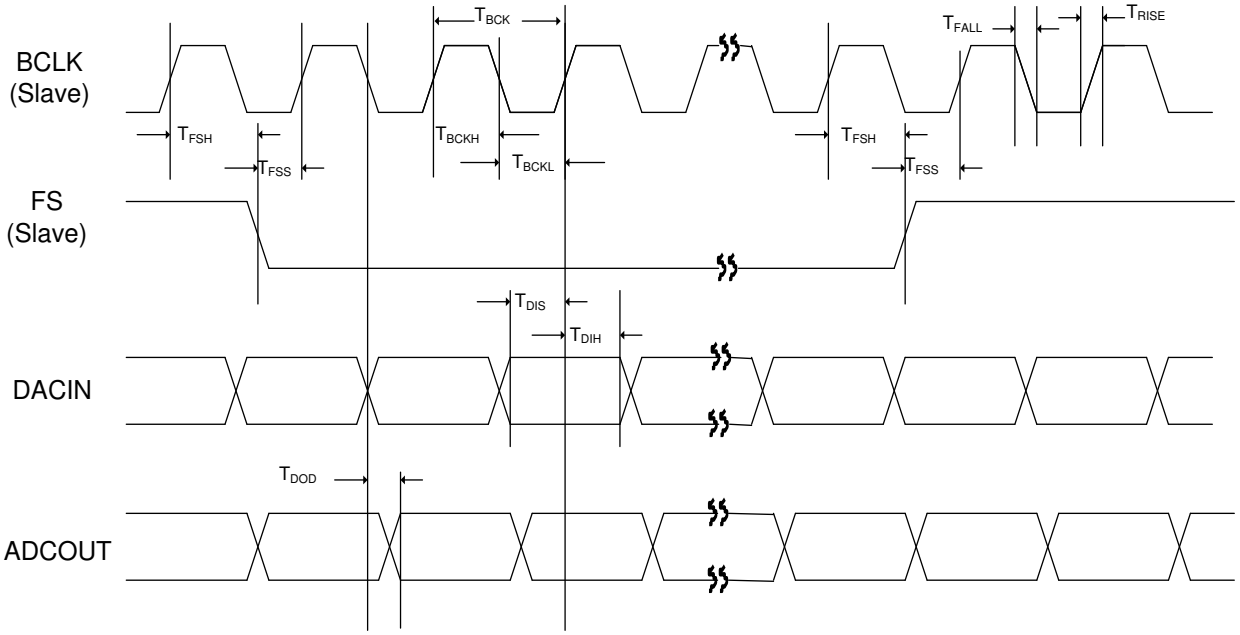


Figure 35 Audio Interface Slave Mode Timing

SYMBOL	DESCRIPTION	MIN	TYP	MAX	UNIT
T_{BCK}	BCLK Cycle Time in Slave Mode	50	-	-	ns
T_{BCKH}	BCLK High Pulse Width in Slave Mode	20	-	-	ns
T_{BCKL}	BCLK Low Pulse Width in Slave Mode	20	-	-	ns
T_{FSS}	FS to BCLK Rising Edge Setup Time in Slave Mode	20	-	-	ns
T_{FSH}	BCLK Rising Edge to FS Hold Time in Slave Mode	20	-	-	ns
T_{RISE}	Rise Time for All Audio Interface Signals	-	-	$0.135T_{BCK}$	ns
T_{FALL}	Fall Time for All Audio Interface Signals	-	-	$0.135T_{BCK}$	ns
T_{DIS}	DACIN to BCLK Rising Edge Setup Time	15	-	-	ns
T_{DIH}	BCLK Rising Edge to DACIN Hold Time	15	-	-	ns
T_{DOD}	BCLK Falling Edge to ADCOUT Delay Time	-	-	10	ns

Table 14 Audio Interface Slave Mode Timing Parameters

8.2.2 Digital Audio Interface Master Mode

provides the timinig for Audio Interface Master Mode

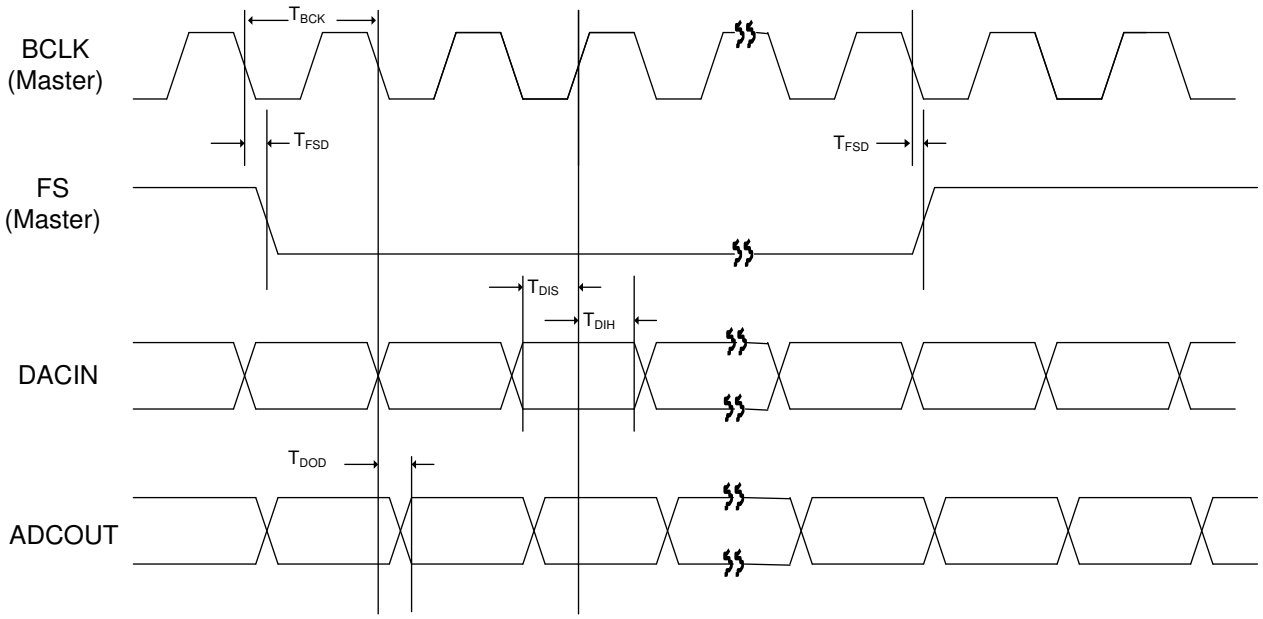


Figure 36 Audio Interface Master Mode Timing

SYMBOL	DESCRIPTION	MIN	TYP	MAX	UNIT
T_{BCK}	BCLK Cycle Time in Master Mode	50	60	-	ns
T_{FSD}	BCLK Falling Edge to FS Delay Time in Master Mode	-	-	10	ns
T_{DIS}	DACIN to BCLK Rising Edge Setup Time	15	-	-	ns
T_{DIH}	BCLK Rising Edge to DACIN Hold Time	15	-	-	ns
T_{DOD}	BCLK Falling Edge to ADCOUT Delay Time	-	-	10	ns

Table 15 Audio Interface Master Mode Timing Parameters

8.2.3 PCM Audio Interface Slave Mode

I2S or PCM Audio Data can be processed using either Slave Mode or Master Mode. The timing diagram for PCM Audio Data in Slave Mode is shown in Figure 20.

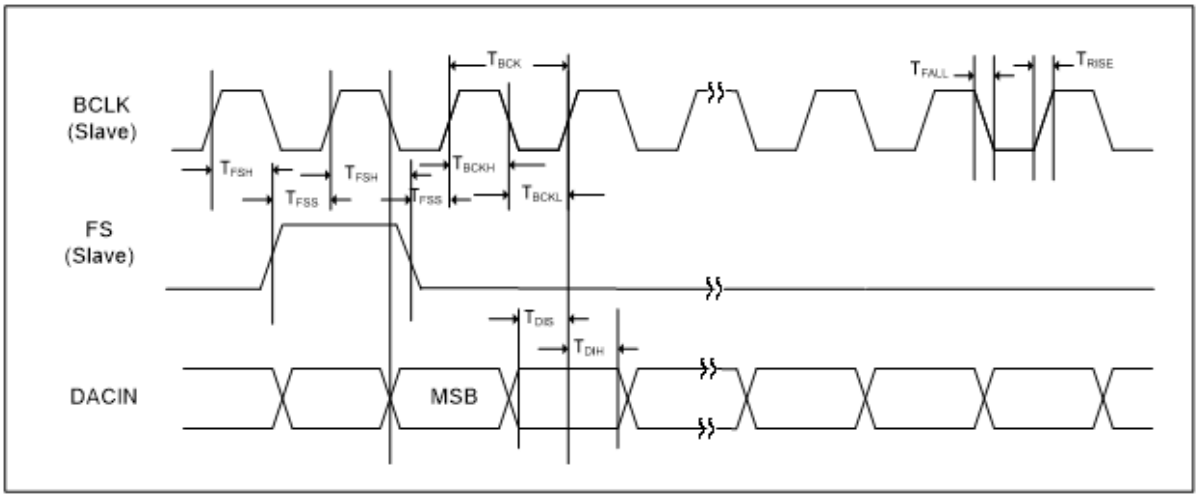


Figure 37 PCM Audio Interface Slave Mode

8.2.4 PCM Audio Interface Master Mode

I2S or PCM Audio Data can be processed using either Master or Slave Mode. The timing diagram for PCM Audio Data in Master Mode is shown in Figure 20.

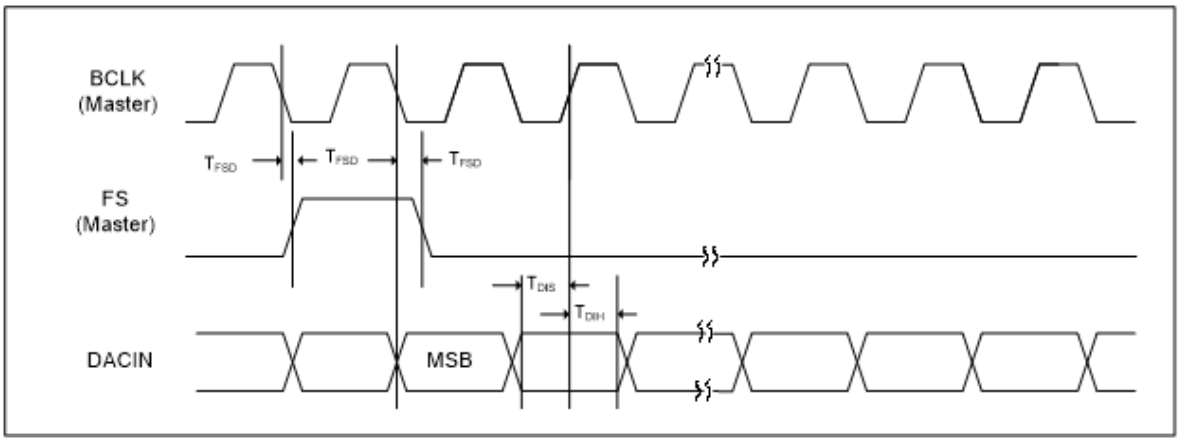


Figure 38 PCM Audio Interface Master Mode Timing

8.2.5 PCM Time Slot Audio Interface Slave Mode

PCM Time Slot Data can be processed using either Slave Mode or Master Mode. The timing diagram for PCM Time Slot Audio Data in Slave Mode is shown in Figure 2039.

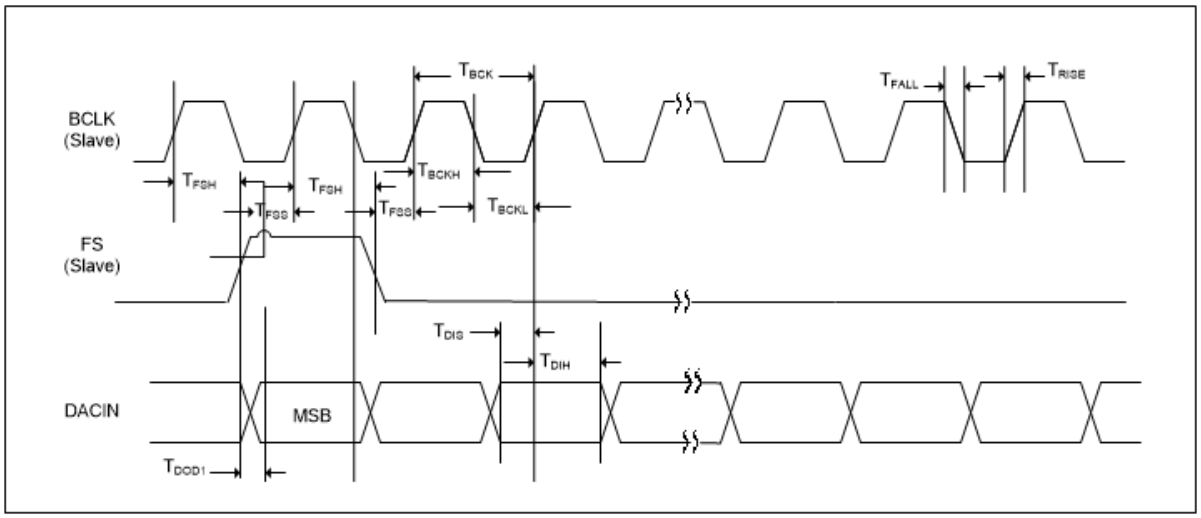


Figure 39 PCM Time Slot Audio Interface Slave Mode Timing

8.2.6 PCM Time Slot Audio Interface Master Mode

The timing diagram for PCM Time Slot Audio Data in Master Mode is shown in Figure 2040.

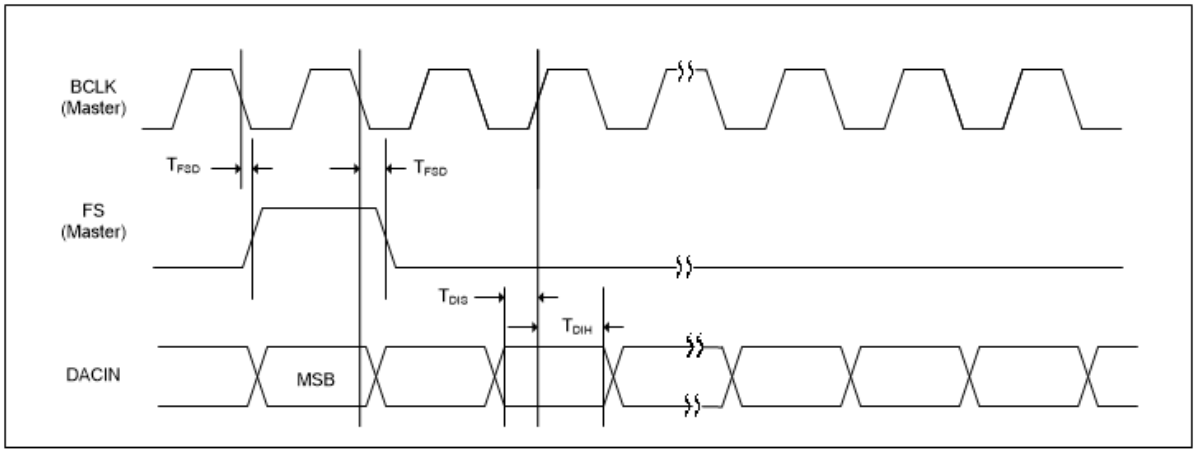


Figure 40 PCM Time Slot Audio Interface Master Mode Timing

9. Outputs

The NAU88L21 provides a pair of Class G ground-reference headphone outputs.

9.1 Class G Headphone Driver and Charge Pump

The NAU88L21 uses Class G speaker drivers powered by a charge pump for the headphones. For typical operation with large and small signals the charge pump provides ±1.8V and ±0.9V, respectively. These output drivers are driven by dedicated left and right DACs and can provide 30mW of power to a 32Ω load (in CSP package).

Three capacitors are needed to generate the negative voltage from the positive 1.8V. Typically, 2μF ceramic capacitors are used.

- The Fly Back capacitor is connected between pins CPCA and CPCB.
- The Positive Output Decoupling capacitor is applied from pin CPVOUTP to ground (VSSCP).
- The Negative Output Decoupling capacitor is applied from pin CPOUTN to ground (VSSCP).

The Class G will be turned on only if DAC signal level is bigger than the threshold in the register settings, and the peak output can be also configured differently by register settings.

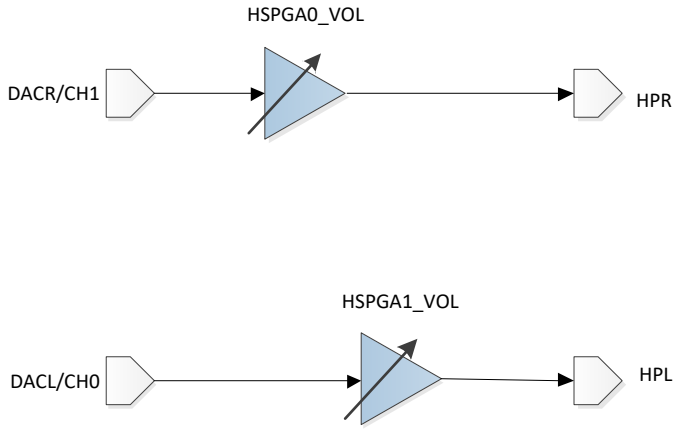


Figure 41: DAC to Headphone out path diagram

10. Control and Status Registers

REG	Function	Name	Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	Description
0	HARDWARE_RST	HARDWARE_RESET																	**Hardware Reset** (Write any value ***once*** to reset all the registers.)
1	ENA_CTRL	CMLCK_ENB																	**PGA Common Mode Lock Enable Control** 0 = Enable (DEFAULT) 1 = Disable
		CLK_DAC_INV																	**DAC Clock Inversion In Analog Domain Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		RDACEN																	**Right Channel DAC Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		LDACEN																	**Left Channel DAC Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		RADCEN																	**Right Channel ADC Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		LADCEN																	**Left Channel ADC Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		DCLK_ADC_EN																	**ADC Clock Enable Control** 0 = Disable 1 = Enable (DEFAULT)
		DCLK_DAC_EN																	**DAC Clock Enable Control** 0 = Disable 1 = Enable (DEFAULT)
		CLK_IMM_EN																	**IMM Clock Enable Control** 0 = Disable 1 = Enable (DEFAULT)
		CLK_I2S_EN																	**I2S Clock Enable Control** 0 = Disable 1 = Enable (DEFAULT)
		CLK_BIST_EN																	**BIST Clock Enable Control** 0 = Disable 1 = Enable (DEFAULT)
		CLK_OTP_EN																	**OTP (One Time Programming) Clock Enable Control** 0 = Disable 1 = Enable (DEFAULT)
		CLK_DRC_EN																	**DRC Clock Enable Control** 0 = Disable 1 = Enable (DEFAULT)
		DEFAULT	0	0	0	0	0	0	0	0	1	1	1	1	1	1	1	1	**0x00FF**
3	CLK_DIVIDER	SYSCLK_SRC																	**Master Clock Source Select** 0 = MCLK_PIN (DEFAULT) 1 = ½ DCO_CLK
		CLK_CODEC_SRC																	**ADC & DAC Clock Source Select** 0 = From internal MCLK (DEFAULT) 1 = From MCLK_PIN or ½ DCO_CLK
		CLK_DAC_PL																	**DAC Clock Polarity** 0 = Non-inverted (DEFAULT) 1 = Inverted
		CLK_ADC_PL																	**ADC Clock Polarity** 0 = Non-inverted (DEFAULT) 1 = Inverted
		CLK_GPIO_SRC																	**Scaling Divider For GPIO Clock From MCLK** 00 = 1/8 (DEFAULT) 01 = 1 10 = 1/2 11 = 1/4

REG	Function	Name	Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	Description
		CLK_ADC_SRC																	**Scaling Divider For ADC Clock From CODEC_SRC** 00 = 1 01 = 1/2 (DEFAULT) 10 = 1/4 11 = 1/8
		CLK_DAC_SRC																	**Scaling Divider For DAC Clock From CODEC_SRC** 00 = 1 01 = 1/2 (DEFAULT) 10 = 1/4 11 = 1/8
		MCLK_SRC																	**Scaling Divider For MCLK From SYSCLK_SRC** 0000 = 1 (DEFAULT) 0001 = Inverted 0010 = 1/2 0011 = 1/4 0100 = 1/8 0101 = 1/16 0110 = 1/32 0111 = 1/3 1000 = 1 1001 = Inverted 1010 = 1/6 1011 = 1/12 1100 = 1/24 1101 = 1/48 1110 = 1/96 1111 = 1/5
		DEFAULT	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**1**	**0**	**1**	**0**	**0**	**0**	**0**	**0x0050**
4	**FLL1**	FLLISELDAC																	**Increase Drive Strength Of FLL DAC** 000 = (DEFAULT)
		ICTRL_LATCH																	**FLL Latch Drive Strength Multiplier** (When FLL running at high frequency with long decimal number, DSP needs to operate at high speed. By adjusting ICTRL_LATCH, FLL DSP can optimize between performance and power consumption (111 has highest power consumption for FLL DSP.) On the other hand, (DCO frequency)/(FLL input reference frequency)=integer, default setting can be used to reduce power. This register is using thermometer coding.) 000 = (DEFAULT) 001 = 1x 011 = 2x 111 = 3x
		ICTRL_V2I																	**Amp Half Bias-Current Select** (Amp bias current must be reduced to 50% of its nominal value.) 00 = No power reduction (DEFAULT) 01 = Half bias current on FLL_BIAS_AMP2X 10 = Half bias current on FLL_BIAS_AMP 11 = Half current on both amps
		FLL_LOCK_BP																	**Manual Force of FLL Lock Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		FLL_RATIO																	**Input Clock Frequency Select** 0000001 = For input clock frequency ≥ 512KHz 0000010 = For input clock frequency ≥ 256KHz 0000100 = For input clock frequency ≥ 128KHz 0001000 = For input clock frequency ≥ 64KHz 0010000 = For input clock frequency ≥ 32KHz 0100000 = For input clock frequency ≥ 8KHz 1000000 = For input clock frequency ≥ 4KHz
		DEFAULT	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0x0000**
5	**FLL2**	DOUT2DCO_RSV																	**FLL DCO Frequency Free-running Mode**
		DEFAULT	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**1**	**0**	**1**	**1**	**1**	**1**	**0**	**0**	**0x00BC**
6	**FLL3**	GAIN_ERR																	**FLL Gain Error** (The threshold is comparison between DCO and target frequency. 1111 has the most accurate DCO to target frequency. However, the gain error setting conditionally and inversely depends on FLL input reference clock rate. Higher FLL reference input frequency can only set lower gain error, such as 0000 for input reference from MCLK=12.288MHz. On the other side, if FLL reference input is from Frame sync, 48KHz, higher error gain can apply such as 1111.) 0000 = (DEFAULT) 0001 = x1 0010 = x2 0011 = x3 0100 = x4 0101 = x5 0110 = x6 0111 = x8

REG	Function	Name	Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	Description
																			1000 = x9 1001 = x10 1010 = x12 1011 = x16 1100 = x17 1101 = x18 1110 = x20 1111 = x24
		FLL_CLK_REF_SRC																	**FLL Reference CLK Source Select** 00 = MCLK pin (DEFAULT) 01 = MCLK pin 10 = BCLK pin
		FLL_INTEGER																	**FLL 10-bit Integer Input**
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	1	0	0	0	**0x0008**
7	FLL4	HIGHBWE																	**High Bandwidth Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		FLL_CLK_REF_DIV_4CHK																	**FLL CLK_REF Divider For Accurate Lock Detection** 000 = 1 (DEFAULT) 001 = 1/2 010 = 1/4 011 = 1/8 100 = 1/16 101 = 1/32
		FLL_CLK_REF_DIV																	**FLL Pre-Scale Divider** 00 = 1 (DEFAULT) 01 = 1/2 10 = 1/4 11 = 1/8
		FLL_N2																	**FLL 10-bit Integer DCO Divider For FLL Filter Clock** (The value is in orders of 2. When 0x8[13]=1, it selects DCO clock as FLL filter clock. The filter clock rate needs to be less than 1Mhz. With setting proper value, filter clock can be divided down from DCO clock. For example, DCO runs at 96Mhz, by setting value 0x60=96, filter clock becomes 1MHz.)
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	1	0	0	0	0	**0x0010**
8	FLL5	PDB_DACICTRL																	**FLL Loop Filter To Reduce FLL Output Noise Enable Control** (Especially, (DCO frequency)/(FLL input reference frequency) is not an integer.) 0 = Disable (DEFAULT) 1 = Enable (By REG0x09[13:12])
		CHB_FILTER_EN																	**Select Filter Clock Source Select** 0 = Select REFCLK 1 = Select divided DCO clock based on register FLL_N2 (DEFAULT)
		CLK_FILTER_SW																	**IDAC Input Select** 0 = Select filter output (DEFAULT) 1 = Select accumulator output when feedback divider is integer, it can use for saving power but more jitter
		FILTER_SW																	**FLL Loop Filter Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		FLL_LOCK_LENGTH																	**Set FLL Lock-In Length** (Set the time that FLL must stay within the lock-in range before lock signal goes high.)
		DEFAULT	0	1	0	0	0	0	0	0	0	0	0	0	0	0	0	0	**0x4000**
9	FLL6	DCO_EN																	**FLL Free-running Mode Enable Control** (Need to enable 0x76[12] **BIASEN**) 0 = Disable (DEFAULT) 1 = Enable
		SDM_EN																	**FLL Sigma-Delta Modulator Enable Control** (To create decimal part of frequency, if (DCO frequency)/(FLL input reference frequency) is not a integer. If the ratio is integer, it still can be on for lower noise output but higher power consumption.) 0 = Disable 1 = Enable (DEFAULT)

REG	Function	Name	Bit																Description
			15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	
		CUTOFF500																	**FLL 500KHz Cut-off Frequency Enable Control** (If 0x8[14]=1, it sets loop filter cutoff frequency at 600Khz. It will give the best FLL performance with the highest power consumption.) 0 = Disable 1 = Enable (DEFAULT)
		CUTOFF600																	**FLL 600KHz Cut-off Frequency Enable Control** (If 0x8[14]=1, it sets loop filter cutoff frequency at 600Khz. It will give a moderate FLL performance with moderate power consumption.) 0 = Disable (DEFAULT) 1 = Enable
		VREFSEL																	**VREF Select** 00 = 1.8V 01 = 1.56V 10 = 1.65V (DEFAULT) 11 = 1.75V
		CHKFS256_EN																	**Samples/frame Sync Enable Control** 0 = Disable (DEFAULT) 1 = Enable the function to check for 256
		FS8X_SEL																	**Frame Sync Select** 0 = Total samples per 4 frame sync 1 = Total samples per 8 frame sync (DEFAULT)
		FLL_FLTR_DITHER_SEL																	**Filter Output Random Bit Select** 00 = No dither (DEFAULT) 01 = The LSB is a random bit 10 = Two LSBs are random bits 11 = Three LSBs are random bits
		FLL_SD_DITHER_SEL																	**Input Of SD Modulator Random Bit Select** 00 = No dither (DEFAULT) 01 = The LSB is a random bit 10 = Two LSBs are random bits 11 = Three LSBs are random bits
		DLR																	**FLL Dynamic Lock Range** 0000 = (DEFAULT)
		DEFAULT	**0**	**1**	**1**	**0**	**1**	**0**	**0**	**1**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0x6900**
A	**FLL7**	**FLL_FRAC_H**																	**MSB Portion Of FLL 24-bit Fractional Input FLL_FRAC[23:16]**
		DEFAULT	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**1**	**1**	**0**	**0**	**0**	**1**	**0x0031**
B	**FLL8**	**FLL_FRAC_L**																	**LSB Portion Of FLL 24-bit Fractional Input FLL_FRAC[15:0]**
		DEFAULT	**0**	**0**	**1**	**0**	**0**	**1**	**1**	**0**	**1**	**1**	**1**	**0**	**1**	**0**	**0**	**1**	**0x26E9**
D	**JACK_DET_CTRL**	**MANU_SPKR_DWN1R**																	**Manual Access SPKR_DWN1R** 0 = Pull down (DEFAULT)
		MANU_SPKR_DWN1L																	**Manual Access SPKR_DWN1L** 0 = Pull down (DEFAULT)
		JK_1_PL																	**Jack Detection Source 1 Configuration** 00 = From GPIO2JD1 (DEFAULT) 01 = From inverted GPIO2JD1 10 = Ignore the input and set to 0 11 = Ignore the input and set to 1
		JD_RESTART																	**Manual Restart Jack Detection** (Toggle this bit to 1 and then to 0 to restart the jack detection.)
		DB_BP_MODE																	**Jack Detect De-bounce Bypass** 0 = Enable de-bounce circuit (need to set REG4B[0] = 1 to enable the CLK) (DEFAULT) 1 = Bypass the de-bounce circuit
		INSERT_DT																	**Insertion De-bounce Time** 2^(INSERT_DT +2) ms
		EJECT_DT																	**Ejection De-bounce Time** 2^(EJECT_DT +2) ms
		JKDET_PL																	**Jack Insertion/ Detection Logic Polarity** 0 = Falling edge (DEFAULT) 1 = Rising edge

REG	Function	Name	Bit																Description
			15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	
		JKDET_LOGIC																	**Jack Detection Logic Control** 0 = OR gate (DEFAULT) 1 = AND gate
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
F	INTERRUPT_MASK	IRQ_PL																	**IRQ Pin Logic Select** 0 = Active low (DEFAULT) 1 = Active high
		IRQ_PS																	**IRQ Pin Pull Select** 0 = Pull down (DEFAULT) 1 = Pull up
		IRQ_PE																	**IRQ Pin Pull Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		IRQ_DS																	**IRQ Pin Drive Current Select** 0 = Low drive current (DEFAULT) 1 = High drive current
		IRQ_OE																	**IRQ Pin Output Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		APR_EMRGENCY_SHTDWN1_INTP_MASK																	**APR Emergency Shutdown Interrupt Mask** 0 = Unmask interrupt; turn off IRQ pad, but interrupt register status is controlled by 0x12[9] (DEFAULT) 1 = Mask interrupt; turn on IRQ pad
		RMS_INTP_MASK																	**RMS Interrupt Mask** 0 = Unmask interrupt; turn off IRQ pad, but interrupt register status is controlled by 0x12[8] (DEFAULT) 1 = Mask the interrupt; turn on IRQ pad
		KEY_RELEASE_INTP_MASK																	**Key Release Interrupt Mask** 0 = Unmask interrupt; turn off IRQ pad, but interrupt register status is controlled by 0x12[7] (DEFAULT) 1 = Mask the interrupt; turn on IRQ pad
		KEY_INTP_MASK																	**Key Pressed Interrupt Mask** 0 = Unmask interrupt; turn off IRQ pad, but interrupt register status is controlled by 0x12[6] (DEFAULT) 1 = Mask the interrupt; turn on IRQ pad
		MCLKDET_INTP_MASK																	**Missing MCLK Detection Interrupt Mask** 0 = Unmask interrupt; turn off IRQ pad, but interrupt register status is controlled by 0x12[5] (DEFAULT) 1 = Mask the interrupt; turn on IRQ pad
		MIC_DET_INTP_MASK																	**MIC Detection Interrupt Mask** 0 = Unmask interrupt; turn off IRQ pad, but interrupt register status is controlled by 0x12[4] (DEFAULT) 1 = Mask the interrupt; turn on IRQ pad
		JK_EJECT_INTP_MASK																	**Jack Ejection Interrupt Mask** 0 = Unmask interrupt; turn off IRQ pad, but interrupt register status is controlled by 0x12[2] (DEFAULT) 1 = Mask the interrupt; turn on IRQ pad
		JK_DET_INTP_MASK																	**Jack Insertion Interrupt Mask** 0 = Unmask interrupt; turn off IRQ pad, but interrupt register status is controlled by 0x12[0] (DEFAULT) 1 = Mask the interrupt; turn on IRQ pad
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
10	IRQ_STATUS	APR_EMRG_SHTDWN																	**APR Emergency Short Circuit Shutdown IRQ Status**
		RMS_INT																	**Impedance Measurement IRQ Status**
		KEY_RELEASE_INT																	**Key Release For Key Detection IRQ Status**
		KEY_INT																	**Key Detection IRQ Status**
		MCLK_DET_INT																	**Missing MCLK Detection IRQ Status**
		MIC_DET_INT																	**MIC Detection IRQ Status**
		JACK_EJCT_IRQ																	**Jack Ejection IRQ Status** 00 = Cleared state 01 = Jack ejection detected 10 = A jack eject interrupt was cleared due to a jack insertion 11 = Undefined

REG	Function	Name	Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	Description
		JACK_DET_IRQ																	**Jack insertion IRQ status** 00 = Cleared state 01 = Jack insertion detected 10 = A jack insert interrupt was cleared due to a jack ejection 11 = Undefined
		DEFAULT	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**READ ONLY**
11	INT_CLR_KEY_STATUS	INT_CLR_KEY_STATUS																	**Write Operation** (Write bits[15:0] clear corresponding REG10 [15:0] Write 1s to bits that you want to reset to 0, except) Bit0 or Bit1 = clear Jack insertion interrupt Bit2 or Bit3 = clear Jack ejection interrupt
		DEFAULT	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**READ/WRITE**
12	INTERRUPT_DIS_CTRL (WRITE MODE)	APR_EMRG_SHTDWN_INT_DIS																	**APR Emergency Short Circuit Shutdown Interrupt Disable Control** 0 = Enable interrupt; interrupt status read from register or IRQ pad 1 = Disable interrupt and IRQ pad (DEFAULT)
		RMS_INT_DIS																	**RMS Impedance Measurement Interrupt Disable Control** 0 = Enable interrupt; interrupt status read from register or IRQ pad 1 = Disable interrupt and IRQ pad (DEFAULT)
		KEY_RELEASE_INT_DIS																	**Key Release Interrupt Disable Control** 0 = Enable interrupt; interrupt status read from register or IRQ pad 1 = Disable interrupt and IRQ pad (DEFAULT)
		KEY_INT_DIS																	**Key Interrupt Disable Control** 0 = Enable interrupt; interrupt status read from register or IRQ pad 1 = Disable interrupt and IRQ pad (DEFAULT)
		MCLKDET_INT_DIS																	**MCLK Detection Interrupt Disable Control** 0 = Enable interrupt; interrupt status read from register or IRQ pad 1 = Disable interrupt and IRQ pad (DEFAULT)
		MIC_DET_INT_DIS																	**MIC Detection/Headset Configuration Interrupt Disable Control** 0 = Enable interrupt; interrupt status read from register or IRQ pad 1 = Disable interrupt and IRQ pad (DEFAULT)
		JACK_EJCT_INT_DIS																	**Jack Ejection Interrupt Disable Control** 0 = Enable interrupt; interrupt status read from register or IRQ pad 1 = Disable interrupt and IRQ pad (DEFAULT)
		JACK_DET_INT_DIS																	**Jack Insertion/Detection Interrupt Disable Control** 0 = Enable interrupt; interrupt status read from register or IRQ pad 1 = Disable interrupt and IRQ pad (DEFAULT)
		DEFAULT	**1**	**1**	**1**	**1**	**1**	**1**	**1**	**1**	**1**	**1**	**1**	**1**	**1**	**1**	**1**	**1**	**0xFFFF**
13	DMIC_CTRL	DMIC_DS																	**DMIC Clock Drive Current Select** (For high ***Cload*** > 20pF, enable high drive current.) 0 = Low drive current (DEFAULT) 1 = High drive current
		DMIC_SLEW																	**DMIC Clock Slew Rate Select** (For high ***Cload*** > 20pF, use faster slew rate.) 000 = Slowest slew rate (DEFAULT) ▼ 111 = Fastest slew rate
		CLK_DMIC_SRC																	**DMIC Clock Speed Select** 00 = ADC clock (DEFAULT) 01 = ADC clock / 2 10 = ADC clock / 4 11 = ADC clock / 8
		DMICEN																	**Digital Microphone Mode Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		DEFAULT	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0x0000**

REG	Function	Name	Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	Description
1A	GPIO12_CTRL	GPIO2OUT																	**GPIO2 Programmable Output**
		GPIO2_PS																	**GPIO2JD1 Pull Select** 0 = Pull up (DEFAULT) 1 = Pull down
		GPIO2_DS																	**GPIO2JD1 Drive Current Select** 0 = Low drive current (DEFAULT) 1 = High drive current
		GPIO2_PE																	**GPIO2JD1 Pin Pull Enable Control** 0 = Enable (DEFAULT) 1 = Disable
		GPIO2_OE																	**GPIO2JD1 Output Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		GPIO1POL																	**GPIO1 Polarity** 0 = Non-inverted (DEFAULT) 1 = Inverted logic of the CSB/GPIO1 function output selected by GPIO1SEL
		GPIO1SEL																	**CSB/GPIO1 Function Select** 000 = output 0 (DEFAULT) 001 = Jack status from the AND/OR logic 010 = SCLK_I 011 = SD_I 100 = output divided FLL clock 101 = FLL locked condition (logic 1 = PLL locked) 110 = SD_O 111 = OSC_CLK
		GPIO1_PS																	**GPIO1CSB Pull Select** (If GPIO1_PE=1) 0 = Pull up (DEFAULT) 1 = Pull down
		GPIO1_DS																	**GPIO1CSB Drive Current Select** 0 = Low drive current (DEFAULT) 1 = High drive current
		GPIO1_PE																	**GPIO1CSB Pin Pull Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		GPIO1_OE																	**GPIO1CSB Output Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	**0x0000**
1B	TDM_CTRL	TDM																	**TDM Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		PCM_OFFSET_MODE_CTRL																	**PCM Offset In TDM Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		ADCPHS0																	**ADC Audio Data Left-right Ordering Select** 0 = Left ADC data in left phase of LRP (DEFAULT) 1 = Left ADC data in right phase of LRP (left-right reversed)
		DACPHS1																	**DAC Right Channel Audio Data Left-right Ordering Select** 0 = Right DAC data in right phase of LRP (DEFAULT) 1 = Right DAC data in left phase of LRP (left-right reversed)
		DACPHS0																	**DAC Left Channel Audio Data Left-right Ordering Select** 0 = Left DAC data in left phase of LRP (DEFAULT) 1 = Left DAC data in right phase of LRP (left-right reversed)

REG	Function	Name	Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	Description
		DAC_LEFT_SEL																	**DAC Left Channel Source Under TDM Mode (TDM = 1 & PCM_OFFSET_MODE_CTRL = 0)** I2S : 000 = From Slot 0 (DEFAULT) 001= From Slot 1 010 = From Slot 2 011 = From Slot 3 100 = RESERVED 101 = RESERVED 110 = RESERVED 111 = RESERVED PCM: 000 = From slot 0 (DEFAULT) 001 = From slot 1 010 = From slot 2 011 = From slot 3 100 = From slot 4 101 = From slot 5 110 = From slot 6 111 = From slot 7
		DAC_RIGHT_SEL																	**DAC Right Channel Source Under TDM Mode** I2S: 000 = From Slot 0 (DEFAULT) 001 = From Slot 1 010 = From Slot 2 011 = From Slot 3 100 = RESERVED 101 = RESERVED 110 = RESERVED 111 = RESERVED PCM: 000 = From slot 0 (DEFAULT) 001 = From slot 1 010 = From slot 2 011 = From slot 3 100 = From slot 4 101 = From slot 5 110 = From slot 6 111 = From slot 7
		ADC_TX_SEL_L																	**ADC Left Channel Source Under TDM/I2S Mode** 00 = From slot 0 (DEFAULT) 01 = From slot 2 10 = From slot 4 11 = From slot 6
		ADC_TX_SEL_R																	**ADC Right Channel Source Under TDM/I2S Mode** 00 = From slot 1 (DEFAULT) 01 = From slot 3 10 = From slot 5 11 = From slot 7
		DEFAULT	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0x0000**
1C	I2S_PCM_CTRL1	DACCM0																	**DAC Companding Mode Select** 00 = Off (DEFAULT - Normal linear operation) 01 = RESERVED 10 = μ-law companding 11 = A-law companding
		ADCCM0																	**ADC Companding Mode Select** 00 = Off (DEFAULT - Normal linear operation) 01 = RESERVED 10 = μ-law companding 11 = A-law companding
		ADDAP0																	**ADC Output Data Stream Directly Routed To DAC Input Data Path Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		CMB8_0																	**8-bit Word For Companding Mode Of Operation Enable Control** 0 = Normal operation (DEFAULT - No companding) 1 = 8-bit operation for companding mode
		UA_OFFSET																	**uLaw Offset Select** 0 = 1's complement (DEFAULT) 1 = 2's complement
		BCP0																	**Bit Clock Phase Inversion Option For BCLK** 0 = Non-inverted (DEFAULT) 1 = Inverted
		LRP0																	**PCMA & PCMB Left-right Word Ordering Select** 0 = Right Justified/Left Justified/I2S/PCMA mode (DEFAULT) 1 = PCMB Mode Enable - MSB is valid on 1st rising edge of BCLK after rising edge of FS
		WLEN0																	**Word Length of Audio Data Stream Select** 00 = 16-bit word length 01 = 20-bit word length

REG	Function	Name	Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	Description
																			10 = 24-bit word length (DEFAULT) 11 = 32-bit word length
		AIFMT0																	**Audio Interface Data Format Select** 00 = Right justified 01 = Left justified 10 = Standard I2S format (DEFAULT) 11 = PCMA or PCMB audio data format option
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	1	0	1	0	0x000A
1D	I2S_PCM_CTRL2	I2S_TRI																	**I2S Tri State Enable Control** 0 = Normal mode 1 = Output high Z (DEFAULT)
		I2S_DRV																	**I2S Drive Enable Control** 0 = Normal mode (DEFAULT) 1 = Always out
		LRC_DIV																	**LRC(FS) Divider From BCLK Frequency** 00 = 1/256 (DEFAULT) 01 = 1/128 10 = 1/64 11 = 1/32
		PCM_TS_EN0																	**PCM Time Slot Function Enable Control** (Only PCM_A_MODE or PCM_B_MODE (STEREO Only) can be used when PCM Mode is selected.) 0 = Disable time slot function for PCM mode (DEFAULT) 1 = Enable time slot function for PCM mode
		TRI0																	**Without TDM Mode** 0 = Drive the full clock of LSB (DEFAULT) 1 = Tri-state the 2nd half of LSB
		PCM8BIT0																	**8-Bit PCM Select** 0 = Use I2S_PCM_CTRL.WLEN to select word length (DEFAULT) 1 = PCM select 8-bit word length
		PCM_TS_SEL																	**RESERVED**
		ADCDAT0_PE																	**ADCDAT IO Pull Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		ADCDAT0_PS																	**ADCDAT IO Pull Up/Down Enable Control** 0 = Pull down (DEFAULT) 1 = Pull up
		ADCDAT0_OE																	**ADCDAT IO Output Enable Control** 0 = ADCDAT not always out (when no data out, ADCOUT pin becomes high.) 1 = ADCDAT always out (DEFAULT)
		MS0																	**Master/Slave Mode Enable Control** 0 = Slave mode (DEFAULT) 1 = Master mode
		BCLK_DIV																	**BCLK Divider From MCLK Frequency** 000 = 1 (DEFAULT) 001 = 1/2 010 = 1/4 011 = 1/8 100 = 1/16 101 = 1/32
		DEFAULT	1	0	0	0	0	0	0	0	0	0	0	1	0	0	0	0	0x8010
1E	LEFT_TIME_SLOT	FS_ERR_CMP_SEL																	**Triggers Short Frame Sync Signal** (If frame sync is less than) 00 = 252 x MCLK (DEFAULT) 01 = 253 x MCLK 10 = 254 x MCLK 11 = 255 x MCLK
		DIS_FS_SHORT_DET																	**Short Gram Sync Detection Logic Enable Control** 0 = Enable (DEFAULT) 1 = Disable
		TSLOT_L0																	**Left channel PCM Time Slot Start Value / PCM TDM Offset Mode Slot Start Value**
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
1F	RIGHT_TIME_SLOT	TSLOT_R0																	**Right channel PCM Time Slot Start Value / unused for PCM TDM Offset Mode**
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
		BIQ0_A1_L																	**Program ADC BIQ0_A1 Parameter Bit[15:0]**

REG	Function	Name	Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	Description
21	BIQ0_COF1	DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
22	BIQ0_COF2	BIQ0_A1_H																	Program ADC BIQ0_A1 Parameter Bit[18:16]
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
23	BIQ0_COF3	BIQ0_A2_L																	Program ADC BIQ0_A2 Parameter Bit[15:0]
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
24	BIQ0_COF4	BIQ0_A2_H																	Program ADC BIQ0_A2 Parameter Bit[18:16]
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
25	BIQ0_COF5	BIQ0_B0_L																	Program ADC BIQ0_B0 Parameter Bit[15:0]
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
26	BIQ0_COF6	BIQ0_B0_H																	Program ADC BIQ0_B0 Parameter Bit[18:16]
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
27	BIQ0_COF7	BIQ0_B1_L																	Program ADC BIQ0_B1 Parameter Bit[15:0]
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
28	BIQ0_COF8	BIQ0_B1_H																	Program ADC BIQ0_B1 Parameter Bit[18:16]
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
29	BIQ0_COF9	BIQ0_B2_L																	Program ADC BIQ0_B2 Parameter Bit[15:0]
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
2A	BIQ0_COF10	BIQ0_EN																	**BIQ0 ADC Path Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		BIQ0_B2_H																	Program ADC BIQ0_B2 Parameter Bit[18:16]
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000 R/W
2B	ADC_RATE	ADC_L_SRC																	**ADC Left Channel Source Select** **In Non-DMIC Mode** 0 = Latch left channel analog data input into the left channel filter (DEFAULT) 1 = Latch right channel analog data input into the left channel filter **In DMIC Mode** 0 = Left channel in rising edge (DEFAULT) 1 = Left channel in falling edge
		ADC_R_SRC																	**ADC Right Channel Source Select** **In Non-DMIC Mode** 0 = Latch right channel analog data input into the right channel filter (DEFAULT) 1 = Latch left channel analog data input into the right channel filter **In DMIC Mode** 0 = Right channel in falling edge (DEFAULT) 1 = Right channel in rising edge
		SMPL_RATE																	**Generating 2.048MKHz based on Sample Rates** 000 = 48K (DEFAULT) 001 = 32K 110 = 96K 111 = 192K
		GAINCMP																	**RESERVED**
		ADC_RATE																	**ADC SINC Down Select** 00 = Down 32 01 = Down 64 10 = Down 128 (DEFAULT) 11 = Down 256
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	0	0x0002
2C	DAC_CTRL1	CICCLP_OFF																	0 = (DEFAULT)

REG	Function	Name	Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	Description
		CIC_GAIN_ADJ																	**Gain Adjustment** (Fine tunes the DAC output)
		DAC_RATE																	**DAC Oversample Rate Select** 000 = 64 001 = 256 010 = 128 (DEFAULT) 100 = 32
		DEFAULT	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**1**	**0**	**0**	**0**	**0**	**0**	**1**	**0**	**0x0082**
2D	DAC_TRL2	**RESERVED**																	**RESERVED**
		SDMOD_DITHER																	**Bit Numbers Of Dithering On SD Modulator** (Step size is 1bit.) 00000 = No dithering (DEFAULT) 00001 = 1 00010 = 2 00011 = 3 00100 = 4 00101 = 5 00110 = 6 00111 = 7 01000 = 8 01001 = 9 01010 = 10 01011 = 11 01100 = 12 01101 = 13 01110 = 14 01111 = 15
		RESERVED																	**RESERVED**
		RESERVED																	**RESERVED**
		DEFAULT	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0x0000**
2F	DAC_DGAIN_CTRL	DAC1_TO_DAC0_ST																	**DAC CH1 to DAC CH0 Crosstalk Suppression Sidetone Select** (Step size is 0.5dB.) 0xFF = +24dB 0xFE = +23.5dB ▼ 0xCF = 0dB ▼ 0x43 = -70dB 0x42 = RESERVED ▼ 0x0F = RESERVED 0x0E = Mute 0x00 = Mute (DEFAULT)
		DAC0_TO_DAC1_ST																	**DAC CH0 to DAC CH1 Crosstalk Suppression Sidetone Select** (Step size is 0.5dB.) 0xFF = +24dB 0xFE = +23.5dB ▼ 0xCF = 0dB ▼ 0x43 = -70dB 0x42 = RESERVED ▼ 0x0F = RESERVED 0x0E = Mute 0x00 = Mute (DEFAULT)
		DEFAULT	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0x0000**
30	ADC_DGAIN_CTRL	ADC_TO_DAC_ST0																	**ADC to DAC CH0 Sidetone Select** (Step size is 3dB.) 0x00 = Mute (DEFAULT) 0x01 = -42dB ▼ 0x0E = -3dB 0x0F = 0dB

REG	Function	Name	Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	Description
		ADC_TO_DAC_ST1					■	■	■	■									**ADC to DAC CH1 Sidetone Select** (Step size is 3dB.) 0x00 = Mute (DEFAULT) 0x01 = -42dB ▼ 0x0E = -3dB 0x0F = 0dB
		DAC_ST_SEL0									■								**DAC CH0 Sidetone Source Select** 0 = Select ADC CH0 as the side tone source of the DAC CH0 (DEFAULT) 1 = Select ADC CH1 as the side tone source of the DAC CH0
		DAC_ST_SEL1										■							**DAC CH1 Sidetone Source Select** 0 = Select ADC CH1 as the side tone source of the DAC CH1 (DEFAULT) 1 = Select ADC CH0 as the side tone source of the DAC CH1
		DEFAULT	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0x0000**
31	MUTE_CTRL	PGA_SMUTE_STEP	■	■															**Analog Attn Mute Step Select** 00 = 128 sample 01 = 32 sample (DEFAULT) 10 = 16 sample 11 = 1 sample
		DAC_SLOW_UM			■														**DAC Slow Soft Unmute Enable Control** 0 = Disable (16 MCLK per step soft unmute) (DEFAULT) 1 = Enable (512 MCLK per step soft unmute)
		DAC_ZC_UP_EN				■													**DAC Zero Crossing Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		AMUTE_EN					■												**Auto Mute Enable Control** (Generate null output to analog circuitry when 1024 consecutive zeros are detected. De-assert as soon as first non-zero sample is detected.) 0 = Disable (DEFAULT) 1 = Enable
		AMUTE_CTRL						■											**Auto Mute Control** 0 = Both DAC channels must have 0 values for 1024 samples before AMUTE turns on (DEFAULT) 1 = Either Ch0 or Ch1 must have 1024 consecutive zero samples
		SMUTE_EN							■										**Soft Mute Enable Control** 0 = Gradually increase DAC volume to volume register setting (DEFAULT) 1 = Gradually lower DAC volume to zero
		SUMTE_CTRL								■									**DAC Limiter Output Enable Control** 0 = (DEFAULT) 1 = (When soft mute is enabled, DAC limiter output is also muted to remove any DC offset produced by the audio processing block.)
		ADC_ZC_UP_EN														■			**ADC Zero Crossing Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		ADC_SMUTE_EN															■		**ADC Soft Mute Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		DEFAULT	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0x0000**
32	HSVOL_CTRL	HSPGA_ATTN_EN	■																**Headphone Diver Manual Attn Enable Control** (With HSPGA_ATTN_EN and AMUTE_EN enabled) 0 = Disable (DEFAULT) 1 = Enable
		HSPGA_ATTN_AUTO_MODE		■															**Headphone Driver Auto Attn Enable Control** (With HSPGA_ATTN_AUTO_MODE and AMUTE_EN enabled) 0 = Disable (DEFAULT) 1 = Enable
		MUTE_HSPGA2			■														**Right Channel Headphone Driver Manual Mute Enable Control** 0 = Disable (DEFAULT)

REG	Function	Name	Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	Description
																			1 = Enable
		MUTE_HSPGA1																	**Left Channel Headphone Driver Manual Mute Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		HSPGA1_VOL																	**Left Channel Headphone Driver Volume Control** 00 = 0dB (DEFAULT) 01 = -3dB 10 = -6dB 11 = -9dB
		HSPGA2_VOL																	**Right Channel Headphone Driver Volume Control** 00 = 0dB (DEFAULT) 01 = -3dB 10 = -6dB 11 = -9dB
		DEFAULT	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0x0000**
34	**DAC_CTRL**	DGAINR_DAC																	**DAC Right Volume Control** (Step size is 0.5dB.) 0xFF = +24dB 0xFE = +23.5dB ▼ 0xCF = 0dB (DEFAULT) ▼ 0x4B = -66dB 0x4A = RESERVED ▼ 0x0F = RESERVED 0x0E = Mute 0x00 = Mute
		DGAINL_DAC																	**DAC Left Volume Control** (Step size is 0.5dB.) 0xFF = +24dB 0xFE = +23.5dB ▼ 0xCF = 0dB (DEFAULT) ▼ 0x4B = -66dB 0x4A = RESERVED ▼ 0x0F = RESERVED 0x0E = Mute 0x00 = Mute
		DEFAULT	**1**	**1**	**0**	**0**	**1**	**1**	**1**	**1**	**1**	**1**	**0**	**0**	**1**	**1**	**1**	**1**	**0xCFCF**
35	**ADC_DGAIN_CTRL1**	DGAINR_ADC																	**ADC Right Volume Control** (Step size is 0.5dB.) 0xFF = +24dB 0xFE = +23.5dB ▼ 0xCF = 0dB (DEFAULT) ▼ 0x4B = -66dB 0x4A = RESERVED ▼ 0x0F = RESERVED 0x0E = Mute 0x00 = Mute
		DGAINL_ADC																	**ADC Left Volume Control** (Step size is 0.5dB.) 0xFF = +24dB 0xFE = +23.5dB ▼ 0xCF = 0dB (DEFAULT) ▼ 0x4B = -66dB 0x4A = RESERVED ▼ 0x0F = RESERVED 0x0E = Mute 0x00 = Mute
		DEFAULT	**1**	**1**	**0**	**0**	**1**	**1**	**1**	**1**	**1**	**1**	**0**	**0**	**1**	**1**	**1**	**1**	**0xCFCF**

REG	Function	Name	Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	Description
36	ADC_DRC_KNEE_IP12	DRC_ENA_ADC																	**DRC ADC Channel Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		DRC_KNEE2_IP_ADC																	**DRC ADC Knee Point 2 Select** (Step size is 1dB.) 0x00 = 0dB 0x01 = -1dB ▼ 0x14 = -20dB (DEFAULT) ▼ 0x3E = -62dB 0x3F = -63dB
		DRC_SMTH_ENA_ADC																	**DRC ADC Smooth Filter Enable Control** 0 = Disable 1 = Enable (DEFAULT)
		DRC_KNEE1_IP_ADC																	**DRC ADC Knee Point 1 Select** (Step size is 1dB.) 0x00 = 0dB 0x01 = -1dB ▼ 0x06 = -6dB (DEFAULT) ▼ 0x1E = -30dB 0x1F = -31dB
		DEFAULT	**0**	**0**	**0**	**1**	**0**	**1**	**0**	**0**	**1**	**0**	**0**	**0**	**0**	**1**	**1**	**0**	**0x1486**
37	ADC_DRC_KNEE_IP34	DRC_KNEE4_IP_ADC																	**DRC ADC Knee Point 4 Select** (Step size is 1dB.) 0x00 = -35dB 0x01 = -36dB ▼ 0x0F = -50dB (DEFAULT) ▼ 0x1E = -65dB 0x1F = -66dB
		DRC_KNEE3_IP_ADC																	**DRC ADC Knee Point 3 Select** (Step size is 1dB.) 0x00 = -18dB 0x01 = -19dB ▼ 0x12 = -36dB (DEFAULT) ▼ 0x1E = -48dB 0x1F = -49dB
		DEFAULT	**0**	**0**	**0**	**0**	**1**	**1**	**1**	**1**	**0**	**0**	**0**	**1**	**0**	**0**	**1**	**0**	**0x0F12**
38	ADC_DRC_SLOPES	DRC_NG_SLP_ADC																	**DRC ADC Noise Gate Slope** 00 = 1:1 01 = 2:1 10 = 4:1 (DEFAULT) 11 = 8:1
		DRC_EXP_SLP_ADC																	**DRC ADC Expansion Slope** 00 = 1:1 01 = 2:1 10 = 4:1 (DEFAULT) 11 = RESERVED
		DRC_CMP2_SLP_ADC																	**DRC ADC Compressor Slope (Lower Region)** 000 = 0 001 = 1:2 010 = 1:4 011 = 1:8 100 = 1:16 101-110 = RESERVED 111 = 1 (DEFAULT)
		DRC_CMP1_SLP_ADC																	**DRC ADC Compressor Slope (Higher Region)** 000 = 0 001 = 1:2 010 = 1:4 011 = 1:8 100 = 1:16 101-110 = RESERVED 111 = 1 (DEFAULT)
		DRC_LMT_SLP_ADC																	**DRC ADC Limiter Slope** 000 = 0 001 = 1:2 010 = 1:4 011 = 1:8 100 = 1:16 101 = 1:32 110 = 1:64 111 = 1 (DEFAULT)
		DEFAULT	**0**	**0**	**1**	**0**	**0**	**1**	**0**	**1**	**1**	**1**	**1**	**1**	**1**	**1**	**1**	**1**	**0x25FF**

REG	Function	Name	Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	Description
39	ADC_DRC_ATKDCY	DRC_PK_COEF1_ADC																	**DRC ADC Peak Detection Attack Time** (Ts = 1/SMPL_RATE) 0000 = Ts 0001 = 3*Ts 0010 = 7*Ts 0011 = 15*Ts (DEFAULT) 0100 = 31*Ts 0101 = 63*Ts 0110 = 127*Ts 0111 = 255*Ts 1001= 511*Ts
		DRC_PK_COEF2_ADC																	**DRC ADC Peak Detection Release Time** (Ts = 1/SMPL_RATE) 0000 = 63*Ts 0001 = 127*Ts 0010 = 255*Ts 0011 = 511*Ts 0100 = 1023*Ts (DEFAULT) 0101 = 2047*Ts 0110 = 4095*Ts 0111 = 8191*Ts 1001= 16383*Ts
		DRC_ATK_ADC																	**DRC ADC Attack Time** (Ts = 1/SMPL_RATE) 0000 = Ts 0001 = 3*Ts 0010 = 7*Ts 0011 = 15*Ts 0100 = 31*Ts 0101 = 63*Ts (DEFAULT) 0110 = 127*Ts 0111 = 255*Ts 1000 = 511*Ts 1001 = 1023*Ts 1010 = 2047*Ts 1011 = 4095*Ts 1100 = 8191*Ts
		DRC_DCY_ADC																	**DRC ADC Decay Time** (Ts = 1/SMPL_RATE) 0000 = 63*Ts 0001 = 127*Ts 0010 = 255*Ts 0011 = 511*Ts 0100 = 1023*Ts 0101 = 2047*Ts 0110 =4095*Ts 0111 = 8191*Ts (DEFAULT) 1000 = 16383*Ts 1001 = 32757*Ts 1010 = 65535*Ts
		DEFAULT	**0**	**0**	**1**	**1**	**0**	**1**	**0**	**0**	**0**	**1**	**0**	**1**	**0**	**1**	**1**	**1**	**0x3457**
3A	DAC_DRC_KNEE_IP12	DRC_ENA_DAC																	**DRC DAC Channel Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		DRC_KNEE2_IP_DAC																	**DRC DAC Knee Point 2 Select** (Step size is 1dB.) 0x00 = 0dB 0x01 = -1dB ▼ 0x14 = -20dB (DEFAULT) ▼ 0x1E = -62dB 0x1F = -63dB
		DRC_SMTH_ENA_DAC																	**DRC DAC Smooth Filter Enable Control** 0 = Disable 1 = Enable (DEFAULT)
		DRC_KNEE1_IP_DAC																	**DRC DAC Knee Point 1 Select** (Step size is 1dB.) 0x00 = 0dB 0x01 = -1dB ▼ 0x06 = -6dB (DEFAULT) ▼ 0x1E = -30dB 0x1F = -31dB
		DEFAULT	**0**	**0**	**0**	**1**	**0**	**1**	**0**	**0**	**1**	**0**	**0**	**0**	**0**	**1**	**1**	**0**	**0x1486**
3B	DAC_DRC_KNEE_IP34	DRC_KNEE4_IP_DAC																	**DRC DAC Knee Point 4 Select** (Step size is 1dB.) 0x00 = -35dB 0x01 = -36dB ▼ 0x0F = -50dB (DEFAULT) ▼ 0x1E = -65dB 0x1F = -66dB

REG	Function	Name	Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	Description
		DRC_KNEE3_IP_DAC																	**DRC DAC Knee Point 3 Select** (Step size is 1dB.) 0x00 = -18dB 0x01 = -19dB ▼ 0x12 = -36dB (DEFAULT) ▼ 0x1E = -48dB 0x1F = -49dB
		DEFAULT	**0**	**0**	**0**	**0**	**1**	**1**	**1**	**1**	**0**	**0**	**0**	**1**	**0**	**0**	**1**	**0**	**0x0F12**
3C	**DAC_DRC_SLOPES**	DRC_NG_SLP_DAC																	**DRC DAC Noise Gate Slope** 00 = 1:1 01 = 2:1 10 = 4:1 (DEFAULT) 11 = 8:1
		DRC_EXP_SLP_DAC																	**DRC DAC Expansion Slope** 00 = 1:1 01 = 2:1 10 = 4:1 (DEFAULT) 11 = 8:1
		DRC_CMP2_SLP_DAC																	**DRC DAC Compressor Slope (Lower Region)** 000 = 0 001 = 1:2 010 = 1:4 011 = 1:8 100 = 1:16 101-110 = RESERVED 111 = 1 (DEFAULT)
		DRC_CMP1_SLP_DAC																	**DRC DAC Compressor Slope (Higher Region)** 000 = 0 001 = 1:2 010 = 1:4 011 = 1:8 100 = 1:16 101-110 = RESERVED 111 = 1 (DEFAULT)
		DRC_LMT_SLP_DAC																	**DRC DAC Limiter Slope** 000 = 0 001 = 1:2 (DEFAULT) 010 = 1:4 011 = 1:8 100 = 1:16 101 = 1:32 110 = 1:64 111 = 1
		DEFAULT	**0**	**0**	**1**	**0**	**0**	**1**	**0**	**1**	**1**	**1**	**1**	**1**	**1**	**0**	**0**	**1**	**0x25F9**
3D	**DAC_DRC_ATKDCY**	DRC_PK_COEF1_DAC																	**DRC DAC Peak Detection Attack Time** (Ts = 1/SMPL_RATE) 0000 = Ts 0001 = 3*Ts 0010 = 7*Ts 0011 = 15*Ts (DEFAULT) 0100 = 31*Ts 0101 = 63*Ts 0110 = 127*Ts 0111 = 255*Ts 1XXX = RESERVED
		DRC_PK_COEF2_DAC																	**DRC DAC Peak Detection Release Time** (Ts = 1/SMPL_RATE) 0000 = 63*Ts 0001 = 127*Ts 0010 = 255*Ts 0011 = 511*Ts 0100 = 1023*Ts 0101 = 2047*Ts (DEFAULT) 0110 = 4095*Ts 0111 = 8191*Ts 1XXX = RESERVED
		DRC_ATK_DAC																	**DRC DAC Attack Time** (Ts = 1/SMPL_RATE) 0000 = Ts 0001 = 3*Ts 0010 = 7*Ts 0011 = 15*Ts 0100 = 31*Ts 0101 = 63*Ts (DEFAULT) 0110 = 127*Ts 0111 = 255*Ts 1000 = 511*Ts 1001 = 1023*Ts 1010 = 2047*Ts 1011 = 4095*Ts 1100 = 8191*Ts
		DRC_DCY_DAC																	**DRC DAC Decay Time** (Ts = 1/SMPL_RATE) 0000 = 63*Ts 0001 = 127*Ts 0010 = 255*Ts 0011 = 511*Ts 0100 = 1023*Ts 0101 = 2047*Ts 0110 = 4095*Ts 0111 = 8191*Ts (DEFAULT) 1000 = 16383*Ts 1001 =32757*Ts 1010 = 65535*Ts
		DEFAULT	**0**	**0**	**1**	**1**	**0**	**1**	**0**	**0**	**0**	**1**	**0**	**1**	**0**	**1**	**1**	**1**	**0x3457**
41	**BIQ1_COF1**	**BIQ1_A1_L**																	**Program DAC BIQ1_A1 Parameter Bit[15:0]**

REG	Function	Name	Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	Description
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
42	BIQ1_COF2	BIQ1_A1_H																	Program DAC BIQ1_A1 Parameter Bit[18:16]
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
43	BIQ1_COF3	BIQ1_A2_L																	Program DAC BIQ1_A2 Parameter Bit[15:0]
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
44	BIQ1_COF4	BIQ1_A2_H																	Program DAC BIQ1_A2 Parameter Bit[18:16]
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
45	BIQ1_COF5	BIQ1_B0_L																	Program DAC BIQ1_B0 Parameter Bit[15:0]
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
46	BIQ1_COF6	BIQ1_B0_H																	Program DAC BIQ1_B0 Parameter Bit[18:16]
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
47	BIQ1_COF7	BIQ1_B1_L																	Program DAC BIQ1_B1 Parameter Bit[15:0]
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
48	BIQ1_COF8	BIQ1_B1_H																	Program DAC BIQ1_B1 Parameter Bit[18:16]
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
49	BIQ1_COF9	BIQ1_B2_L																	Program DAC BIQ1_B2 Parameter Bit[15:0]
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
4A	BIQ1_COF10	BIQ1_EN																	**BIQ1 DAC Path Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		BIQ1_B2_H																	Program DAC BIQ1_B2 Parameter Bit[18:16]
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
4B	CLASSG_CTRL	CLASSG_CLK_SRC																	**Class G Function Clock Divider** 00 = Clock 2MHz (DEFAULT) 01 = 1/3 MCLK 10 = MCLK 11 = Disable CLK
		CLASSG_TIMER																	**Define The Number Of Milliseconds** (When a Class-G mode signal to go low after it has been below the threshold.) 000001 = 1ms 000010 = 2ms 000100 = 8ms 001000 = 16ms 010000 = 32ms 100000 = 64ms
		CLASSG_THRSLD																	**Threshold for DAC Signal Level Comparison To Generate Class-G Mode Signal** 00 = 1/16 Full Scale (DEFAULT) 01 = 1/8 Full Scale 10 = 3/16 Full Scale 11 = 1/4 Full Scale
		CLASSG_CMP_EN																	**Class-G Compare Path Enable Bit** (Each bit enables according DAC path.) 0 = Disable (DEFAULT) 1 = Enable Bit0 = Left DAC Bit1 = Right DAC
		CLASSG_EN																	**Class-G Function Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
		RESERVED																	RESERVED

REG	Function	Name	Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	Description
4C	IMM_MODE_CTRL	IMM_THRESHOLD																	**Impedance Measurement Threshold To Avoid False Detection** (Each increase raises the floor of the ADC requiring higher signal levels before activation.) 0x00 = [23:0] Full Range (DEFAULT) 0x01 = [23:1] ▼ 0x14 = [23:20] 0x15 = [23:21]
		IMM_GEN_VOL																	**Signal Level Of The 23Hz Sinewave Generation For Impedance Measurement** 00 = 1/2 Full Scale 01 = 1/4 Full Scale (DEFAULT) 10 = 1/8 Full Scale 11 = 1/16 Full Scale
		IMM_CYCLE_CNT																	**Number Of MCLK** (Used to calculate the impedance) 00 = 1024 01 = 2048 (DEFAULT) 10 = 4096 11 = 8192
		IMM_MODE																	**Impedance Measurement Mode Enable Conrol** 0 = Disable (DEFAULT) 1 = Enable
		DACIN_SRC																	**DAC Filter Input Source Selection** (IMM_MODE enabled from built-in sine generator) 00 = From DRC DAC Output (DEFAULT) 01 = From DAC Mixer Output 10 = From u/A-law decode output 11 = None
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	**0x0000**
4D	IMM_RMS_L																		**Left Headset Speaker Impedance Readout** (It is recommended to characterize this before use with known Impedance values.)
		DEFAULT	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	**READ ONLY**
4E	FUSE_CTRL2	FUSEIN_L																	**The Lower 16 bits Of The FUSEIN** (These register bits are OR ed with the Fuse latches and can be used for test characterization except during reset or after power on reset.)
		DEFAULT	0	0	0	0	0	1	0	0	0	0	0	0	0	0	0	0	**0x0000**
4F	FUSE_CTRL3	RESERVED																	**RESERVED**
		FUSEIN_H																	**The Higher 2 Bits Of The FUSEIN** (These register bits are OR ed with the Fuse latches and can be used for test characterization except during reset or after power on reset.)
		DEFAULT	0	0	0	0	0	1	0	0	0	0	0	0	0	0	0	0	**0x0000**
51	FUSE_CTRL1	FUSE_PRG_MODE																	**Set eFuse mode** 0 = Not in the programming mode (DEFAULT) 1 = In the programming mode
		FUSWBNKIN																	**Set Bank of 32 eFuse Used For Programming The eFuse** 0 = Bank 0 (DEFAULT) 1 = Bank 1
		FUSEPRGBNK																	(Set this signal to 1 will instantly program the eFuse that selects the bank of 32 eFuses during a read operation.)
		FUSEPRGEN																	(Set this signal to 1 will program the selected eFuse.)
		RUSEREAD																	(Set this signal to 1 will read the bank of 32 eFuses selected by the fuse bank eFuse.)
		FUSERESETB																	(Set this signal to 0 will reset the 32 eFuse latches. This signal should be 1 after any read cycle.)
		FUSESEL																	(The eFuse address bus for programming. Only one bit can be programmed at a time.)
		DEFAULT	0	0	0	0	0	1	0	0	0	0	0	0	0	0	0	0	**0x0400**
53	OTPDOUT_1	OTPDOUT_L																	**OTP Read Out Data Low 16 Bits**
		DEFAULT	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	**READ ONLY**

REG	Function	Name	Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	Description
54	OTPDOUT_2	OTPDOUT_H																	OTP Read Out Data High 2 Bits
		DEFAULT	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	READ ONLY
55	MISC_CTRL	RAM_TEST_START																	Ram Test Enable Control 0 = Disable (DEFAULT) 1 = Enable
		D2A_LOOP																	ADC To DAC Loop 0 = Disable (DEFAULT) 1 = Enable (use ADC decimation filter output as DAC Left filter input)
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
58	I2C_DEVICE_ID	I2C_DEVICE_ID																	I2C Device ID Read In
		KEYDET																	Key Detect Status Bit
		MICDET																	MICDETECT Status Bit
		SILICON REVISION ID																	Silicon Revision Bits
		DEFAULT	X	0	0	1	1	0	1	X	0	0	1	0	0	0	0	0	READ ONLY
59	SARDOUT_RAM_STATUS	RATM_TEST_FINISH																	RAM Test Status Bit 0 = Test not finished 1 = Test finished
		RAM_TEST_FAIL																	RAM Test Result Bit 0 = Test passed 1 = Test failed
		ANALOG_MUTE																	Analog Mute Flag Bit 0 = Disable 1 = Enable
		DEFAULT	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	READ ONLY
5A	SOFTWARE_RST	SW_RESET																	Software Reset (Write any value ***twice*** to reset all internal states without resetting the config registers.)
66	BIAS_ADJ	TESTRL																	Headphone Impedance Enable Control (Test/ IMM_MODE) 0 = Disable (DEFAULT) 1 = Enable
		MUTEL																	Left PGA Mute Enable Control 0 = Disable (DEFAULT) 1 = Enable
		MUTER																	Right PGA Mute Enable Control 0 = Disable (DEFAULT) 1 = Enable
		TESTDAC																	DAC Right, Left Test Only
		RESERVED																	RESERVED
		VMIDEN																	VMID Enable Control 0 = Disable (DEFAULT) 1 = Enable
		VMIDSEL																	VMID Tie-off Impedance Select 00 = Open (DEFAULT) 10 = 125 KOhms 01 = 25 KOhms 11 = 2.5 KOhms
		RESERVED																	RESERVED
		RESERVED																	RESERVED
		BIASADJ																	PGA Master Bias Current Power Select 00 = Normal operation (DEFAULT) 01 = 9% reduced bias current from normal 10 = 17% reduced bias current from normal 11 = 11% increased bias current from normal
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0x0000
68	TRIM_SETTINGS	RESERVED																	RESERVED

REG	Function	Name	Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	Description
69	ANALOG_CONTROL_1	TESTDACIN	■	■															**DAC Test Signal** 00 & 11 = GND (DEFAULT) 01 & 10 = High & Low
		PULLUP_GPIO2								■									**GPIO2JD1 Pull Up Select** 0 = 1MOhm (DEFAULT) 1 = 100KOhm
		GPIO2THL									■	■							**GPIO2 JKDET1 Threshold Low Select** 00 = 0.22 x VDDA (DEFAULT) 10 = 0.40 x VDDA 11 = 0.5 x VDDA
		GPIO2THH											■	■					**GPIO2 JKDET1 Threshold High Select** 00 = 0.85 x VDDA (DEFAULT) 10 = 0.78 x VDDA 11 = 0.6 x VDDA
		RESERVED													■				**RESERVED**
		JD1POL														■			**JKDETL JD1 Polarity** 0 = Non-inverted (DEFAULT) 1 = Inverted
		JKDETLPOL															■		**JKDETL Output Polarity** 0 = Non-inverted (DEFAULT) 1 = Inverted
		ENJKDETL																■	**Enable Jack Tip Insertion Detection Circuit**
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	**0x0000**
6A	ANALOG_CONTROL_2	ANALOG_CONTROL				■													**Headphone Driver Class-AB Bias Current Adjust In Non-Class-G Mode** 0 = Normal (DEFAULT) 1 = 2x
							■												**Headphone Driver Bias Current Adjust In Class-G Mode** 0 = Normal (DEFAULT) 1 = 0.5x
								■											**Headphone Driver Bias Current Adjust In non Class-G Mode** 0 = Normal (DEFAULT) 1 = 2.5x
									■										**Headphone Out Boost Driver Bias Current Adjust in Class-G Mode 1** 0 = Normal (DEFAULT) 1 = Low
										■									**Headphone Out Boost Driver Bias Current Adjust in Class-G Mode 2** 0 = Normal (DEFAULT) 1 = Low
		HP_AB_ADJ									■								**Headphone Driver Bias Adjust In Class-AB** 0 = Normal (DEFAULT) 1 = Increase bias
		RESERVED										■	■	■	■	■			**RESERVED**
		CAPMSB															■		**DAC Reference Decoupling Capacitor Enable MSB**
		CAPLSB																■	**DAC Reference Decoupling Capacitor Enable LSB**
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	**0x0000**
6B		MUTENL											■						**MICLN Input to PGA Mute Enable Control** 0 = Disable (DEFAULT) 1 = Enable When enabling one of MICNL or MICPL mute, PGAL becomes a single-ended configuration, and PGAL gain becomes doubled. When both MICL and MICLP muted, the PGAL has no input. Same operaton as MUTENR and MUTEPR.
		MUTENR												■					**MICRN Input to PGA Mute Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		MUTEPL													■				**MICLP Input to PGA Mute Enable Control** 0 = Disable (DEFAULT) 1 = Enable

REG	Function	Name	Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	Description
		MUTEPR																	**MICRP Input to PGA Mute Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		RESERVED																	**RESERVED**
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	**0x0000**
71	ANALOG_ADC_1	RESERVED																	**RESERVED**
		PDMICDET																	**MIC Detection Power Down Control** 0 = Power on MIC detection (DEFAULT) 1 = Power down MIC detection
		DEFAULT	0	0	0	0	0	0	0	0	0	0	0	1	0	0	0	1	**0x0011**
72	ANALOG_ADC_2	RESERVED																	**RESERVED**
		ADC_UPL																	**Left Channel PGA Bias Current Increase Enable Control** (For driving ADC at high sample rates) 0 = Disable (DEFAULT) 1 = Enable
		ADC_UPR																	**Right Channel PGA Bias Current Increase Enable Control** (For driving ADC at high sample rates) 0 = Disable (DEFAULT) 1 = Enable
		BIAS																	**ADC Bias Current Select** 00 = Nominal (DEFAULT) 01 = Double 10 = Half 11 = Quarter
		VREFSEL																	**ADC VREF Select** 00 = Analog supply (DEFAULT) 01 = VMID 10 = VMID + 0.5dB 11 = VMID + 1dB
		RESERVED																	**RESERVED**
		PDNOTL																	**Left ADC Analog Power Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		PDNOTR																	**Right ADC Analog Power Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		DEFAULT	0	0	0	0	0	0	0	0	0	0	1	0	0	0	0	0	**0x0020**
73	RDAC	DAC_EN																	**DAC Enable Control** 0 = Disable (DEFAULT) 1 = Enable Bit0 = Left DAC Bit1 = Right DAC
		CLK_DAC_EN																	**DAC Clock Enable Control** 0 = Disable (DEFAULT) 1 = Enable Bit0 = Left DAC Bit1 = Right DAC
		FC_CTR																	**DAC Smoothing Filter On HS Output Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		CLK_DAC_DELAY																	**DAC Clock Delay Select** 000 = Delay 0 nsec (DEFAULT) 100 = Delay 4 nsec 001 = Delay 1 nsec 101 = Delay -3 nsec 010 = Delay 2 nsec (Recommended) 110 = Delay -2 nsec 011 = Delay 3 nsec 111 = Delay -1 nsec

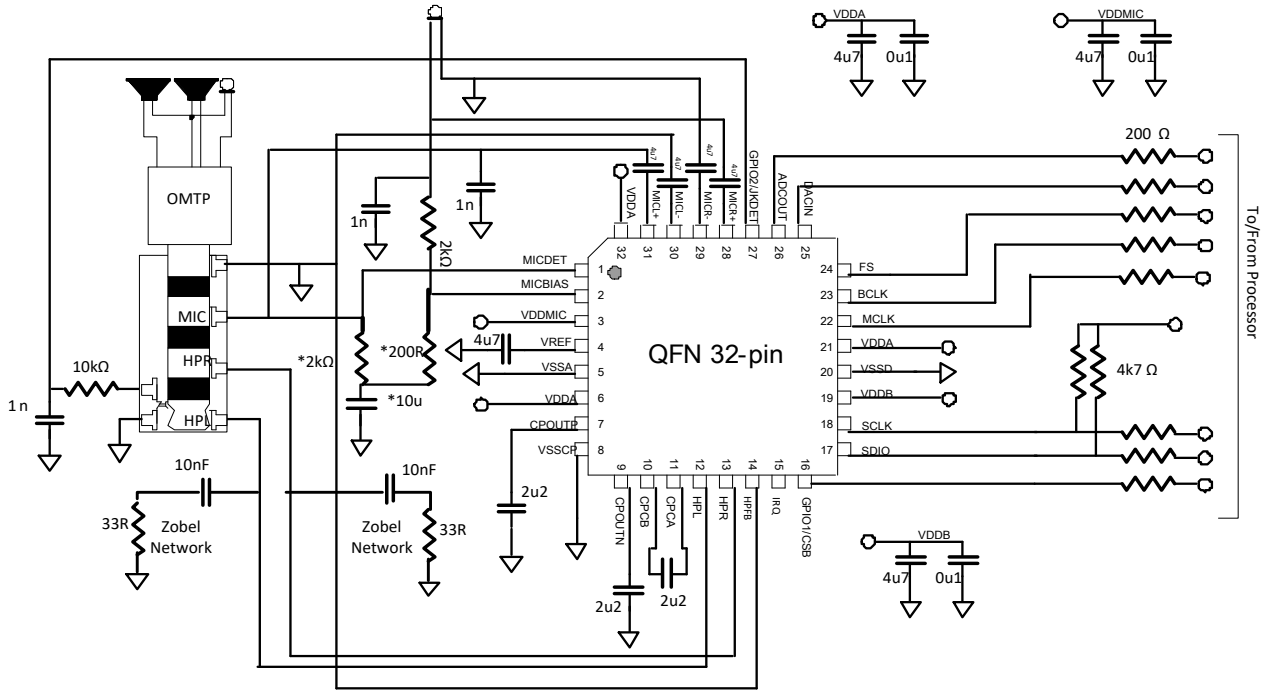
REG	Function	Name	Bit																Description
			15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	
		DACVREFSEL																	**DAC Full Scale Reference Voltage Select** (By setting this value, it will change DAC full scale output. For best performance, use default value.) 00 = External VDDA 01 = 1.5V 10 = 1.6V (DEFAULT) 11 = 1.7V
		DEFAULT	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**1**	**0**	**0**	**0**	**0x0008**
74	MIC_BIAS	INT2KA																	**MICBIAS1 Internal 2K Ohm Resistor For MCGND Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		LOWNOISE																	**Low Power / Low Noise Mode Select** 0 = Low power mode (DEFAULT) 1 = Low noise mode
		POWERUP																	**MICBIAS1 Power Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		MICBIASLVL1																	**MICBIAS1 Output Level Select** 000 = VDDA 001 = 1x 010 = 1.1x 011 = 1.2x 100 = 1.3x 101 = 1.4x 110 = 1.53x 111 = 1.53x
		DEFAULT	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**1**	**1**	**0**	**0x0006**
76	BOOST	CLR_APR_EMRGENCY_SHTDWN																	**Clear Headset Short Circuit Shutdown IRQ** 0 = (DEFAULT) 1 = Reset (Momentary)
		STG2_SEL																	**PGA In Class-A Mode Of Operation Enable Instead Of Class-AB Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		PDVMDFST																	**VMID Pre-charge Disable Control** 0 = Disable (DEFAULT) 1 = Enable
		BIASEN																	**Global Analog Bias Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		DISCHRG																	**Charge Input Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		BYPS_IBCTR																	**Bypass PGA Current Control Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		BOOSTDIS																	**HP Boost Driver Disable Control** 0 = Enable (DEFAULT) 1 = Disable
		BOOSTGDIS																	**HP Boost Driver In Class-G Mode Disable Control** 0 = Enable (DEFAULT) 1 = Disable
		SHRT_SHTDWN_DIG_EN																	**Short Circuit Shut Down Digital Part Enable Control** 0 = Disable (DEFAULT) 1 = Enable

REG	Function	Name	Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	Description
		EN_SHRT_SHTDWN																	**Automatic Short-circuit Shutdown Enable Control** 0 = (Driver shuts down after 16.3 µsec debounce when shortage detected. IRQ pin Interrupt is generated. When SHRT_ SHTDWN_ DIG_EN = 0, APR_EMRGNCY_SHTDWN is cleared if the IRQ pin Interrupt cleared. Users need to clear IRQ pin interrupt. When SHRT_ SHTDWN_ DIG_EN = 1, APR_EMRGNCY_SHTDWN is cleared 1630 µsec after shortage removed. Users need to clear IRQ pin interrupt.) 1 = (Headset driver power will be down immediately when shortage detected. No interrupt will be generated.)
		HS_SHRT_THRESHLD																	**Headset Short Circuit Protection Limit** 00= 115mA at +FS (DEFAULT) 11= 155mA at +FS
		PAMP_THRSHLD																	**Adjust HS Boost P-driver Bias Current** 00 = Normal (DEFAULT) 11 = Decrease current
		NAMP_THRSHLD																	**Adjust HS Boost N-driver Bias Current** 00 = Normal (DEFAULT) 11 = Decrease current
		DEFAULT	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0x0000**
77	FEPGA	ACDC_CTRL																	**Charge Input To VREF Enable Control** (Effective when DISCHRG = 1) 0 = Disable (DEFAULT) 1 = Enable Bit0 = Charges MICP to VREF Bit1 = Charges MICN to VREF
		CMLCK_ADJ																	**PGA Common Mode Threshold Lock Adjust** 00 = (DEFAULT)
		IB_LOOP_CTR																	**PGA Current Trim** 0 = (DEFAULT)
		IBCTR_CODE																	**PGA Current Trim** 000 = (DEFAULT)
		FEPGA_MODEL																	**Left PGA Mode Select** 0 = Disable (DEFAULT) 1 = Enable MODE[0] = Anti-aliasing filter adjust MODE[1] = Disconnects MICP & MICN MODE[2] = No function MODE[3] = Shorts the inputs and terminates with 12kOhm differentially
		FEPGA_MODER																	**Right PGA Mode Select** 0 = Disable (DEFAULT) 1 = Enable MODE[0] = Anti-aliasing filter adjust MODE[1] = Disconnects MICP & MICN MODE[2] = No function MODE[3] = Shorts the inputs and terminates with 12kOhm differentially
		DEFAULT	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0x0000**
7E	PGA_GAIN	PGA_GAINL																	**Left PGA Gain Control** (Step size is 1dB.) 0x00 = -1dB (DEFAULT) 0x01 = 0dB ▼ 0x24 = 35dB 0x25 = 36dB

REG	Function	Name	Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	Description
		PGA_GAINR																	**Right PGA Gain Control** (Step size is 1dB.) 0x00 = -1dB (DEFAULT) 0x01 = 0dB ▼ 0x24 = 35dB 0x25 = 36dB
		DEFAULT	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0x0000**
7F	POWER_UP_CONTROL	PUPL																	**Left PGA Power Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		PUPR																	**Right PGA Power Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		PUP_INTEG																	**Output Integrator Power Enable Control** 0 = Disable (DEFAULT) 1 = Enable Bit0 = Left HP driver Bit1 = Right HP driver
		PUP_DRV_INSTG																	**Output Driver Power Enable Control** (To reduce pop noise, turn on this ***first***, then turn on PUP_MAIN_DRV) 0 = Disable (DEFAULT) 1 = Enable Bit0 = Left HP driver Bit1 = Right HP driver
		PUP_MAIN_DRV																	**Main Driver Power Enable Control** 0 = Disable (DEFAULT) 1 = Enable Bit0 = Left HP driver Bit1 = Right HP driver
		DEFAULT	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0x0000**
80	CHARGE_PUMP_AND_POWER_DOWN_CONTROL	RESERVED																	**RESERVED**
		BCLK_DS																	**BCLK IO Drive Strength Control** 0 = Normal (DEFAULT) 1 = Stronger
		FS_DS																	**FS IO Drive Strength Control** 0 = Normal (DEFAULT) 1 = Stronger
		ADCDAT_DS																	**ADCDAT IO Drive Strength Control** 0 = Normal (DEFAULT) 1 = Stronger
		SDA_DS																	**SDA IO Drive Strength Control** 0 = Normal 1 = Stronger (DEFAULT)
		JAMNODCLW																	**RESERVED**
		PDB_DAC																	**DAC Right / Left Power Down Bar Enable Control** 00 = Disable 11 = Enable (DEFAULT)
		JAMFORCE2																	**Register Output Force 1 Control** (Charge pump clock to fast) 0 = Disable (DEFAULT) 1 = Enable
		JAMFORCE1																	**Register Output Force 2 Control** (Charge pump clock to fast) 0 = Disable (DEFAULT) 1 = Enable
		RNIN																	**Charge Pump Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		PRECHARGE																	**VPOS Pre-charge Enable Control** (For faster startup) 0 = Disable (DEFAULT) 1 = Enable

REG	Function	Name	Bit																Description
			15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0	
		DISCHARGEVEE																	**VEE Pad Discharge Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		DISCHARGEVPOS																	**VPOS Pad Discharge Enable Control** 0 = Disable (DEFAULT) 1 = Enable
		SHCIRSEL2																	**Charge Up Current Limit 2** 0 = Low (DEFAULT) 1 = High
		SHCIRSEL1																	**Charge Up Current Limit 1** 0 = Low (DEFAULT) 1 = High
		DEFAULT	**0**	**0**	**0**	**0**	**1**	**0**	**1**	**1**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0x0B00**
81	**CHARGE_PUMP_INPUT_READ**	**APR_EMRGNCY_SHTDWN**																	**APR Emergency Short Circuit Shutdown IRQ**
		MODE1BUF																	**Monitor MODE1 State Of Charge Pump Block**
		NODCBUF																	**Monitor Charge Pump Drawing DC Current** 0 = Drawing 1 = Not drawing (DEFAULT)
		RN2BUF																	**Monitor Charge Pump Enable Status** 0 = Off (DEFAULT) 1 = On
		VPOSOK																	**Monitor High Voltage Status Of VPOS** 0 = Possible short circuit (DEFAULT) 1 = Max output (Normal operation)
		VCOMPBUF																	**Monitor Low Voltage & Low Current Status Of Charge Pump** 0 = No current 1 = With current (DEFAULT)
		FORCE1BUF																	**Monitor Charge Pump Frequency Status** 0 = Normal 1 = Max frequency (DEFAULT)
		DEFAULT	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**READ ONLY 0x0013**
82	**GENERAL_STATUS**	**JK_EJECT_INTR**																	**JACK Ejection Interrupt**
		JK_INSERT_INTR																	**JACK Insertion Interrupt**
		JKDET_ON																	**Pre-debounce JACK Status**
		JKDETL																	**JKDETL**
		FUSEBNKOUT																	**Fuse Bank Select Output**
		GPIO2_IN																	**GPIO2 Input**
		GPIO1_IN																	**GPIO1 Input**
		DEFAULT	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**X**	**READ ONLY 0x0020**

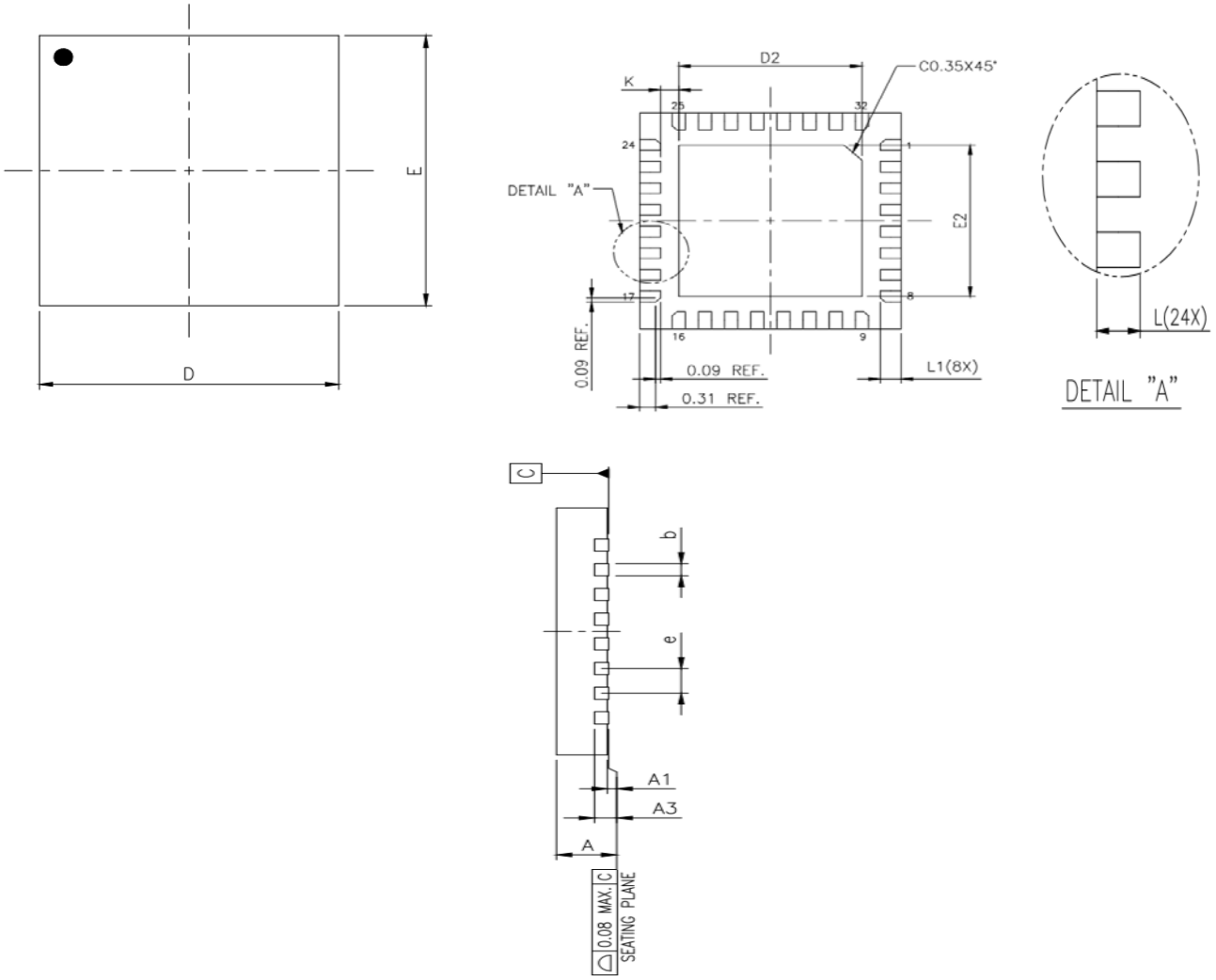
11. Typical Application Diagram



Note: * indicates optional components for improved noise reduction (refer to section 3.5)

12. Package Information

32-lead plastic QFN 32L; 5X5mm2, 0.8mm thickness, 0.5mm lead pitch
(Saw Type) EP SIZE 3.5X3.5 mm



JEDEC OUTLINE	MO-220		
PKG CODE	WQFN(X532)		
SYMBOLS	MIN.	NOM.	MAX.
A	0.70	0.75	0.80
A1	0.00	0.02	0.05
A3	0.203 REF.		
b	0.18	0.25	0.30
D	4.90	5.00	5.10
E	4.90	5.00	5.10
e	0.50 BSC		
L	0.35	0.40	0.45
L1	0.33	0.40	0.43
K	0.20	—	—

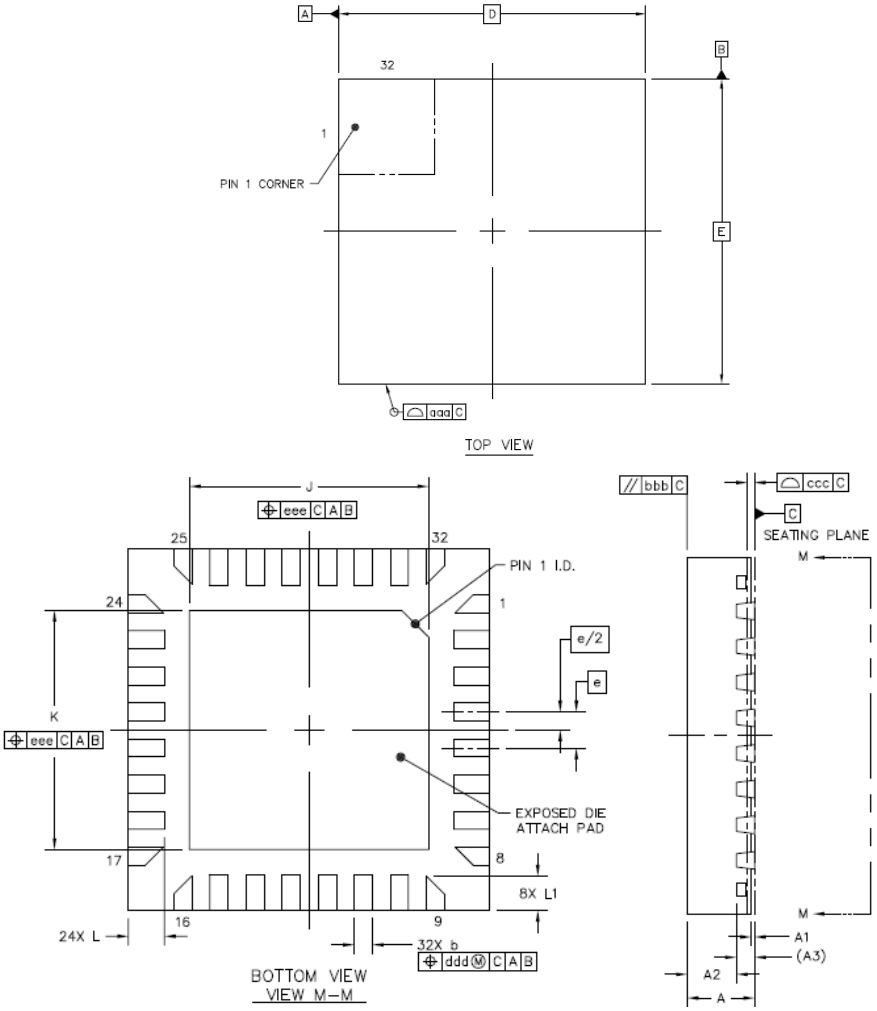
PAD SIZE	D2			E2			LEAD FINISH		JEDEC CODE
	MIN.	NOM.	MAX.	MIN.	NOM.	MAX.	Pure Tin	PPF	
150X15* MIL	3.45	3.50	3.55	3.45	3.50	3.55	V	X	W(V)HHD-5

"*"表示汎用字元,此汎用字元可能被其它不同字元所取代,實際的字元請參照bonding diagram所示.
"*" is an universal character, which means maybe replaced by specific character, the actual character please refers to the bonding diagram.

NOTES :

1. ALL DIMENSIONS ARE IN MILLIMETERS.
2. DIMENSION b APPLIES TO METALLIZED TERMINAL AND IS MEASURED BETWEEN 0.15mm AND 0.30mm FROM THE TERMINAL TIP. IF THE TERMINAL HAS THE OPTIONAL RADIUS ON THE OTHER END OF THE TERMINAL, THE DIMENSION b SHOULD NOT BE MEASURED IN THAT RADIUS AREA.
3. BILATERAL COPLANARITY ZONE APPLIES TO THE EXPOSED HEAT SINK SLUG AS WELL AS THE TERMINALS.

32-lead plastic QFN 32L; 4X4mm2, 0.8mm(Max) thickness, 0.4mm lead pitch
(Saw Type) EP SIZE 3.5X3.5 mm



		SYMBOL	MIN	NOM	MAX
TOTAL THICKNESS		A	0.7	0.75	0.8
STAND OFF		A1	0	0.035	0.05
MOLD THICKNESS		A2	---	0.55	0.57
L/F THICKNESS		A3		0.203 REF	
LEAD WIDTH		b	0.15	0.2	0.25
BODY SIZE	X	D		4 BSC	
	Y	E		4 BSC	
LEAD PITCH		e		0.4 BSC	
EP SIZE	X	J	2.55	2.65	2.75
	Y	K	2.55	2.65	2.75
LEAD LENGTH		L	0.35	0.4	0.45
		L1	0.332	0.382	0.432
PACKAGE EDGE TOLERANCE		aaa		0.1	
MOLD FLATNESS		bbb		0.1	
COPLANARITY		ccc		0.08	
LEAD OFFSET		ddd		0.1	
EXPOSED PAD OFFSET		eee		0.1	

13. ORDERING INFORMATION

Part Number	Dimension	Package	Package Material
NAU88L21YG	5x5 mm	QFN-32	Green
NAU88L21IG	4x4 mm	QFN-32	Green

NAU88L21 _ _

Package Material:

G = Green Package (Halogen-free, RoHS-compliant) & TSCA-compliant

Package Type:

Y = QFN-32

I = QFN-32

14. REVISION HISTORY

REVISION	DATE	DESCRIPTION
1.0	Feb 18, 2019	Initial Release
1.1	Mar 8, 2019	Add Cap-free and internal Resistor in MICBIAS
1.2	Jun 12, 2019	Add Zebol Network in Application circuit.
1.3	Sep 22, 2019	Added RC for MICDET – noise coupling.
1.4	Oct 17, 2019	Modified Figure 42:2-Wire Read Sequence.
1.5	Nov 8, 2019	Add QFN4x4mm2 IC package
1.6	Jan 17, 2020	Enhance FLL application note
1.7	Feb 24, 2020	Changed VDDC to VDDA Updated headphone performance MIPS400/500 informatin added Register 0x6[11:10] Device ID Reg0x58[5:2]=0x1823
1.8	Apl 5, 2020	Pin 21 VDDA pin description change VDDA ISD change Headset standby mode current consumption changed HeadPHone offset voltage change ADC SNR Fs change Vih change for VDDA Register 0x58 changed Register setting for DAC OSR cases Register 0x2C DAC OSR description Enrich FLL register description
1.9	Jun 4, 2020	Whole register map updated
2.0	Nov 2, 2020	Digital Audio Interface timing digrams
2.1	Dec 8, 2020	FEPGA input path enriched
2.2	Jan 20, 2021	Remove Reg55 SPI description Add I2C initial configuration content
2.3	Jan 22, 2021	Reg58 device ID bits adjustment Removal OSR=500
2.4	Jun 11, 2021	Adding descrption for DACVREF 0x72[8:9] Adding descrption for DACVREF 0x73[3:2]
2.5	Aug 3, 2021	Sec 2.2 Power up and start sequence Figure 7 ADC digital path Figure 12 DAC digital path Update 5x5 mm^2 pakcage information Update ADC DRC limitation
2.6	Aug 20, 2021	Update Reg0x31[8]
2.7	Sep 27, 2021	Update Reg0x68 Update Reg0x13[11:8] Update Register Table Format
2.8	Feb 28, 2022	Add MIC and button detection threshold conditions Update Reg0x0F[9:0] description Update Reg0x12[9:0] description
2.9	Nov 4, 2022	Upate Figure 2 with 0x6B register addition 0x6B register descritpion
3.0	Dec 22, 2022	Update ADC DAC parameter table

		Update Halogen-free, RoHS-compliant and TSCA-compliant description
3.1	Feb 1, 2023	Update REG0x37, REG0x3B

IMPORTANT NOTICE

Nuvoton Products are neither intended nor warranted for usage in systems or equipment, any malfunction or failure of which may cause loss of human life, bodily injury or severe property damage. Such applications are deemed, "Insecure Usage".

Insecure usage includes, but is not limited to: equipment for surgical implementation, atomic energy control instruments, airplane or spaceship instruments, the control or operation of dynamic, brake or safety systems designed for vehicular use, traffic signal instruments, all types of safety devices, and other applications intended to support or sustain life.

All Insecure Usage shall be made at customer's risk, and in the event that third parties lay claims to Nuvoton as a result of customer's Insecure Usage, customer shall indemnify the damages and liabilities thus incurred by Nuvoton.