**DESCRIPTION**

The WM8974 is a low power, high quality mono CODEC designed for portable applications such as Digital Still Camera or Digital Voice Recorder.

The device integrates support for a differential or single ended mic, and includes drivers for speakers or headphone, and mono line output. External component requirements are reduced as no separate microphone or headphone amplifiers are required.

Advanced Sigma Delta Converters are used along with digital decimation and interpolation filters to give high quality audio at sample rates from 8 to 48ks/s. Additional digital filtering options are available in the ADC path, to cater for application filtering such as 'wind noise reduction', plus an advanced mixed signal ALC function with noise gate is provided. The digital audio interface supports A-law and µ-law companding.

An on-chip PLL is provided to generate the required Master Clock from an external reference clock. The PLL clock can also be output if required elsewhere in the system.

The WM8974 operates at supply voltages from 2.5 to 3.6V, although the digital supplies can operate at voltages down to 1.71V to save power. The speaker and mono outputs use a separate supply of up to 5V which enables increased output power if required. Different sections of the chip can also be powered down under software control by way of the selectable two or three wire control interface.

**FEATURES**

**Mono CODEC:**
- Audio sample rates: 8, 11.025, 16, 22.05, 24, 32, 44.1, 48kHz
- DAC SNR 98dB, THD -84dB ('A'-weighted @ 8 – 48ks/s)
- ADC SNR 94dB, THD -83dB ('A'-weighted @ 8 – 48ks/s)
- On-chip Headphone/Speaker Driver with 'cap-less' connect
  - 40mW output power into 16Ω / 3.3V SPKVDD
  - BTL speaker drive 0.9W into 8Ω / 5V SPKVDD
- Additional MONO Line output
- Multiple analogue or 'Aux' inputs, plus analogue bypass path

**Mic Preamps:**
- Differential or single end Microphone Interface
  - Programmable preamp gain
  - Psuedo-differential inputs with common mode rejection
  - Programmable ALC / Noise Gate in ADC path
- Low-noise bias supplied for electret microphones

**OTHER FEATURES**
- 5 band EQ (record or playback path)
- Digital Playback Limiter
- Programmable ADC High-Pass Filter (wind noise reduction)
- Programmable ADC Notch Filter
- On-chip PLL
- Low power, low voltage
  - 2.5V to 3.6V (digital: 1.71V to 3.6V)
  - power consumption <10mA all-on 48ks/s mode
- 4x4x0.9mm 24 lead QFN package

**APPLICATIONS**
- Digital Still Camera Audio CODEC
- Wireless VoIP and other communication handsets / headsets
- Portable audio recorder
- General Purpose low power audio CODEC
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PIN CONFIGURATION

WM8974
(Top View)

ORDERING INFORMATION

<table>
<thead>
<tr>
<th>ORDER CODE</th>
<th>TEMPERATURE RANGE</th>
<th>PACKAGE</th>
<th>MOISTURE SENSITIVITY LEVEL</th>
<th>PACKAGE BODY TEMPERATURE</th>
</tr>
</thead>
<tbody>
<tr>
<td>WM8974CGEFL/V</td>
<td>-40°C to +85°C</td>
<td>24-lead QFN (4x4x0.9mm) (Pb-free)</td>
<td>MSL3</td>
<td>260°C</td>
</tr>
<tr>
<td>WM8974CGEFL/RV</td>
<td>-40°C to +85°C</td>
<td>24-lead QFN (4x4x0.9mm) (Pb-free, tape and reel)</td>
<td>MSL3</td>
<td>260°C</td>
</tr>
</tbody>
</table>

Note:
Reel Quantity = 3,500
## PIN DESCRIPTION

<table>
<thead>
<tr>
<th>PIN NO</th>
<th>NAME</th>
<th>TYPE</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>MICBIAS</td>
<td>Analogue Output</td>
<td>Microphone bias</td>
</tr>
<tr>
<td>2</td>
<td>AVDD</td>
<td>Supply</td>
<td>Analogue supply (feeds ADC and DAC)</td>
</tr>
<tr>
<td>3</td>
<td>AGND</td>
<td>Supply</td>
<td>Analogue ground (feeds ADC and DAC)</td>
</tr>
<tr>
<td>4</td>
<td>DCVDD</td>
<td>Supply</td>
<td>Digital core supply</td>
</tr>
<tr>
<td>5</td>
<td>DBVDD</td>
<td>Supply</td>
<td>Digital buffer (input/output) supply</td>
</tr>
<tr>
<td>6</td>
<td>DGND</td>
<td>Supply</td>
<td>Digital ground</td>
</tr>
<tr>
<td>7</td>
<td>ADCDAT</td>
<td>Digital Output</td>
<td>ADC digital audio data output</td>
</tr>
<tr>
<td>8</td>
<td>DACDAT</td>
<td>Digital Input</td>
<td>DAC digital audio data input</td>
</tr>
<tr>
<td>9</td>
<td>FRAME</td>
<td>Digital Input / Output</td>
<td>DAC and ADC sample rate clock or frame synch</td>
</tr>
<tr>
<td>10</td>
<td>BCLK</td>
<td>Digital Input / Output</td>
<td>Digital audio port clock</td>
</tr>
<tr>
<td>11</td>
<td>MCLK</td>
<td>Digital Input</td>
<td>Master clock input</td>
</tr>
<tr>
<td>12</td>
<td>CSB/GPIO</td>
<td>Digital Input / Output</td>
<td>3-Wire MPU chip select or general purpose input/output pin.</td>
</tr>
<tr>
<td>13</td>
<td>SCLK</td>
<td>Digital Input</td>
<td>3-Wire MPU clock Input / 2-Wire MPU Clock Input</td>
</tr>
<tr>
<td>14</td>
<td>SDIN</td>
<td>Digital Input / Output</td>
<td>3-Wire MPU data Input / 2-Wire MPU Data Input</td>
</tr>
<tr>
<td>15</td>
<td>MODE</td>
<td>Digital Input</td>
<td>Control interface mode selection pin.</td>
</tr>
<tr>
<td>16</td>
<td>MONOOUT</td>
<td>Analogue Output</td>
<td>Mono output</td>
</tr>
<tr>
<td>17</td>
<td>SPKOUTP</td>
<td>Analogue Output</td>
<td>Speaker output positive</td>
</tr>
<tr>
<td>18</td>
<td>SPKGNDD</td>
<td>Supply</td>
<td>Speaker ground (feeds speaker and mono output amps only)</td>
</tr>
<tr>
<td>19</td>
<td>SPKOUTN</td>
<td>Analogue Output</td>
<td>Speaker output Negative</td>
</tr>
<tr>
<td>20</td>
<td>SPKVDD</td>
<td>Supply</td>
<td>Speaker supply (feeds speaker and mono output amps only)</td>
</tr>
<tr>
<td>21</td>
<td>AUX</td>
<td>Analogue Input</td>
<td>Auxiliary analogue input</td>
</tr>
<tr>
<td>22</td>
<td>VMID</td>
<td>Reference</td>
<td>Decoupling for midrail reference voltage</td>
</tr>
<tr>
<td>23</td>
<td>MICN</td>
<td>Analogue Input</td>
<td>Microphone negative input</td>
</tr>
<tr>
<td>24</td>
<td>MICP</td>
<td>Analogue Input</td>
<td>Microphone positive input (common mode)</td>
</tr>
</tbody>
</table>

**Note:**

It is recommended that the QFN ground paddle should be connected to analogue ground on the application PCB.
ABSOLUTE MAXIMUM RATINGS

Absolute Maximum Ratings are stress ratings only. Permanent damage to the device may be caused by continuously operating at or beyond these limits. Device functional operating limits and guaranteed performance specifications are given under Electrical Characteristics at the test conditions specified.

ESD Sensitive Device. This device is manufactured on a CMOS process. It is therefore generically susceptible to damage from excessive static voltages. Proper ESD precautions must be taken during handling and storage of this device.

Cirrus Logic tests its package types according to IPC/JEDEC J-STD-020B for Moisture Sensitivity to determine acceptable storage conditions prior to surface mount assembly. These levels are:

- MSL1 = unlimited floor life at <30°C / 85% Relative Humidity. Not normally stored in moisture barrier bag.
- MSL2 = out of bag storage for 1 year at <30°C / 60% Relative Humidity. Supplied in moisture barrier bag.
- MSL3 = out of bag storage for 168 hours at <30°C / 60% Relative Humidity. Supplied in moisture barrier bag.

The Moisture Sensitivity Level for each package type is specified in Ordering Information.

<table>
<thead>
<tr>
<th>CONDITION</th>
<th>MIN</th>
<th>MAX</th>
</tr>
</thead>
<tbody>
<tr>
<td>DBVDD, DCVDD, AVDD supply voltages</td>
<td>-0.3V</td>
<td>+4.2V</td>
</tr>
<tr>
<td>SPKVDD supply voltage</td>
<td>-0.3V</td>
<td>+7V</td>
</tr>
<tr>
<td>Voltage range digital inputs</td>
<td>DGND -0.3V to DVDD +0.3V</td>
<td></td>
</tr>
<tr>
<td>Voltage range analogue inputs</td>
<td>AGND -0.3V to AVDD +0.3V</td>
<td></td>
</tr>
<tr>
<td>Operating temperature range, $T_A$</td>
<td>-40°C to +85°C</td>
<td></td>
</tr>
<tr>
<td>Storage temperature prior to soldering</td>
<td>30°C max / 85% RH max</td>
<td></td>
</tr>
<tr>
<td>Storage temperature after soldering</td>
<td>-65°C to +150°C</td>
<td></td>
</tr>
</tbody>
</table>

Notes:
1. Analogue and digital grounds must always be within 0.3V of each other.
2. All digital and analogue supplies are completely independent from each other.

RECOMMENDED OPERATING CONDITIONS

<table>
<thead>
<tr>
<th>PARAMETER</th>
<th>SYMBOL</th>
<th>MIN</th>
<th>TYP</th>
<th>MAX</th>
<th>UNIT</th>
</tr>
</thead>
<tbody>
<tr>
<td>Digital supply range (Core)</td>
<td>DCVDD</td>
<td>1.71</td>
<td>3.6</td>
<td>V</td>
<td></td>
</tr>
<tr>
<td>Digital supply range (Buffer)</td>
<td>DBVDD</td>
<td>1.71</td>
<td>3.6</td>
<td>V</td>
<td></td>
</tr>
<tr>
<td>Analogue supplies range</td>
<td>AVDD</td>
<td>2.5</td>
<td>3.6</td>
<td>V</td>
<td></td>
</tr>
<tr>
<td>Speaker supply</td>
<td>SPKVDD</td>
<td>2.5</td>
<td>5.5</td>
<td>V</td>
<td></td>
</tr>
<tr>
<td>Ground</td>
<td>DGND, AGND, SPKGND</td>
<td>0</td>
<td></td>
<td>V</td>
<td></td>
</tr>
</tbody>
</table>

Notes:
1. When using PLL, DCVDD must be 1.9V or higher.
2. AVDD must be ≥ DCVDD.
3. DBVDD must be ≥ DCVDD.
4. In non-boosted mode, SPKVDD must be ≥ AVDD, if boosted SPKVDD must be ≥ 1.5x AVDD.
5. When using PLL, DCVDD must be ≥ 1.9V.
ELECTRICAL CHARACTERISTICS

Test Conditions
DCVDD = 1.8V, AVDD = DBVDD = 3.3V, SPKVDD = 3.3V, T\textsubscript{A} = +25\textdegree C, 1kHz signal, fs = 48kHz, 24-bit audio data unless otherwise stated.

<table>
<thead>
<tr>
<th>PARAMETER</th>
<th>SYMBOL</th>
<th>TEST CONDITIONS</th>
<th>MIN</th>
<th>TYP</th>
<th>MAX</th>
<th>UNIT</th>
</tr>
</thead>
<tbody>
<tr>
<td>Microphone Inputs (MICN, MICP)</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Full-scale Input Signal Level (Note 1) – note this changes with AVDD</td>
<td>( V_{\text{INFS}} )</td>
<td>PGABOOST = 0dB, INPPGAVOL = 0dB</td>
<td>1.0</td>
<td>0</td>
<td></td>
<td>Vrms dBV</td>
</tr>
<tr>
<td>Mic PGA equivalent input noise</td>
<td>At 35.25dB gain</td>
<td></td>
<td>150</td>
<td></td>
<td></td>
<td>uV</td>
</tr>
<tr>
<td>Input resistance</td>
<td>( R_{\text{MICN}} )</td>
<td>Gain set to 35.25dB</td>
<td>1.6</td>
<td></td>
<td></td>
<td>k\Omega</td>
</tr>
<tr>
<td>Input resistance</td>
<td>( R_{\text{MICN}} )</td>
<td>Gain set to 0dB</td>
<td>47</td>
<td></td>
<td></td>
<td>k\Omega</td>
</tr>
<tr>
<td>Input resistance</td>
<td>( R_{\text{MICN}} )</td>
<td>Gain set to -12dB</td>
<td>75</td>
<td></td>
<td></td>
<td>k\Omega</td>
</tr>
<tr>
<td>Input resistance</td>
<td>( R_{\text{MICP}} )</td>
<td>MICP2INPPGA = 1</td>
<td>94</td>
<td></td>
<td></td>
<td>k\Omega</td>
</tr>
<tr>
<td>Input resistance</td>
<td>( R_{\text{MICP}} )</td>
<td>MICP2INPPGA = 0</td>
<td>94</td>
<td></td>
<td></td>
<td>k\Omega</td>
</tr>
<tr>
<td>Input Capacitance</td>
<td>( C_{\text{MICN}} )</td>
<td></td>
<td>10</td>
<td></td>
<td></td>
<td>pF</td>
</tr>
</tbody>
</table>

MIC Input Programmable Gain Amplifier (PGA)

<table>
<thead>
<tr>
<th>PARAMETER</th>
<th>SYMBOL</th>
<th>TEST CONDITIONS</th>
<th>MIN</th>
<th>TYP</th>
<th>MAX</th>
<th>UNIT</th>
</tr>
</thead>
<tbody>
<tr>
<td>Programmable Gain</td>
<td></td>
<td></td>
<td>-12</td>
<td>35.25</td>
<td>dB</td>
<td></td>
</tr>
<tr>
<td>Programmable Gain Step Size</td>
<td></td>
<td></td>
<td>Guaranteed monotonic</td>
<td>0.75</td>
<td></td>
<td>dB</td>
</tr>
<tr>
<td>Mute Attenuation</td>
<td></td>
<td></td>
<td></td>
<td>108</td>
<td></td>
<td>dB</td>
</tr>
</tbody>
</table>

Selectlable Input Gain Boost (0/+20dB)

<table>
<thead>
<tr>
<th>PARAMETER</th>
<th>SYMBOL</th>
<th>TEST CONDITIONS</th>
<th>MIN</th>
<th>TYP</th>
<th>MAX</th>
<th>UNIT</th>
</tr>
</thead>
<tbody>
<tr>
<td>Gain Boost</td>
<td></td>
<td></td>
<td>0</td>
<td>20</td>
<td></td>
<td>dB</td>
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</tbody>
</table>

Automatic Level Control (ALC)/Limiter – ADC only

<table>
<thead>
<tr>
<th>PARAMETER</th>
<th>SYMBOL</th>
<th>TEST CONDITIONS</th>
<th>MIN</th>
<th>TYP</th>
<th>MAX</th>
<th>UNIT</th>
</tr>
</thead>
<tbody>
<tr>
<td>Target Record Level</td>
<td></td>
<td></td>
<td>-28.5</td>
<td>-6</td>
<td></td>
<td>dB</td>
</tr>
<tr>
<td>Programmable Gain</td>
<td></td>
<td></td>
<td>-12</td>
<td>35.25</td>
<td>dB</td>
<td></td>
</tr>
<tr>
<td>Programmable Gain Step Size</td>
<td></td>
<td></td>
<td>Guaranteed Monotonic</td>
<td>0.75</td>
<td></td>
<td>dB</td>
</tr>
<tr>
<td>Gain Hold Time (Note 2)</td>
<td>( t_{\text{HOLD}} )</td>
<td>MCLK=12.288MHz (Note 4)</td>
<td>0, 2.67, 5.33, 10.67, … , 43691 (time doubles with each step)</td>
<td></td>
<td></td>
<td>ms</td>
</tr>
<tr>
<td>Gain Ramp-Up (Decay) Time (Note 3)</td>
<td>( t_{\text{DCY}} )</td>
<td>ALCMODE=0 (ALC), MCLK=12.288MHz (Note 4)</td>
<td>3.3, 6.6, 13.1, … , 3360 (time doubles with each step)</td>
<td></td>
<td></td>
<td>ms</td>
</tr>
<tr>
<td></td>
<td></td>
<td>ALCMODE=1 (limiter), MCLK=12.288MHz (Note 4)</td>
<td>0.73, 1.45, 2.91, … , 744 (time doubles with each step)</td>
<td></td>
<td></td>
<td>ms</td>
</tr>
<tr>
<td>Gain Ramp-Down (Attack) Time (Note 3)</td>
<td>( t_{\text{ATK}} )</td>
<td>ALCMODE=0 (ALC), MCLK=12.288MHz (Note 4)</td>
<td>0.83, 1.66, 3.33, … , 852 (time doubles with each step)</td>
<td></td>
<td></td>
<td>ms</td>
</tr>
<tr>
<td></td>
<td></td>
<td>ALCMODE=1 (limiter), MCLK=12.288MHz (Note 4)</td>
<td>0.18, 0.36, 0.73, … , 186 (time doubles with each step)</td>
<td></td>
<td></td>
<td>ms</td>
</tr>
</tbody>
</table>

Analogue to Digital Converter (ADC)

<table>
<thead>
<tr>
<th>PARAMETER</th>
<th>SYMBOL</th>
<th>TEST CONDITIONS</th>
<th>MIN</th>
<th>TYP</th>
<th>MAX</th>
<th>UNIT</th>
</tr>
</thead>
<tbody>
<tr>
<td>Signal to Noise Ratio (Note 5)</td>
<td>SNR</td>
<td>A-weighted, 0dB PGA gain</td>
<td>85</td>
<td>94</td>
<td></td>
<td>dB</td>
</tr>
<tr>
<td>Total Harmonic Distortion (Note 6)</td>
<td>THD</td>
<td>-1dBFS input, 0dB PGA gain</td>
<td>-75</td>
<td>-83</td>
<td></td>
<td>dB</td>
</tr>
</tbody>
</table>

Auxiliary Analogue Input (AUX)

<table>
<thead>
<tr>
<th>PARAMETER</th>
<th>SYMBOL</th>
<th>TEST CONDITIONS</th>
<th>MIN</th>
<th>TYP</th>
<th>MAX</th>
<th>UNIT</th>
</tr>
</thead>
<tbody>
<tr>
<td>Full-scale Input Signal Level (0dB) – note this changes with AVDD</td>
<td>( V_{\text{INFS}} )</td>
<td></td>
<td>1.0</td>
<td></td>
<td></td>
<td>Vrms dBV</td>
</tr>
<tr>
<td>Input Resistance</td>
<td>( R_{\text{AUXIN}} )</td>
<td>AUXMODE=0</td>
<td>20</td>
<td></td>
<td></td>
<td>k\Omega</td>
</tr>
<tr>
<td>Input Capacitance</td>
<td>( C_{\text{AUXIN}} )</td>
<td></td>
<td>10</td>
<td></td>
<td></td>
<td>pF</td>
</tr>
</tbody>
</table>
Test Conditions
DCVDD = 1.8V, AVDD = DBVDD = 3.3V, SPKVDD = 3.3V, TA = +25°C, 1kHz signal, fs = 48kHz, 24-bit audio data unless otherwise stated.

<table>
<thead>
<tr>
<th>PARAMETER</th>
<th>SYMBOL</th>
<th>TEST CONDITIONS</th>
<th>MIN</th>
<th>TYP</th>
<th>MAX</th>
<th>UNIT</th>
</tr>
</thead>
<tbody>
<tr>
<td>Digital to Analogue Converter (DAC) to MONO output</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Signal to Noise Ratio (Note 5)</td>
<td>SNR</td>
<td>A-weighted</td>
<td>90</td>
<td>98</td>
<td></td>
<td>dB</td>
</tr>
<tr>
<td>Total Harmonic Distortion + Noise (Note 6)</td>
<td>THD+N</td>
<td>RL = 10 kΩ full-scale signal</td>
<td>-84</td>
<td></td>
<td></td>
<td>dB</td>
</tr>
<tr>
<td>0dB Full Scale output voltage (Note 9)</td>
<td></td>
<td>MONOBOOST=0</td>
<td>AVDD / 3.3</td>
<td></td>
<td></td>
<td>VRMS</td>
</tr>
<tr>
<td></td>
<td></td>
<td>MONOBOOST=1</td>
<td>1.5 x (AVDD/3.3)</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Speaker Output PGA

| Programmable Gain | | -57 | 6 | | dB |
| Programmable Gain Step Size | Guaranteed monotonic | | 1 | | dB |

BTL Speaker Output (SPKOUTP, SPKOUTN with 8Ω bridge tied load)

<table>
<thead>
<tr>
<th>Output Power</th>
<th>PO</th>
<th>Output power is very closely correlated with THD; see below</th>
</tr>
</thead>
<tbody>
<tr>
<td>Total Harmonic Distortion + Noise (Note 6)</td>
<td>THD+N</td>
<td>PO =180mW, RL = 8Ω, SPKVDD=3.3V</td>
</tr>
<tr>
<td></td>
<td></td>
<td>PO =400mW, RL = 8Ω, SPKVDD=3.3V</td>
</tr>
<tr>
<td></td>
<td></td>
<td>PO =360mW, RL = 8Ω, SPKVDD=5V</td>
</tr>
<tr>
<td></td>
<td></td>
<td>PO =800mW, RL = 8Ω, SPKVDD=5V</td>
</tr>
<tr>
<td>Signal to Noise Ratio</td>
<td>SNR</td>
<td>SPKVDD=3.3V, RL = 8Ω</td>
</tr>
<tr>
<td></td>
<td></td>
<td>SPKVDD=5V, RL = 8Ω</td>
</tr>
<tr>
<td>Power Supply Rejection Ratio</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

‘Headphone’ output (SPKOUTP, SPKOUTN with resistive load to ground)

| Signal to Noise Ratio | SNR | | 100 | | dB |
| Total Harmonic Distortion + Noise (Note 6) | THD+N | Po=20mW, RL = 16Ω, SPKVDD=3.3V | 0.02 | -74 | | dB |
| | | Po=20mW, RL = 32Ω, SPKVDD=3.3V | 0.017 | -75 | | dB |

Microphone Bias

| Bias Voltage (MBVSEL=0) | V_MICBIAS | 0.9 x AVDD | V |
| Bias Voltage (MBVSEL=1) | V_MICBIAS | 0.75 x AVDD | V |
| Bias Current Source | I_MICBIAS | 3 | mA |
| Output Noise Voltage | Vn | 1K to 20kHz | 15 | nV/√Hz |

Digital Input / Output

| Input HIGH Level | V_H | 0.7 x DVDD | V |
| Input LOW Level | V_L | 0.3 x DVDD | V |
| Output HIGH Level | V_OH | I_OH=1mA | 0.9 x DVDD | V |
| Output LOW Level | V_DL | I_DL=1mA | 0.1 x DVDD | V |
TERMINOLOGY

1. MICN input only in single ended microphone configuration. Maximum input signal to MICP without distortion is -3dBV.

2. Hold Time is the length of time between a signal detected being too quiet and beginning to ramp up the gain. It does not apply to ramping down the gain when the signal is too loud, which happens without a delay.

3. Ramp-up and Ramp-Down times are defined as the time it takes for the PGA to change its gain by 6dB.

4. All hold, ramp-up and ramp-down times scale proportionally with MCLK.

5. Signal-to-noise ratio (dB) – SNR is a measure of the difference in level between the full scale output and the output with no signal applied. (No Auto-zero or Automute function is employed in achieving these results).

6. THD+N (dB) – THD+N is a ratio, of the rms values, of (Noise + Distortion)/Signal.

7. The maximum output voltage can be limited by the speaker power supply. If MONOBOOST=1 then SPKVDD should be 1.5xAVDD or higher to prevent clipping taking place in the output stage.
SIGNAL TIMING REQUIREMENTS

SYSTEM CLOCK TIMING

![Figure 1 System Clock Timing Requirements](image)

Test Conditions
DCVDD=1.8V, DBVDD=AVDD=SPKVDD=3.3V, DGND=AGND=SPKGND=0V, TA = +25°C

<table>
<thead>
<tr>
<th>PARAMETER</th>
<th>SYMBOL</th>
<th>CONDITIONS</th>
<th>MIN</th>
<th>TYP</th>
<th>MAX</th>
<th>UNIT</th>
</tr>
</thead>
<tbody>
<tr>
<td>System Clock Timing Information</td>
<td>MCLK cycle time</td>
<td>T_{MCLKY}</td>
<td>MCLK as direct SYSCLK source (CLKSEL=0)</td>
<td>81.38</td>
<td>ns</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>MCLK as input to PLL (see note) (CLKSEL=1)</td>
<td>20</td>
<td>ns</td>
<td></td>
</tr>
<tr>
<td>MCLK duty cycle</td>
<td>T_{MCLKDS}</td>
<td>60:40</td>
<td>40:60</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Note: PLL pre-scaling and PLL N and K values should be set appropriately so that SYSCLK is no greater than 12.288MHz.

AUDIO INTERFACE TIMING – MASTER MODE

![Figure 2 Digital Audio Data Timing – Master Mode](image)

Test Conditions
DCVDD=1.8V, DBVDD=AVDD=SPKVDD=3.3V, DGND=AGND=SPKGND=0V, TA=+25°C, Master Mode, fs=48kHz, MCLK=256fs, 24-bit data, unless otherwise stated.

<table>
<thead>
<tr>
<th>PARAMETER</th>
<th>SYMBOL</th>
<th>MIN</th>
<th>TYP</th>
<th>MAX</th>
<th>UNIT</th>
</tr>
</thead>
<tbody>
<tr>
<td>Audio Data Input Timing Information</td>
<td>FRAME propagation delay from BCLK falling edge</td>
<td>t_{DL}</td>
<td>10</td>
<td>ns</td>
<td></td>
</tr>
<tr>
<td></td>
<td>ADCDAT propagation delay from BCLK falling edge</td>
<td>t_{DDA}</td>
<td>10</td>
<td>ns</td>
<td></td>
</tr>
<tr>
<td></td>
<td>DACDAT setup time to BCLK rising edge</td>
<td>t_{DST}</td>
<td>10</td>
<td>ns</td>
<td></td>
</tr>
<tr>
<td></td>
<td>DACDAT hold time from BCLK rising edge</td>
<td>t_{DHT}</td>
<td>10</td>
<td>ns</td>
<td></td>
</tr>
</tbody>
</table>

Note: BCLK period should always be greater than MCLK period.
Audio Interface Timing – Slave Mode

![Waveform Diagram]

**Figure 3** Digital Audio Data Timing – Slave Mode

**Test Conditions**
DCVDD = 1.8 V, DBVDD = AVDD = SPKVDD = 3.3 V, DGND = AGND = SPKGND = 0 V, $T_a = +25^\circ$C, Slave Mode, $f_s = 48$ kHz, MCLK = 256fs, 24-bit data, unless otherwise stated.

<table>
<thead>
<tr>
<th>PARAMETER</th>
<th>SYMBOL</th>
<th>MIN</th>
<th>TYP</th>
<th>MAX</th>
<th>UNIT</th>
</tr>
</thead>
<tbody>
<tr>
<td>BCLK cycle time</td>
<td>$t_{BCY}$</td>
<td>160</td>
<td></td>
<td>ns</td>
<td></td>
</tr>
<tr>
<td>BCLK pulse width high</td>
<td>$t_{BCH}$</td>
<td>64</td>
<td></td>
<td>ns</td>
<td></td>
</tr>
<tr>
<td>BCLK pulse width low</td>
<td>$t_{BCL}$</td>
<td>64</td>
<td></td>
<td>ns</td>
<td></td>
</tr>
<tr>
<td>FRAME set-up time to BCLK rising edge</td>
<td>$t_{RBU}$</td>
<td>10</td>
<td></td>
<td>ns</td>
<td></td>
</tr>
<tr>
<td>FRAME hold time from BCLK rising edge</td>
<td>$t_{RH}$</td>
<td>10</td>
<td></td>
<td>ns</td>
<td></td>
</tr>
<tr>
<td>DACDAT hold time from BCLK rising edge</td>
<td>$t_{DS}$</td>
<td>10</td>
<td></td>
<td>ns</td>
<td></td>
</tr>
<tr>
<td>DACDAT set-up time to BCLK rising edge</td>
<td>$t_{DS}$</td>
<td>10</td>
<td></td>
<td>ns</td>
<td></td>
</tr>
<tr>
<td>ADCDAT propagation delay from BCLK falling edge</td>
<td>$t_{DD}$</td>
<td>20</td>
<td></td>
<td>ns</td>
<td></td>
</tr>
</tbody>
</table>
## Control Interface Timing – 3-Wire Mode

**Test Conditions**
DCVDD = 1.8V, DBVDD = AVDD = SPKVDD = 3.3V, DGND = AGND = SPKGND = 0V, T_A = +25°C, Slave Mode, fs = 48kHz, MCLK = 256fs, 24-bit data, unless otherwise stated.

<table>
<thead>
<tr>
<th>PARAMETER</th>
<th>SYMBOL</th>
<th>MIN</th>
<th>TYP</th>
<th>MAX</th>
<th>UNIT</th>
</tr>
</thead>
<tbody>
<tr>
<td>Program Register Input Information</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>SCLK rising edge to CSB rising edge</td>
<td>t_{CSB}</td>
<td>80</td>
<td></td>
<td></td>
<td>ns</td>
</tr>
<tr>
<td>SCLK pulse cycle time</td>
<td>t_{SCY}</td>
<td>200</td>
<td></td>
<td></td>
<td>ns</td>
</tr>
<tr>
<td>SCLK pulse width low</td>
<td>t_{SCL}</td>
<td>80</td>
<td></td>
<td></td>
<td>ns</td>
</tr>
<tr>
<td>SCLK pulse width high</td>
<td>t_{SCH}</td>
<td>80</td>
<td></td>
<td></td>
<td>ns</td>
</tr>
<tr>
<td>SDIN to SCLK set-up time</td>
<td>t_{DSU}</td>
<td>40</td>
<td></td>
<td></td>
<td>ns</td>
</tr>
<tr>
<td>SCLK to SDIN hold time</td>
<td>t_{DSO}</td>
<td>40</td>
<td></td>
<td></td>
<td>ns</td>
</tr>
<tr>
<td>CSB pulse width low</td>
<td>t_{CSL}</td>
<td>40</td>
<td></td>
<td></td>
<td>ns</td>
</tr>
<tr>
<td>CSB pulse width high</td>
<td>t_{CSH}</td>
<td>40</td>
<td></td>
<td></td>
<td>ns</td>
</tr>
<tr>
<td>CSB rising to SCLK rising</td>
<td>t_{CSS}</td>
<td>40</td>
<td></td>
<td></td>
<td>ns</td>
</tr>
<tr>
<td>Pulse width of spikes that will be suppressed</td>
<td>t_{ps}</td>
<td>0</td>
<td></td>
<td>5</td>
<td>ns</td>
</tr>
</tbody>
</table>

![Diagram of Control Interface Timing – 3-Wire Serial Control Mode](image)
CONTROL INTERFACE TIMING – 2-WIRE MODE

Test Conditions
DCVDD=1.8V, DBVDD=AVDD=SPKVDD=3.3V, DGND=AGND=SPKGND=0V, TA = +25°C, Slave Mode, f_s = 48kHz, MCLK = 256fs, 24-bit data, unless otherwise stated.

<table>
<thead>
<tr>
<th>PARAMETER</th>
<th>SYMBOL</th>
<th>MIN</th>
<th>TYP</th>
<th>MAX</th>
<th>UNIT</th>
</tr>
</thead>
<tbody>
<tr>
<td>Program Register Input Information</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>SCLK Frequency</td>
<td></td>
<td>0</td>
<td>526</td>
<td>kHz</td>
<td></td>
</tr>
<tr>
<td>SCLK Low Pulse-Width</td>
<td>t_1</td>
<td>1.3</td>
<td></td>
<td>ns</td>
<td></td>
</tr>
<tr>
<td>SCLK High Pulse-Width</td>
<td>t_2</td>
<td>600</td>
<td></td>
<td>ns</td>
<td></td>
</tr>
<tr>
<td>Hold Time (Start Condition)</td>
<td>t_3</td>
<td>600</td>
<td></td>
<td>ns</td>
<td></td>
</tr>
<tr>
<td>Setup Time (Start Condition)</td>
<td>t_4</td>
<td>600</td>
<td></td>
<td>ns</td>
<td></td>
</tr>
<tr>
<td>Data Setup Time</td>
<td>t_5</td>
<td>100</td>
<td></td>
<td>ns</td>
<td></td>
</tr>
<tr>
<td>SDIN, SCLK Rise Time</td>
<td>t_6</td>
<td>300</td>
<td></td>
<td>ns</td>
<td></td>
</tr>
<tr>
<td>SDIN, SCLK Fall Time</td>
<td>t_7</td>
<td>300</td>
<td></td>
<td>ns</td>
<td></td>
</tr>
<tr>
<td>Setup Time (Stop Condition)</td>
<td>t_8</td>
<td>600</td>
<td></td>
<td>ns</td>
<td></td>
</tr>
<tr>
<td>Data Hold Time</td>
<td>t_9</td>
<td>900</td>
<td></td>
<td>ns</td>
<td></td>
</tr>
<tr>
<td>Pulse width of spikes that will be suppressed</td>
<td>t_ps</td>
<td>0</td>
<td>5</td>
<td>ns</td>
<td></td>
</tr>
</tbody>
</table>
INTRODUCTION

The WM8974 is a low power audio codec combining a high quality mono audio DAC and ADC, with flexible line and microphone input and output processing. Applications for this device include digital still cameras with mono audio, record and playback capability, voice recorders, wireless VoIP headsets and games console accessories.

The chip offers great flexibility in use, and so can support many different modes of operation as follows:

MICROPHONE INPUTS

Two microphone inputs are provided, allowing for either a differential microphone input or a single ended microphone to be connected. These inputs have a user programmable gain range of -12dB to +35.25dB using internal resistors. After the input PGA stage comes a boost stage which can add a further 20dB of gain. A microphone bias is output from the chip which can be used to bias the microphones. The signal routing can be configured to allow manual adjustment of mic levels, or to allow the ALC loop to control the level of mic signal that is transmitted.

Total gain through the microphone paths of up to +55.25dB can be selected.

PGA AND ALC OPERATION

A programmable gain amplifier is provided in the input path to the ADC. This may be used manually or in conjunction with a mixed analogue/digital automatic level control (ALC) which keeps the recording volume constant.

AUX INPUT

The device includes a mono input, AUX, that can be used as an input for warning tones (beep) etc. The output from this circuit can be summed into the mono output and/or the speaker output paths, so allowing for mixing of audio with ‘backing music’ etc., as required. This path can also be summed into the input in a flexible fashion, either to the input PGA as a second microphone input or as a line input. The configuration of this circuit, with integrated on-chip resistors allows several analogue signals to be summed into the single AUX input if required.

ADC

The mono ADC uses a multi-bit high-order oversampling architecture to deliver optimum performance with low power consumption. Various sample rates are supported, from the 8ks/s rate typically used in voice dictation, up to the 48ks/s rate used in high quality audio applications.

HI-FI DAC

The hi-fi DAC provides high quality audio playback suitable for all portable mono audio type applications.

DIGITAL FILTERING

Advanced Sigma Delta Converters are used along with digital decimation and interpolation filters to give high quality audio at sample rates from 8ks/s to 48ks/s.

Application specific digital filters are also available which help to reduce the effect of specific noise sources such as ‘wind noise’. The filters include a programmable ADC high-pass filter, a programmable ADC notch filter and a 5-band equaliser that can be applied to either the ADC or the DAC path in order to improve the overall audio sound from the device.

OUTPUT MIXING AND VOLUME ADJUST

Flexible mixing is provided on the outputs of the device; a mixer is provided for the speaker outputs, and an additional mono summer for the mono output. These mixers allow the output of the DAC, the output of the ADC volume control and the Auxiliary input to be combined. The output volume can be adjusted using the integrated digital volume control and there is additional analogue gain adjustment capability on the speaker output.

AUDIO INTERFACES

The WM8974 has a standard audio interface, to support the transmission of audio data to and from the chip. This interface is a 4 wire standard audio interface which supports a number of audio data
formats including I²S, DSP Mode, MSB-First, left justified and MSB-First, right justified, and can operate in master or slave modes.

CONTROL INTERFACES

To allow full software control over all its features, the WM8974 offers a choice of 2 or 3 wire MPU control interface. It is fully compatible and an ideal partner for a wide range of industry standard microprocessors, controllers and DSPs. The selection between 2-wire mode and 3-wire mode is determined by the state of the MODE pin. If MODE is high then 3-wire control mode is selected, if MODE is low then 2-wire control mode is selected.

In 2 wire mode, only slave operation is supported, and the address of the device is fixed as 0011010.

CLOCKING SCHEMES

WM8974 offers the normal audio DAC clocking scheme operation, where 256fs MCLK is provided to the DAC/ADC. However, a PLL is also included which may be used to generate the internal master clock frequency in the event that this is not available from the system controller. The PLL uses an input reference (typically, the 12MHz USB clock) to generate high quality audio clocks. If the PLL is not required for generation of these clocks, it can be reconfigured to generate alternative clocks which may then be output on the CSB/GPIO pin and used elsewhere in the system.

POWER CONTROL

The design of the WM8974 has given much attention to power consumption without compromising performance. It operates at low supply voltages, and includes the facility to power off any unused parts of the circuitry under software control, includes standby and power off modes.

INPUT SIGNAL PATH

The WM8974 has 3 flexible analogue inputs: two microphone inputs, and an auxiliary input. These inputs can be used in a variety of ways. The input signal path before the ADC has a flexible PGA block which then feeds into a gain boost/mixer stage.

MICROPHONE INPUTS

The WM8974 can accommodate a variety of microphone configurations including single ended and differential inputs. The inputs through the MICN, MICP and optionally AUX pins are amplified through the input PGA as shown in Figure 6.

A pseudo differential input is the preferential configuration where the positive terminal of the input PGA is connected to the MICP input pin by setting MICP2INPPGA=1. The microphone ground should then be connected to MICN (when MICN2INPPGA=1) or optionally to AUX (when AUX2INPPGA=1) input pins.

Alternatively a single ended microphone can be connected to the MICN input with MICN2INPPGA set to 1. The non-inverting terminal of the input PGA should be connected internally to VMID by setting MICP2INPPGA to 0.

In differential mode the larger signal should be input to MICP and the smaller (e.g. noisy ground connection) should be input to MICN.
Figure 6 Microphone Input PGA Circuit (switch positions shown are for differential mic input)

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R44</td>
<td>0</td>
<td>MICP2INPPGA</td>
<td>1</td>
<td>Connect input PGA amplifier positive terminal to MICP or VMID.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = input PGA amplifier positive terminal connected to VMID</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = input PGA amplifier positive terminal connected to MICP through variable resistor string</td>
</tr>
<tr>
<td></td>
<td>1</td>
<td>MICN2INPPGA</td>
<td>1</td>
<td>Connect MICN to input PGA negative terminal.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = MICN not connected to input PGA</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = MICN connected to input PGA amplifier negative terminal.</td>
</tr>
<tr>
<td></td>
<td>2</td>
<td>AUX2INPPGA</td>
<td>0</td>
<td>Select AUX amplifier output as input PGA signal source.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = AUX not connected to input PGA</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = AUX connected to input PGA amplifier negative terminal.</td>
</tr>
</tbody>
</table>

The input PGA is enabled by the IPPGAEN register bit.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R2</td>
<td>2</td>
<td>INPPGAEN</td>
<td>0</td>
<td>Input microphone PGA enable</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = disabled</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = enabled</td>
</tr>
</tbody>
</table>
INPUT PGA VOLUME CONTROL

The input microphone PGA has a gain range from -12dB to +35.25dB in 0.75dB steps. The gain from the MICN input to the PGA output and from the AUX amplifier to the PGA output are always common and controlled by the register bits INPPGAVOL[5:0]. These register bits also affect the MICP pin when MICP2INPPGA=1.

When the Automatic Level Control (ALC) is enabled the input PGA gain is then controlled automatically and the INPPGAVOL bits should not be used.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R45</td>
<td>5:0</td>
<td>INPPGAVOL</td>
<td>010000</td>
<td>Input PGA volume</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>000000 = -12dB</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>000001 = -11.25dB</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>010000 = 0dB</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>111111 = 35.25dB</td>
</tr>
<tr>
<td></td>
<td>6</td>
<td>INPPGAMUTE</td>
<td>0</td>
<td>Mute control for input PGA:</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0=Input PGA not muted, normal operation</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1=Input PGA muted (and disconnected from</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>the following input BOOST stage).</td>
</tr>
<tr>
<td></td>
<td>7</td>
<td>INPPGAZC</td>
<td>0</td>
<td>Input PGA zero cross enable:</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0=Update gain when gain register changes</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1=Update gain on 1st zero cross after gain</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>register write.</td>
</tr>
</tbody>
</table>

Table 1 Input PGA Volume Control

AUXILIARY INPUT

An auxiliary input circuit (Figure 7) is provided which consists of an amplifier which can be configured either as an inverting buffer for a single input signal or as a mixer/summer for multiple inputs with the use of external resistors. The circuit is enabled by the register bit AUXEN.

![Figure 7 Auxiliary Input Circuit](image)

The AUXMODE register bit controls the auxiliary input mode of operation:

In buffer mode (AUXMODE=0) the switch labelled AUXSW in Figure 7 is open and the signal at the AUX pin will be buffered and inverted through the aux circuit using only the internal components.
In mixer mode (AUXMODE=1) the on-chip input resistor is bypassed, this allows the user to sum in multiple inputs with the use of external resistors. When used in this mode there will be gain variations through this path from part to part due to the variation of the internal 20kΩ resistors relative to the higher tolerance external resistors.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R1</td>
<td>6</td>
<td>AUXEN</td>
<td>0</td>
<td>Auxiliary input buffer enable</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = OFF</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = ON</td>
</tr>
<tr>
<td>R44</td>
<td>3</td>
<td>AUXMODE</td>
<td>0</td>
<td>0 = inverting buffer</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = mixer (on-chip input resistor bypassed)</td>
</tr>
</tbody>
</table>

Table 2 Auxiliary Input Buffer Control

INPUT BOOST

The input BOOST circuit has 3 selectable inputs: the input microphone PGA output, the AUX amplifier output and the MICP input pin (when not using a differential microphone configuration). These three inputs can be mixed together and have individual gain boost/adjust as shown in Figure 8.

![Input Boost Stage Diagram](image)

Figure 8 Input Boost Stage

The input PGA path can have a +20dB boost (PGABOOST=1) a 0dB pass through (PGABOOST=0) or be completely isolated from the input boost circuit (INPPGAMUTE=1).

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R45</td>
<td>6</td>
<td>INPPGAMUTE</td>
<td>0</td>
<td>Mute control for input PGA:</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0=Input PGA not muted, normal operation</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1=Input PGA muted (and disconnected from the following input BOOST stage).</td>
</tr>
<tr>
<td>R47</td>
<td>8</td>
<td>PGABOOST</td>
<td>0</td>
<td>0 = PGA output has +0dB gain through input BOOST stage.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = PGA output has +20dB gain through input BOOST stage.</td>
</tr>
</tbody>
</table>

Table 3 Input BOOST Stage Control
The Auxiliary amplifier path to the BOOST stage is controlled by the AUX2BOOSTVOL[2:0] register bits. When AUX2BOOSTVOL=000 this path is completely disconnected from the BOOST stage. Settings 001 through to 111 control the gain in 3dB steps from -12dB to +6dB.

The MICP path to the BOOST stage is controlled by the MICP2BOOSTVOL[2:0] register bits. When MICP2BOOSTVOL=000 this input pin is completely disconnected from the BOOST stage. Settings 001 through to 111 control the gain in 3dB steps from -12dB to +6dB.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R47</td>
<td>2:0</td>
<td>AUX2BOOSTVOL</td>
<td>000</td>
<td>Controls the auxiliary amplifier to the input boost stage: 000=Path disabled (disconnected) 001= -12dB gain through boost stage 010= -9dB gain through boost stage ... 111= +6dB gain through boost stage</td>
</tr>
<tr>
<td></td>
<td>6:4</td>
<td>MICP2BOOSTVOL</td>
<td>000</td>
<td>Controls the MICP pin to the input boost stage (NB, when using this path set MICP2INPPGA=0): 000=Path disabled (disconnected) 001= -12dB gain through boost stage 010= -9dB gain through boost stage ... 111= +6dB gain through boost stage</td>
</tr>
</tbody>
</table>

Table 4 Input BOOST Stage Control

The BOOST stage is enabled under control of the BOOSTEN register bit.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R2</td>
<td>4</td>
<td>BOOSTEN</td>
<td>0</td>
<td>Input BOOST enable</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = Boost stage OFF</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = Boost stage ON</td>
</tr>
</tbody>
</table>

Table 5 Input BOOST Enable Control

MICROPHONE BIASING CIRCUIT

The MICBIAS output provides a low noise reference voltage suitable for biasing electret type microphones and the associated external resistor biasing network. Refer to the Applications Information section for recommended external components. The MICBIAS voltage can be altered via the MBVSEL register bit. If MBVSEL = 0, the MICBIAS voltage is 0.9 × AVDD. If MBVSEL = 1, the MICBIAS voltage is 0.75 × AVDD. The output can be enabled or disabled using MICBEN.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R1</td>
<td>4</td>
<td>MICY</td>
<td>0</td>
<td>Microphone Bias Enable</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = OFF (high impedance output)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = ON</td>
</tr>
</tbody>
</table>

Table 6 Microphone Bias Enable

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R44</td>
<td>8</td>
<td>MBVSEL</td>
<td>0</td>
<td>Microphone Bias Voltage Control</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = 0.9 × AVDD</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = 0.75 × AVDD</td>
</tr>
</tbody>
</table>

Table 7 Microphone Bias Voltage Control
The internal MICBIAS circuitry is shown in Figure 9. Note that the maximum source current capability for MICBIAS is 3mA. The external biasing resistors therefore must be large enough to limit the MICBIAS current to 3mA.

![Microphone Bias Schematic](Image)

**Figure 9 Microphone Bias Schematic**

**ANALOGUE TO DIGITAL CONVERTER (ADC)**

The WM8974 uses a multi-bit, oversampled sigma-delta ADC channel. The use of multi-bit feedback and high oversampling rates reduces the effects of jitter and high frequency noise. The ADC Full Scale input level is proportional to AVDD. With a 3.3V supply voltage, the full scale level is $1.0\,\text{V}_{\text{rms}}$. Any voltage greater than full scale may overload the ADC and cause distortion.

**ADC DIGITAL FILTERS**

The ADC filters perform true 24 bit signal processing to convert the raw multi-bit oversampled data from the ADC to the correct sampling frequency to be output on the digital audio interface. The digital filter path is illustrated in Figure 10.

![ADC Digital Filter Path](Image)

**Figure 10 ADC Digital Filter Path**
The ADC is enabled by the ADCEN register bit.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R2</td>
<td>0</td>
<td>ADCEN</td>
<td>0</td>
<td>0 = ADC disabled</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = ADC enabled</td>
</tr>
</tbody>
</table>

Table 8 ADC Enable

The polarity of the output signal can also be changed under software control using the ADCPOL register bit. The oversampling rate of the ADC can be adjusted using the ADCOSR register bit. With ADCOSR=0 the oversample rate is 64x which gives lowest power operation and when ADCOSR=1 the oversample rate is 128x which gives best performance.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R14</td>
<td>3</td>
<td>ADCOSR</td>
<td>0</td>
<td>ADC oversample rate select:</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0=64x (lower power)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1=128x (best performance)</td>
</tr>
<tr>
<td>R14</td>
<td>0</td>
<td>ADCPOL</td>
<td>0</td>
<td>0=normal</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1=inverted</td>
</tr>
</tbody>
</table>

Table 9 ADC Oversample Rate Select

SELECTABLE HIGH-PASS FILTER

A selectable high-pass filter is provided. To disable this filter set HPFEN=0. The filter has two modes controlled by HPFAPP. In Audio Mode (HPFAPP=0) the filter is first order, with a cut-off frequency of 3.7Hz. In Application Mode (HPFAPP=1) the filter is second order, with a cut-off frequency selectable via the HPFCUT register. The cut-off frequencies when HPFAPP=1 are shown in Table 11.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R14</td>
<td>8</td>
<td>HPFEN</td>
<td>1</td>
<td>High-Pass Filter Enable</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0=disabled</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1=enabled</td>
</tr>
<tr>
<td>R14</td>
<td>7</td>
<td>HPFAPP</td>
<td>0</td>
<td>Select audio mode or application mode</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0=Audio mode (1st order, fc = ~3.7Hz)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1=Application mode (2nd order, fc = HPFCUT)</td>
</tr>
<tr>
<td>R14</td>
<td>6:4</td>
<td>HPFCUT</td>
<td>000</td>
<td>Application mode cut-off frequency</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>See Table 11 for details.</td>
</tr>
</tbody>
</table>

Table 10 ADC Filter Select
Note that the High-Pass filter values (when HPFAPP=1) work on the basis that the SR register bits are set correctly for the actual sample rate as shown in Table 11.

**PROGRAMMABLE NOTCH FILTER**

A programmable notch filter is provided. This filter has a variable centre frequency and bandwidth, programmable via two coefficients, a0 and a1. These coefficients should be converted to 2’s complement numbers to determine the register values. a0 and a1 are represented by the register bits NFA0[13:0] and NFA1[13:0]. Because these coefficient values require four register writes to setup there is an NFU (Notch Filter Update) flag which should be set only when all four registers are setup.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R27</td>
<td>6:0</td>
<td>NFA0[13:7]</td>
<td>0</td>
<td>Notch Filter a0 coefficient, bits [13:7]</td>
</tr>
<tr>
<td></td>
<td>7</td>
<td>NFEN</td>
<td>0</td>
<td>Notch filter enable: 0=Disabled 1=Enabled</td>
</tr>
<tr>
<td></td>
<td>8</td>
<td>NFU</td>
<td>0</td>
<td>Notch filter update. The notch filter values used internally only update when one of the NFU bits is set high.</td>
</tr>
<tr>
<td>R28</td>
<td>6:0</td>
<td>NFA0[6:0]</td>
<td>0</td>
<td>Notch Filter a0 coefficient, bits [6:0]</td>
</tr>
<tr>
<td></td>
<td>8</td>
<td>NFU</td>
<td>0</td>
<td>Notch filter update. The notch filter values used internally only update when one of the NFU bits is set high.</td>
</tr>
<tr>
<td></td>
<td>8</td>
<td>NFU</td>
<td>0</td>
<td>Notch filter update. The notch filter values used internally only update when one of the NFU bits is set high.</td>
</tr>
<tr>
<td>R30</td>
<td>6:0</td>
<td>NFA1[6:0]</td>
<td>0</td>
<td>Notch Filter a1 coefficient, bits [6:0]</td>
</tr>
<tr>
<td></td>
<td>8</td>
<td>NFU</td>
<td>0</td>
<td>Notch filter update. The notch filter values used internally only update when one of the NFU bits is set high.</td>
</tr>
</tbody>
</table>
The coefficients are calculated as follows:

\[
a_0 = \frac{1 - \tan(w_b / 2)}{1 + \tan(w_b / 2)}
\]

\[
a_1 = -(1 + a_0)\cos(w_b)
\]

Where:

\[
w_0 = 2\pi f_c / f_s
\]

\[
w_b = 2\pi f_b / f_s
\]

\[f_c = \text{centre frequency in Hz}, \quad f_b = \text{-3dB bandwidth in Hz}, \quad f_s = \text{sample frequency in Hz}\]

The coefficients are calculated as follows:

\[
NFA0 = -a0 \times 2^{13}
\]

\[
NFA1 = -a1 \times 2^{12}
\]

These values are then converted to 2's complement notation to determine the register values.

**NOTCH FILTER WORKED EXAMPLE**

The following example illustrates how to calculate the \(a0\) and \(a1\) coefficients for a desired centre frequency and -3dB bandwidth.

\(f_c = 1000\ \text{Hz}\)

\(f_b = 100\ \text{Hz}\)

\(f_s = 48000\ \text{Hz}\)

\[
w_0 = 2\pi f_c / f_s = 2\pi \times (1000 / 48000) = 0.1308996939\ \text{rads}
\]

\[
w_b = 2\pi f_b / f_s = 2\pi \times (100 / 48000) = 0.01308996939\ \text{rads}
\]

\[
a_0 = \frac{1 - \tan(w_b / 2)}{1 + \tan(w_b / 2)} = \frac{1 - \tan(0.0130899693 / 2)}{1 + \tan(0.0130899693 / 2)} = 0.9869949627
\]

\[
a_1 = -(1 + a_0)\cos(w_b) = -(1 + 0.9869949627)\cos(0.1308996939) = -1.969995945
\]

\(NFn_A0 = -a0 \times 213 = -8085\) (rounded to nearest whole number)

\(NFn_A1 = -a1 \times 212 = 8069\) (rounded to nearest whole number)

These values are then converted to 2's complement:

\(NFA0 = 14'h206B = 14'b10000001101011\)

\(NFA1 = 14'h1F85 = 14'b0111111000101\)
DIGITAL ADC VOLUME CONTROL

The output of the ADCs can be digitally attenuated over a range from –127dB to 0dB in 0.5dB steps. The gain for a given eight-bit code X is given by:

\[ \text{Gain} = 0.5 \times (x - 255) \text{ dB for } 1 \leq x \leq 255, \text{MUTE for } x = 0 \]

Table 13 ADC Volume

INPUT LIMITER / AUTOMATIC LEVEL CONTROL (ALC)

The WM8974 has an automatic PGA gain control circuit, which can function as an input peak limiter or as an automatic level control (ALC).

The Automatic Level Control (ALC) provides continuous adjustment of the input PGA in response to the amplitude of the input signal. A digital peak detector monitors the input signal amplitude and compares it to a register defined threshold level (ALCLVL).

If the signal is below the threshold, the ALC will increase the gain of the PGA at a rate set by ALCDCY. If the signal is above the threshold, the ALC will reduce the gain of the PGA at a rate set by ALCATK.

The ALC has two modes selected by the ALCMODE register: normal mode and peak limiter mode. The ALC/limiter function is enabled by setting the register bit R32[8] ALCSEL.
<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R33 (21h)</td>
<td>3:0</td>
<td>ALCLEVEL [3:0]</td>
<td>1011 (-12dB)</td>
<td>ALC target – sets level at input.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1111 = -6dBFS</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1110 = -7.5dBFS</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1101 = -9dBFS</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1100 = -10.5dBFS</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1011 = -12dBFS</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1010 = -13.5dBFS</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1001 = -15dBFS</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1000 = -16.5dBFS</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0111 = -18dBFS</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0110 = -19.5dBFS</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0101 = -21dBFS</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0100 = -22.5dBFS</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0011 = -24dBFS</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0010 = -25.5dBFS</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0001 = -27dBFS</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0000 = -28.5dBFS</td>
</tr>
<tr>
<td></td>
<td>8</td>
<td>ALCZC</td>
<td>0 (zero cross off)</td>
<td>ALC uses zero cross detection circuit.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = Disabled (recommended)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = Enabled</td>
</tr>
<tr>
<td></td>
<td>7:4</td>
<td>ALCLEVEL [3:0]</td>
<td>0000 (0ms)</td>
<td>ALC hold time before gain is increased.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0000 = 0ms</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0001 = 2.67ms</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0010 = 5.33ms</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0011 = 10.66ms</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0100 = 21.32ms</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0101 = 42.64ms</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0110 = 85.28ms</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0111 = 0.17s</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1000 = 0.34s</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1001 = 0.68s</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1010 or higher = 1.36s</td>
</tr>
<tr>
<td>R34 (22h)</td>
<td>8</td>
<td>ALCMODE</td>
<td>0</td>
<td>Determines the ALC mode of operation:</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = ALC mode (Normal Operation)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = Limiter mode.</td>
</tr>
<tr>
<td></td>
<td>7:4</td>
<td>ALCDCY [3:0]</td>
<td>0011 (26ms/6dB)</td>
<td>Decay (gain ramp-up) time (ALCMODE ==0)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Per step</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0000</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0001</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0010</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>… (time doubles with every step)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1010 or higher</td>
</tr>
<tr>
<td>REGISTER ADDRESS</td>
<td>BIT</td>
<td>LABEL</td>
<td>DEFAULT</td>
<td>DESCRIPTION</td>
</tr>
<tr>
<td>------------------</td>
<td>-----</td>
<td>-------</td>
<td>---------</td>
<td>-------------------------------</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>0011</td>
<td>Decay (gain ramp-up) time</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>(5.8ms/6dB)</td>
<td>(ALCMODE == 1)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Per step</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>0000</td>
<td>90.8us</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>0001</td>
<td>182us</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>0010</td>
<td>363us</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>...</td>
<td>(time doubles with every step)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>1010</td>
<td>93ms</td>
</tr>
<tr>
<td>3:0</td>
<td>ALCATK [3:0]</td>
<td>0010</td>
<td>0000</td>
<td>104us</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>0001</td>
<td>208us</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>0010</td>
<td>416us</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>...</td>
<td>(time doubles with every step)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>1010 or higher</td>
<td>106ms</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>0010</td>
<td>(726us/6dB)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Per step</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>0000</td>
<td>22.7us</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>0001</td>
<td>45.4us</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>0010</td>
<td>90.8us</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>...</td>
<td>(time doubles with every step)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>1010 or higher</td>
<td>23.2ms</td>
</tr>
</tbody>
</table>

Table 14  ALC Control Registers

When the ALC is disabled, the input PGA remains at the last controlled value of the ALC. An input gain update must be made by writing to the INPPGAVOLL/R register bits.
NORMAL MODE

In normal mode, the ALC will attempt to maintain a constant signal level by increasing or decreasing the gain of the PGA. The following diagram shows an example of this.

![Diagram showing ALC Normal Mode Operation with input signal, ALCSEL, PGA gain, and output signal with ALC level.](image)

Figure 11  ALC Normal Mode Operation
LIMITER MODE

In limiter mode, the ALC will reduce peaks that go above the threshold level, but will not increase the PGA gain beyond the starting level. The starting level is the PGA gain setting when the ALC is enabled in limiter mode. If the ALC is started in limiter mode, this is the gain setting of the PGA at start-up. If the ALC is switched into limiter mode after running in ALC mode, the starting gain will be the gain at switchover. The diagram below shows an example of limiter mode.

ATTACK AND DECAY TIMES

The attack and decay times set the update times for the PGA gain. The attack time is the time constant used when the gain is reducing. The decay time is the time constant used when the gain is increasing. In limiter mode, the time constants are faster than in ALC mode. The time constants are shown below in terms of a single gain step, a change of 6dB and a change of 90% of the PGAs gain range.

Note that, these times will vary slightly depending on the sample rate used (specified by the SR register).
### NORMAL MODE

**ALCMODE = 0 (Normal Mode)**

<table>
<thead>
<tr>
<th>ALCATK</th>
<th>$t_{ATK}$</th>
<th>$t_{ATK6dB}$</th>
<th>$t_{ATK90%}$</th>
</tr>
</thead>
<tbody>
<tr>
<td>0000</td>
<td>104µs</td>
<td>832µs</td>
<td>6ms</td>
</tr>
<tr>
<td>0001</td>
<td>208µs</td>
<td>1.66ms</td>
<td>12ms</td>
</tr>
<tr>
<td>0010</td>
<td>416µs</td>
<td>3.33ms</td>
<td>24ms</td>
</tr>
<tr>
<td>0011</td>
<td>832µs</td>
<td>6.66ms</td>
<td>48ms</td>
</tr>
<tr>
<td>0100</td>
<td>1.66ms</td>
<td>13.3ms</td>
<td>96ms</td>
</tr>
<tr>
<td>0101</td>
<td>3.33ms</td>
<td>26.6ms</td>
<td>192ms</td>
</tr>
<tr>
<td>0110</td>
<td>6.66ms</td>
<td>53.2ms</td>
<td>384ms</td>
</tr>
<tr>
<td>0111</td>
<td>13.3ms</td>
<td>106ms</td>
<td>767ms</td>
</tr>
<tr>
<td>1000</td>
<td>26.6ms</td>
<td>213.2ms</td>
<td>1.53s</td>
</tr>
<tr>
<td>1001</td>
<td>53.2ms</td>
<td>426ms</td>
<td>3.07s</td>
</tr>
<tr>
<td>1010</td>
<td>106ms</td>
<td>852ms</td>
<td>6.13s</td>
</tr>
</tbody>
</table>

### LIMITER MODE

**ALCMODE = 1 (Limiter Mode)**

<table>
<thead>
<tr>
<th>ALCATK</th>
<th>$t_{ATKLIM}$</th>
<th>$t_{ATKLIM6dB}$</th>
<th>$t_{ATKLIM90%}$</th>
</tr>
</thead>
<tbody>
<tr>
<td>0000</td>
<td>22.7µs</td>
<td>182µs</td>
<td>1.31ms</td>
</tr>
<tr>
<td>0001</td>
<td>45.4µS</td>
<td>363µS</td>
<td>2.62ms</td>
</tr>
<tr>
<td>0010</td>
<td>90.8µS</td>
<td>726µS</td>
<td>5.23ms</td>
</tr>
<tr>
<td>0011</td>
<td>182µS</td>
<td>1.45ms</td>
<td>10.5ms</td>
</tr>
<tr>
<td>0100</td>
<td>363µS</td>
<td>2.91ms</td>
<td>20.9ms</td>
</tr>
<tr>
<td>0101</td>
<td>726µS</td>
<td>5.81ms</td>
<td>41.8ms</td>
</tr>
<tr>
<td>0110</td>
<td>1.45ms</td>
<td>11.6ms</td>
<td>83.7ms</td>
</tr>
<tr>
<td>0111</td>
<td>2.9ms</td>
<td>23.2ms</td>
<td>167ms</td>
</tr>
<tr>
<td>1000</td>
<td>5.81ms</td>
<td>46.5ms</td>
<td>335ms</td>
</tr>
<tr>
<td>1001</td>
<td>11.6ms</td>
<td>93ms</td>
<td>669ms</td>
</tr>
<tr>
<td>1010</td>
<td>23.2ms</td>
<td>186ms</td>
<td>1.34s</td>
</tr>
</tbody>
</table>

Table 15  ALC Normal Mode (Attack and Decay times)
Table 16 ALC Limiter Mode (Attack and Decay times)

<table>
<thead>
<tr>
<th>ALCDCY</th>
<th>t\textsubscript{DCYLIM}</th>
<th>t\textsubscript{DCYLIM6dB}</th>
<th>t\textsubscript{DCYLIM90%}</th>
</tr>
</thead>
<tbody>
<tr>
<td>0000</td>
<td>90.8µs</td>
<td>726µs</td>
<td>5.23ms</td>
</tr>
<tr>
<td>0001</td>
<td>182µS</td>
<td>1.45ms</td>
<td>10.5ms</td>
</tr>
<tr>
<td>0010</td>
<td>363µS</td>
<td>2.91ms</td>
<td>20.9ms</td>
</tr>
<tr>
<td>0011</td>
<td>726µS</td>
<td>5.81ms</td>
<td>41.8ms</td>
</tr>
<tr>
<td>0100</td>
<td>1.45ms</td>
<td>11.6ms</td>
<td>83.7ms</td>
</tr>
<tr>
<td>0101</td>
<td>2.91ms</td>
<td>23.2ms</td>
<td>167ms</td>
</tr>
<tr>
<td>0110</td>
<td>5.81ms</td>
<td>46.5ms</td>
<td>335ms</td>
</tr>
<tr>
<td>0111</td>
<td>11.6ms</td>
<td>93ms</td>
<td>669ms</td>
</tr>
<tr>
<td>1000</td>
<td>23.2ms</td>
<td>186ms</td>
<td>1.34s</td>
</tr>
<tr>
<td>1001</td>
<td>46.5ms</td>
<td>372ms</td>
<td>2.68s</td>
</tr>
<tr>
<td>1010</td>
<td>93ms</td>
<td>744ms</td>
<td>5.36s</td>
</tr>
</tbody>
</table>

Minumum and Maximum Gain

The ALCMIN and ALCMAX register bits set the minimum/maximum gain value that the PGA can be set to whilst under the control of the ALC. This has no effect on the PGA when ALC is not enabled.

In normal mode, ALCMAX sets the maximum boost which can be applied to the signal. In limiter mode, ALCMAX will normally have no effect (assuming the starting gain value is less than the maximum gain specified by ALCMAX) because the maximum gain is set at the starting gain level.

ALCMIN sets the minimum gain value which can be applied to the signal.

\[
\text{PGA Gain} = 111111 \ (+35.25\text{dB})
\]

\[
\text{PGA Gain} = 000000 \ (-12\text{dB})
\]

Figure 13 ALC Min/Max Gain
<table>
<thead>
<tr>
<th>ALCMAX</th>
<th>Maximum Gain (dB)</th>
</tr>
</thead>
<tbody>
<tr>
<td>111</td>
<td>35.25</td>
</tr>
<tr>
<td>110</td>
<td>29.25</td>
</tr>
<tr>
<td>101</td>
<td>23.25</td>
</tr>
<tr>
<td>100</td>
<td>17.25</td>
</tr>
<tr>
<td>011</td>
<td>11.25</td>
</tr>
<tr>
<td>010</td>
<td>5.25</td>
</tr>
<tr>
<td>001</td>
<td>-0.75</td>
</tr>
<tr>
<td>000</td>
<td>-6.75</td>
</tr>
</tbody>
</table>

Table 18 ALC Max Gain Values

<table>
<thead>
<tr>
<th>ALCMIN</th>
<th>Minimum Gain (dB)</th>
</tr>
</thead>
<tbody>
<tr>
<td>000</td>
<td>-12</td>
</tr>
<tr>
<td>001</td>
<td>-6</td>
</tr>
<tr>
<td>010</td>
<td>0</td>
</tr>
<tr>
<td>011</td>
<td>6</td>
</tr>
<tr>
<td>100</td>
<td>12</td>
</tr>
<tr>
<td>101</td>
<td>18</td>
</tr>
<tr>
<td>110</td>
<td>24</td>
</tr>
<tr>
<td>111</td>
<td>30</td>
</tr>
</tbody>
</table>

Table 19 ALC Min Gain Values

Note that if the ALC gain setting strays outside the ALC operating range, either by starting the ALC outside of the range or changing the ALCMAX or ALCMIN settings during operation, the ALC will immediately adjust the gain to return to the ALC operating range. It is recommended that the ALC starting gain is set between the ALCMAX and ALCMIN limits.

**ALC HOLD TIME (NORMAL MODE ONLY)**

In Normal mode, the ALC has an adjustable hold time which sets a time delay before the ALC begins its decay phase (gain increasing). The hold time is set by the ALCHLD register.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R33</td>
<td>7:4</td>
<td>ALCHLD</td>
<td>0000</td>
<td>ALC hold time before gain is increased.</td>
</tr>
</tbody>
</table>

Table 20 ALC Hold Time

If the hold time is exceeded this indicates that the signal has reached a new average level and the ALC will increase the gain to adjust for that new average level. If the signal goes above the threshold during the hold period, the hold phase is abandoned and the ALC returns to normal operation.
Figure 14 ALCLVL
Figure 15  ALC Hold Time

<table>
<thead>
<tr>
<th>ALCHLD</th>
<th>t_HOLD (s)</th>
</tr>
</thead>
<tbody>
<tr>
<td>0000</td>
<td>0</td>
</tr>
<tr>
<td>0001</td>
<td>2.67ms</td>
</tr>
<tr>
<td>0010</td>
<td>5.34ms</td>
</tr>
<tr>
<td>0011</td>
<td>10.7ms</td>
</tr>
<tr>
<td>0100</td>
<td>21.4ms</td>
</tr>
<tr>
<td>0101</td>
<td>42.7ms</td>
</tr>
<tr>
<td>0110</td>
<td>85.4ms</td>
</tr>
<tr>
<td>0111</td>
<td>171ms</td>
</tr>
<tr>
<td>1000</td>
<td>342ms</td>
</tr>
<tr>
<td>1001</td>
<td>684ms</td>
</tr>
<tr>
<td>1010</td>
<td>1.37s</td>
</tr>
</tbody>
</table>

Table 21  ALC Hold Time Values
PEAK LIMITER

To prevent clipping when a large signal occurs just after a period of quiet, the ALC circuit includes a limiter function. If the ADC input signal exceeds 87.5% of full scale (−1.16dB), the PGA gain is ramped down at the maximum attack rate (as when ALCATK = 0000), until the signal level falls below 87.5% of full scale. This function is automatically enabled whenever the ALC is enabled.

Note: If ALCATK = 0000, then the limiter makes no difference to the operation of the ALC. It is designed to prevent clipping when long attack times are used.

NOISE GATE (NORMAL MODE ONLY)

When the signal is very quiet and consists mainly of noise, the ALC function may cause “noise pumping”, i.e. loud hissing noise during silence periods. The WM8974 has a noise gate function that prevents noise pumping by comparing the signal level at the input pins against a noise gate threshold, NGTH. The noise gate cuts in when:

\[
\text{Signal level at ADC [dBFS]} < \text{NGTH [dBFS]} + \text{PGA gain [dB]} + \text{Mic Boost gain [dB]}
\]

This is equivalent to:

\[
\text{Signal level at input pin [dBFS]} < \text{NGTH [dBFS]}
\]

The PGA gain is then held constant (preventing it from ramping up as it normally would when the signal is quiet).

The table below summarises the noise gate control register. The NGTH control bits set the noise gate threshold with respect to the ADC full-scale range. The threshold is adjusted in 6dB steps. Levels at the extremes of the range may cause inappropriate operation, so care should be taken with set-up of the function. The noise gate only operates in conjunction with the ALC and cannot be used in limiter mode.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
</table>
| R35 (23h) ALC Noise Gate Control | 2:0 | NGTH | 000 | Noise gate threshold:
| | | | 000 = -39dB |
| | | | 001 = -45dB |
| | | | 010 = -51dB |
| | | | 011 = -57dB |
| | | | 100 = -63dB |
| | | | 101 = -69dB |
| | | | 110 = -75dB |
| | | | 111 = -81dB |
| 3 | NGATEN | 0 | Noise gate function enable |
| | | | 1 = enable |
| | | | 0 = disable |

Table 22 ALC Noise Gate Control

The diagrams below show the response of the system to the same signal with and without noise gate.
Figure 16  ALC Operation above Noise Gate Threshold
OUTPUT SIGNAL PATH

The WM8974 output signal paths consist of digital application filters, up-sampling filters, a Hi-Fi DAC, analogue mixers, speaker and mono output drivers. The digital filters and DAC are enabled by bit DACEN. The mixers and output drivers can be separately enabled by individual control bits (see Analogue Outputs). Thus it is possible to utilise the analogue mixing and amplification provided by the WM8974, irrespective of whether the DACs are running or not.

The WM8974 DAC receives digital input data on the DACDAT pin. The digital filter block processes the data to provide the following functions:

- Digital volume control
- Graphic equaliser
- A digital peak limiter.
- Sigma-Delta Modulation

The high performance sigma-delta audio DAC converts the digital data into an analogue signal.
The analogue output from the DAC can then be mixed with the AUX analogue input and the ADC analogue input. The mix is fed to the output drivers, SPKOUTP/N, and MONOOUT.

MONOOUT: can drive a 16Ω or 32Ω headphone or line output or can be a buffered version of VMID (When MONOMUTE=1).

SPKOUTP/N: can drive a 16Ω or 32Ω stereo headphone or stereo line output, or an 8Ω BTL mono speaker.

**DIGITAL HI-FI DAC VOLUME CONTROL**

The signal volume from each hi-fi DAC can be controlled digitally. The gain and attenuation range is –127dB to 0dB in 0.5dB steps. The level of attenuation for an eight-bit code X is given by:

\[0.5 \times (X-255) \text{ dB for } 1 \leq X \leq 255; \quad \text{MUTE for } X = 0\]

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
</table>
| R11 DAC Digital Volume | 7:0 | DACVOL [7:0] | 11111111 (0dB) | DAC Digital Volume Control
|                  |     |          |             | 0000 0000 = Digital Mute
|                  |     |          |             | 0000 0001 = -127dB
|                  |     |          |             | 0000 0010 = -126.5dB
|                  |     |          |             | ... 0.5dB steps up to
|                  |     |          |             | 1111 1111 = 0dB                                                            |

Table 23 DAC Volume

**HI-FI DIGITAL TO ANALOGUE CONVERTER (DAC)**

After passing through the graphic equaliser filters, digital ‘de-emphasis’ can be applied to the audio data if necessary (e.g. when the data comes from a CD with pre-emphasis used in the recording). De-emphasis filtering is available for sample rates of 48kHz, 44.1kHz and 32kHz.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
</table>
| R10 DAC Control  | 5:4 | DEEMPH | 00      | De-Emphasis Control
|                  |     |        |         | 00 = No de-emphasis
|                  |     |        |         | 01 = 32kHz sample rate
|                  |     |        |         | 10 = 44.1kHz sample rate
|                  |     |        |         | 11 = 48kHz sample rate                           |

Table 24 De-Emphasis
The DAC is enabled by the DACEN register bit.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R3</td>
<td>0</td>
<td>DACEN</td>
<td>0</td>
<td>DAC enable</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = DAC disabled</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = DAC enabled</td>
</tr>
</tbody>
</table>

Table 25 DAC Enable

The WM8974 also has a Soft Mute function, which gradually attenuates the volume of the digital signal to zero. When removed, the gain will ramp back up to the digital gain setting. This function is enabled by default. To play back an audio signal, it must first be disabled by setting the DACMU bit to zero.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R10</td>
<td>6</td>
<td>DACMU</td>
<td>0</td>
<td>DAC soft mute enable</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = DACMU disabled</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = DACMU enabled</td>
</tr>
</tbody>
</table>

Table 26 DAC Control Register

The digital audio data is converted to oversampled bit streams in the on-chip, true 24-bit digital interpolation filters. The bit stream data enters a multi-bit, sigma-delta DAC, which converts it to a high quality analogue audio signal. The multi-bit DAC architecture reduces high frequency noise and sensitivity to clock jitter.

The DAC output defaults to non-inverted. Setting DACPOL will invert the DAC output phase.

AUTOMUTE

The DAC has an automute function which applies an analogue mute when 1024 consecutive zeros are detected. The mute is released as soon as a non-zero sample is detected. Automute can be disabled using the AMUTE control bit.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R10</td>
<td>2</td>
<td>AMUTE</td>
<td>0</td>
<td>DAC automute enable</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = automute disabled</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = automute enabled</td>
</tr>
</tbody>
</table>

Table 27 DAC Automute Control Register

DAC OUTPUT LIMITER

The WM8974 has a digital output limiter function. The operation of this is shown in Figure 19. In this diagram the upper graph shows the envelope of the input/output signals and the lower graph shows the gain characteristic.
The limiter has a programmable upper threshold which is close to 0dB. Referring to Table 28, in normal operation (LIMBOOST=000 => limit only) signals below this threshold are unaffected by the limiter. Signals above the upper threshold are attenuated at a specific attack rate (set by the LIMATK register bits) until the signal falls below the threshold. The limiter also has a lower threshold 1dB below the upper threshold. When the signal falls below the lower threshold the signal is amplified at a specific decay rate (controlled by LIMDCY register bits) until a gain of 0dB is reached. Both threshold levels are controlled by the LIMLVL register bits. The upper threshold is 0.5dB above the value programmed by LIMLVL and the lower threshold is 0.5dB below the LIMLVL value.

**VOLUME BOOST**

The limiter has programmable upper gain which boosts signals below the threshold to compress the dynamic range of the signal and increase its perceived loudness. This operates as an ALC function with limited boost capability. The volume boost is from 0dB to +12dB in 1dB steps, controlled by the LIMBOOST register bits.

The output limiter volume boost can also be used as a stand-alone digital gain boost when the limiter is disabled.
<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R24</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>DAC digital limiter control 1</td>
<td>3:0</td>
<td>LIMATK</td>
<td>0010</td>
<td>Limiter Attack time (per 6dB gain change) for 44.1kHz sampling. Note that these will scale with sample rate. 0000=94us 0001=188s 0010=375us 0011=750us 0100=1.5ms 0101=3ms 0110=6ms 0111=12ms 1000=24ms 1001=48ms 1010=96ms 1011 to 1111=192ms</td>
</tr>
<tr>
<td>7:4</td>
<td>LIMDCY</td>
<td>0001</td>
<td></td>
<td></td>
</tr>
<tr>
<td>DAC digital limiter control 2</td>
<td></td>
<td></td>
<td></td>
<td>Limiter Decay time (per 6dB gain change) for 44.1kHz sampling. Note that these will scale with sample rate: 000=750us 0001=1.5ms 0010=3ms 0011=6ms 0100=12ms 0101=24ms 0110=48ms 0111=96ms 1000=192ms 1001=384ms 1010=768ms 1011 to 1111=1.536s</td>
</tr>
<tr>
<td>8</td>
<td>LIMEN</td>
<td>0</td>
<td></td>
<td></td>
</tr>
<tr>
<td>DAC digital limiter control 2</td>
<td></td>
<td></td>
<td></td>
<td>Enable the DAC digital limiter: 0=disabled 1=enabled</td>
</tr>
<tr>
<td>R25</td>
<td>3:0</td>
<td>LIMBOOST</td>
<td>0000</td>
<td>Limiter volume boost (can be used as a stand-alone volume boost when LIMEN=0): 0000=0dB 0001=+1dB 0010=+2dB … (1dB steps) 1011=+11dB 1100=+12dB 1101 to 1111=reserved</td>
</tr>
<tr>
<td>6:4</td>
<td>LIMLVL</td>
<td>000</td>
<td></td>
<td></td>
</tr>
<tr>
<td>DAC digital limiter control 2</td>
<td></td>
<td></td>
<td></td>
<td>Programmable signal threshold level (determines level at which the limiter starts to operate) 000=−1dB 001=−2dB 010=−3dB 011=−4dB 100=−5dB 101 to 111=−6dB</td>
</tr>
</tbody>
</table>

Table 28 DAC Digital Limiter Control
GRAPHIC EQUALISER

A 5-band graphic EQ is provided, which can be applied to the ADC or DAC path under control of the EQMODE register bit.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R18 EQ Control 1</td>
<td>8</td>
<td>EQMODE</td>
<td>1</td>
<td>0 = Equaliser applied to ADC path 1 = Equaliser applied to DAC path</td>
</tr>
</tbody>
</table>

Table 29 EQ DAC or ADC Path Select

The equaliser consists of low and high frequency shelving filters (Band 1 and 5) and three peak filters for the centre bands. Each has adjustable cut-off or centre frequency, and selectable boost (+/- 12dB in 1dB steps). The peak filters have selectable bandwidth.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R18 EQ Band 1 Control</td>
<td>4:0</td>
<td>EQ1G</td>
<td>01100 (0dB)</td>
<td>Band 1 Gain Control. See Table 35 for details.</td>
</tr>
<tr>
<td></td>
<td>6:5</td>
<td>EQ1C</td>
<td>01</td>
<td>Band 1 Cut-off Frequency: 00=80Hz 01=105Hz 10=135Hz 11=175Hz</td>
</tr>
</tbody>
</table>

Table 30 EQ Band 1 Control

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R19 EQ Band 2 Control</td>
<td>4:0</td>
<td>EQ2G</td>
<td>01100 (0dB)</td>
<td>Band 2 Gain Control. See Table 35 for details.</td>
</tr>
<tr>
<td></td>
<td>6:5</td>
<td>EQ2C</td>
<td>01</td>
<td>Band 2 Centre Frequency: 00=230Hz 01=300Hz 10=385Hz 11=500Hz</td>
</tr>
<tr>
<td></td>
<td>8</td>
<td>EQ2BW</td>
<td>0</td>
<td>Band 2 Bandwidth Control 0=narrow bandwidth 1=wide bandwidth</td>
</tr>
</tbody>
</table>

Table 31 EQ Band 2 Control

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R20 EQ Band 3 Control</td>
<td>4:0</td>
<td>EQ3G</td>
<td>01100 (0dB)</td>
<td>Band 3 Gain Control. See Table 35 for details.</td>
</tr>
<tr>
<td></td>
<td>6:5</td>
<td>EQ3C</td>
<td>01</td>
<td>Band 3 Centre Frequency: 00=650Hz 01=850Hz 10=1.1kHz 11=1.4kHz</td>
</tr>
<tr>
<td></td>
<td>8</td>
<td>EQ3BW</td>
<td>0</td>
<td>Band 3 Bandwidth Control 0=narrow bandwidth 1=wide bandwidth</td>
</tr>
</tbody>
</table>

Table 32 EQ Band 3 Control
### Register Address, Bit, Label, Default, Description

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R21</td>
<td>4:0</td>
<td>EQ4G</td>
<td>01100</td>
<td>Band 4 Gain Control. See Table 35 for details.</td>
</tr>
<tr>
<td></td>
<td>6:5</td>
<td>EQ4C</td>
<td>01</td>
<td>Band 4 Centre Frequency:</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>00=1.8kHz</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>01=2.4kHz</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>10=3.2kHz</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>11=4.1kHz</td>
</tr>
<tr>
<td></td>
<td>8</td>
<td>EQ4BW</td>
<td>0</td>
<td>Band 4 Bandwidth Control</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0=narrow bandwidth</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1=wide bandwidth</td>
</tr>
</tbody>
</table>

**Table 33 EQ Band 4 Control**

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R22</td>
<td>4:0</td>
<td>EQ5G</td>
<td>01100</td>
<td>Band 5 Gain Control. See Table 35 for details.</td>
</tr>
<tr>
<td></td>
<td>6:5</td>
<td>EQ5C</td>
<td>01</td>
<td>Band 5 Cut-off Frequency:</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>00=5.3kHz</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>01=6.9kHz</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>10=9kHz</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>11=11.7kHz</td>
</tr>
</tbody>
</table>

**Table 34 EQ Band 5 Control**

<table>
<thead>
<tr>
<th>GAIN REGISTER</th>
<th>GAIN</th>
</tr>
</thead>
<tbody>
<tr>
<td>00000</td>
<td>+12dB</td>
</tr>
<tr>
<td>00001</td>
<td>+11dB</td>
</tr>
<tr>
<td>00010</td>
<td>+10dB</td>
</tr>
<tr>
<td>...</td>
<td>(1dB steps)</td>
</tr>
<tr>
<td>01100</td>
<td>0dB</td>
</tr>
<tr>
<td>01101</td>
<td>-1dB</td>
</tr>
<tr>
<td>11000</td>
<td>-12dB</td>
</tr>
<tr>
<td>11001 to 11111</td>
<td>Reserved</td>
</tr>
</tbody>
</table>

**Table 35 Gain Register Table**
The WM8974 has a single MONO output and two outputs SPKOUTP and SPOUTN for driving a mono BTL speaker. These analogue output stages are supplied from SPKVDD and are capable of driving up to 1.5V rms signals (equivalent to 3V rms into a bridge tied speaker) as shown in Figure 20.

The Mono and speaker outputs have output driving stages which can be controlled by the register bits MONOBOOST and SPKBOOST respectively. Each output stage has a selectable gain boost of 1.5x. When this boost is enabled the output DC level is also level shifted (from AVDD/2 to 1.5xAVDD/2) to prevent the signal from clipping. A dedicated amplifier, as shown in Figure 20, is used to perform the DC level shift operation. This buffer must be enabled using the BUFDCOPEN register bit for this operating mode. It should also be noted that if SPKVDD is not equal to or greater than 1.5xAVDD this boost mode may result in signals clipping. Table 37 summarises the effect of the SPKBOOST/MONOBOOST control bits.
**WM8974**

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R49</td>
<td>2</td>
<td>SPKBOOST</td>
<td>0</td>
<td>Speaker output boost stage control (see Table 37 for details)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = No boost (outputs are inverting buffers)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = 1.5x gain boost</td>
</tr>
<tr>
<td></td>
<td>3</td>
<td>MONOBOOST</td>
<td>0</td>
<td>Mono output boost stage control (see Table 37 for details)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = No boost (output is inverting buffer)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = 1.5x gain boost</td>
</tr>
<tr>
<td>R1</td>
<td>8</td>
<td>BUFDCOPEN</td>
<td>0</td>
<td>Dedicated buffer for DC level shifting output stages when in 1.5x gain boost configuration.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = Buffer disabled</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = Buffer enabled (required for 1.5x gain boost)</td>
</tr>
</tbody>
</table>

**Table 36 Output Boost Control**

<table>
<thead>
<tr>
<th>SPKBOOST/ MONOBOOST</th>
<th>OUTPUT STAGE GAIN</th>
<th>OUTPUT DC LEVEL</th>
<th>OUTPUT STAGE CONFIGURATION</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>1x</td>
<td>AVDD/2</td>
<td>Inverting</td>
</tr>
<tr>
<td>1</td>
<td>1.5x</td>
<td>1.5xAVDD/2</td>
<td>Non-inverting</td>
</tr>
</tbody>
</table>

**Table 37 Output Boost Stage Details**

**SPKOUTP/SPKOUTN OUTPUTS**

The SPKOUT pins can drive a single bridge tied 8Ω speaker or two headphone loads of 16Ω or 32Ω or a line output (see Headphone Output and Line Output sections, respectively). The signal to be output on SPKOUT comes from the Speaker Mixer circuit and can be any combination of the DAC output, the Bypass path (output of the boost stage) and the AUX input. The SPKOUTP/N volume is controlled by the SPKVOL register bits. Note that gains over 0dB may cause clipping if the signal is large. The SPKMUTE register bit causes the speaker outputs to be muted (the output DC level is driven out). The output pins remains at the same DC level (VMIDOP), so that no click noise is produced when muting or un-muting.

The SPKOUTN pin always drives out an inverted version of the SPKOUTP signal.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R50</td>
<td>0</td>
<td>DAC2SPK</td>
<td>1</td>
<td>Output of DAC to speaker mixer input</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = not selected</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = selected</td>
</tr>
<tr>
<td></td>
<td>1</td>
<td>BYP2SPK</td>
<td>0</td>
<td>Bypass path (output of input boost stage) to speaker mixer input</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = not selected</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = selected</td>
</tr>
<tr>
<td></td>
<td>5</td>
<td>AUX2SPK</td>
<td>0</td>
<td>Output of auxiliary amplifier to speaker mixer input</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = not selected</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = selected</td>
</tr>
<tr>
<td>R40</td>
<td>1</td>
<td>SPKATTN</td>
<td>0</td>
<td>Attenuation control for bypass path (output of input boost stage) to speaker mixer input</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = 0dB</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = -10dB</td>
</tr>
</tbody>
</table>

**Table 38 Speaker Mixer Control**
ZERO CROSS TIMEOUT

A zero-cross timeout function is also provided so that if zero cross is enabled on the input or output PGAs the gain will automatically update after a timeout period if a zero cross has not occurred. This is enabled by setting SLOWCLKEN. The timeout period is dependent on the clock input to the digital and is equal to \(2^{21} \times \text{input clock period}\).

### Table 40 Timeout Clock Enable Control

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R7</td>
<td>0</td>
<td>SLOWCLKEN</td>
<td>0</td>
<td>Slow clock enable. Used for both the jack insert detect de-bounce circuit and the zero cross timeout. 0 = slow clock disabled 1 = slow clock enabled</td>
</tr>
</tbody>
</table>

MONO MIXER AND OUTPUT

The MONOOUT pin can drive a 16Ω or 32Ω headphone or a line output or be used as a DC reference for a headphone output (see Headphone Output section). It can be selected to drive out any combination of DAC, Bypass (output of input boost stage) and AUX. This output is enabled by setting bit MONOEN.

### Table 39 SPKOUT Volume Control

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R54</td>
<td>7</td>
<td>SPKZC</td>
<td>0</td>
<td>Speaker Volume control zero cross enable: 1 = Change gain on zero cross only 0 = Change gain immediately</td>
</tr>
<tr>
<td></td>
<td>6</td>
<td>SPKMUTE</td>
<td>0</td>
<td>Speaker output mute enable 0 = Speaker output enabled 1 = Speaker output muted (VMIDOP)</td>
</tr>
<tr>
<td>5:0</td>
<td>SPKVOL [5:0]</td>
<td>111001 (0dB)</td>
<td>Speaker Volume Adjust 11111 = +6dB 11110 = +5dB … (1.0 dB steps) 11101 = 0dB … 00000 = -57dB</td>
<td></td>
</tr>
</tbody>
</table>

### Table 40 Timeout Clock Enable Control

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R7</td>
<td>0</td>
<td>SLOWCLKEN</td>
<td>0</td>
<td>Slow clock enable. Used for both the jack insert detect de-bounce circuit and the zero cross timeout. 0 = slow clock disabled 1 = slow clock enabled</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R56</td>
<td>0</td>
<td>DAC2MONO</td>
<td>0</td>
<td>Output of DAC to mono mixer input 0 = not selected 1 = selected</td>
</tr>
<tr>
<td></td>
<td>1</td>
<td>BYP2MONO</td>
<td>0</td>
<td>Bypass path (output of input boost stage) to mono mixer input 0 = non selected 1 = selected</td>
</tr>
<tr>
<td></td>
<td>2</td>
<td>AUX2MONO</td>
<td>0</td>
<td>Output of Auxilliary amplifier to mono mixer input: 0 = not selected 1 = selected</td>
</tr>
</tbody>
</table>
ENABLING THE OUTPUTS

Each analogue output of the WM8974 can be separately enabled or disabled. The analogue mixer associated with each output has a separate enable. All outputs are disabled by default. To save power, unused parts of the WM8974 should remain disabled.

Outputs can be enabled at any time, but it is not recommended to do so when BUFIO is disabled (BUFIOEN=0), as this may cause pop noise (see “Power Management” and “Applications Information” sections).

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R1</td>
<td>2</td>
<td>BUFIOEN</td>
<td>0</td>
<td>Unused input/output tie off buffer enable</td>
</tr>
<tr>
<td></td>
<td>8</td>
<td>BUFDCOPEN</td>
<td>0</td>
<td>Output stage 1.5xAVDD/2 driver enable</td>
</tr>
<tr>
<td></td>
<td>3</td>
<td>BIASEN</td>
<td>0</td>
<td>Analogue amplifiers bias enable</td>
</tr>
<tr>
<td>R3</td>
<td>2</td>
<td>SPKMIXEN</td>
<td>0</td>
<td>Speaker Mixer enable</td>
</tr>
<tr>
<td></td>
<td>3</td>
<td>MONOMIXEN</td>
<td>0</td>
<td>Mono mixer enable</td>
</tr>
<tr>
<td></td>
<td>5</td>
<td>SPKPEN</td>
<td>0</td>
<td>SPKOUTP enable</td>
</tr>
<tr>
<td></td>
<td>6</td>
<td>SPKNEN</td>
<td>0</td>
<td>SPKOUTN enable</td>
</tr>
<tr>
<td></td>
<td>7</td>
<td>MONOEN</td>
<td>0</td>
<td>MONOOUT enable</td>
</tr>
</tbody>
</table>

**Note:** All “Enable” bits are 1 = ON, 0 = OFF

Table 42 Output Stages Power Management Control

UNUSED ANALOGUE INPUTS/OUTPUTS

Whenever an analogue input/output is disabled, it remains connected to a voltage source (either AVDD/2 or 1.5xAVDD/2 as appropriate) through a resistor. This helps to prevent pop noise when the output is re-enabled. The resistance between the voltage buffer and the output pins can be controlled using the VROI control bit. The default impedance is low, so that any capacitors on the outputs can charge up quickly at start-up. If a high impedance is desired for disabled outputs, VROI can then be set to 1, increasing the resistance to about 30kΩ.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R49</td>
<td>0</td>
<td>VROI</td>
<td>0</td>
<td>VREF (AVDD/2 or 1.5xAVDD/2) to analogue output resistance</td>
</tr>
</tbody>
</table>

0: approx 1kΩ
1: approx 30 kΩ

Table 43 Disabled Outputs to VREF Resistance
A dedicated buffer is available for tying off unused analogue I/O pins as shown in Figure 21. This buffer can be enabled using the BUFIOEN register bit.

If the SPKBOOST or MONOBOOST bits are set then the relevant outputs will be tied to the output of the DC level shift buffer at 1.5xAVDD/2 when disabled.

Table 44 summarises the tie-off options for the speaker and mono output pins.

![Figure 21 Unused Input/Output Pin Tie-off Buffers](image)

<table>
<thead>
<tr>
<th>MONOEN/SPKN/PEN</th>
<th>MONOBOOST/SPKBOOST</th>
<th>VROI</th>
<th>OUTPUT CONFIGURATION</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>0</td>
<td>0</td>
<td>1kΩ tieoff to AVDD/2</td>
</tr>
<tr>
<td>0</td>
<td>0</td>
<td>1</td>
<td>30kΩ tieoff to AVDD/2</td>
</tr>
<tr>
<td>0</td>
<td>1</td>
<td>0</td>
<td>1kΩ tieoff to 1.5xAVDD/2</td>
</tr>
<tr>
<td>0</td>
<td>1</td>
<td>1</td>
<td>30kΩ tieoff to 1.5xAVDD/2</td>
</tr>
<tr>
<td>1</td>
<td>0</td>
<td>X</td>
<td>Output enabled (DC level=AVDD/2)</td>
</tr>
<tr>
<td>1</td>
<td>1</td>
<td>X</td>
<td>Output enabled (DC level=1.5xAVDD/2)</td>
</tr>
</tbody>
</table>

Table 44 Unused Output Pin Tie-off Options
OUTPUT SWITCH

When the device is configured with a 2-wire interface the CSB/GPIO pin can be used as a switch control input to automatically disable the speaker outputs and enable the mono output. For example when a line is plugged into a jack socket. In this mode, enabled by setting GPIOSEL=001, pin CSB/GPIO switches between mono and speaker outputs (e.g. when pin 12 is connected to a mechanical switch in the headphone socket to detect plug-in). The GPIOPOL bit reverses the polarity of the CSB/GPIO input pin.

Note that the speaker outputs and the mono output must be enabled for this function to work (see Table 45). The CSB/GPIO pin has an internal de-bounce circuit when in this mode in order to prevent the output enables from toggling multiple times due to input glitches. This de-bounce circuit is clocked from a slow clock with period \(2^{21} \times \text{MCLK}\), enabled using the SLOWCLKEN register bit.

<table>
<thead>
<tr>
<th>GPIOPOL</th>
<th>CSB/GPIO</th>
<th>SPKENC/ SPKPEN</th>
<th>MONOEN</th>
<th>SPEAKER ENABLED</th>
<th>MONO OUTPUT ENABLED</th>
</tr>
</thead>
<tbody>
<tr>
<td>0 0 X 0</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
</tr>
<tr>
<td>0 1 X 0</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>0 1 X 1</td>
<td>Yes</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
</tr>
<tr>
<td>1 0 X 0</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>1 1 X 0</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
</tr>
<tr>
<td>1 1 X 1</td>
<td>Yes</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
</tr>
</tbody>
</table>

Table 45 Output Switch Operation (GPIOSEL=001)

THERMAL SHUTDOWN

The speaker outputs can drive very large currents. To protect the WM8974 from overheating a thermal shutdown circuit is included. The thermal shutdown can be configured to produce an interrupt when the device reaches approximately 125°C. See General Purpose Input/Output section.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R49</td>
<td>1</td>
<td>TSDEN</td>
<td>1</td>
<td>Thermal Shutdown Enable</td>
</tr>
<tr>
<td>Output control</td>
<td></td>
<td></td>
<td></td>
<td>0 : thermal shutdown disabled</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 : thermal shutdown enabled</td>
</tr>
</tbody>
</table>

Table 46 Thermal Shutdown

SPEAKER OUTPUT

SPKOUTP/N can differentially drive a mono 8Ω Bridge Tied Load (BTL) speaker as shown below.

\[ V_{\text{SPKR}} = L(-R) = L+R \]

Figure 22 Speaker Output Connection
HEADPHONE OUTPUT

The speaker outputs can drive a 16Ω or 32Ω headphone load, either through DC blocking capacitors, or DC coupled without any capacitor.

Headphone Output using DC Blocking Capacitors:

DC Coupled Headphone Output:

When DC blocking capacitors are used, then their capacitance and the load resistance together determine the lower cut-off frequency, \( f_c \). Increasing the capacitance lowers \( f_c \), improving the bass response. Smaller capacitance values will diminish the bass response. Assuming a 16Ω load and \( C_1, C_2 = 220 \mu F \):

\[
 f_c = \frac{1}{2\pi R_L C_1} = \frac{1}{(2\pi \times 16\Omega \times 220\mu F)} = 45 \text{ Hz}
\]

In the DC coupled configuration, the headphone “ground” is connected to the MONOOUT pin. The MONOOUT pin can be configured as a DC output driver by setting the MONOMUTE register bit. The DC voltage on MONOOUT in this configuration is equal to the DC offset on the SPKOUTP and SPKOUTN pins therefore no DC blocking capacitors are required. This saves space and material cost in portable applications.

It is recommended to connect the DC coupled outputs only to headphones, and not to the line input of another device. Although the built-in short circuit protection will prevent any damage to the headphone outputs, such a connection may be noisy, and may not function properly if the other device is grounded.

MONO OUTPUT

The mono output, can be used as a line output, a headphone output or auxiliary ground for cap-less driving of loads by SPKOUT. Recommended external components are shown below.

The DC blocking capacitors and the load resistance together determine the lower cut-off frequency, \( f_c \). Assuming a 10 kΩ load and \( C_1 = 1 \mu F \):

\[
 f_c = \frac{1}{2\pi (R_L+R_1) C_1} = \frac{1}{(2\pi \times 10.1k\Omega \times 1 \mu F)} = 16 \text{ Hz}
\]

Increasing the capacitance lowers \( f_c \), improving the bass response. Smaller values of \( C_1 \) will diminish the bass response. The function of \( R_1 \) is to protect the line outputs from damage when used improperly.
DIGITAL AUDIO INTERFACES

The audio interface has four pins:

- ADCDAT: ADC data output
- DACDAT: DAC data input
- FRAME: Data alignment clock
- BCLK: Bit clock, for synchronisation

The clock signals BCLK, and FRAME can be outputs when the WM8974 operates as a master, or inputs when it is a slave (see Master and Slave Mode Operation, below).

Four different audio data formats are supported:

- Left justified
- Right justified
- I2S
- DSP mode

All of these modes are MSB first. They are described in Audio Data Formats, below. Refer to the Electrical Characteristic section for timing information.

MASTER AND SLAVE MODE OPERATION

The WM8974 audio interface may be configured as either master or slave. As a master interface device the WM8974 generates BCLK and FRAME and thus controls sequencing of the data transfer on ADCDAT and DACDAT. To set the device to master mode register bit MS should be set high. In slave mode (MS=0), the WM8974 responds with data to clocks it receives over the digital audio interfaces.

AUDIO DATA FORMATS

In Left Justified mode, the MSB is available on the first rising edge of BCLK following an FRAME transition. The other bits up to the LSB are then transmitted in order. Depending on word length, BCLK frequency and sample rate, there may be unused BCLK cycles before each FRAME transition.

![Figure 25](image-url)  

**Figure 25**  Left Justified Audio Interface (assuming n-bit word length)

In Right Justified mode, the LSB is available on the last rising edge of BCLK before a FRAME transition. All other bits are transmitted before (MSB first). Depending on word length, BCLK frequency and sample rate, there may be unused BCLK cycles after each FRAME transition.
In I2S mode, the MSB is available on the second rising edge of BCLK following a FRAME transition. The other bits up to the LSB are then transmitted in order. Depending on word length, BCLK frequency and sample rate, there may be unused BCLK cycles between the LSB of one sample and the MSB of the next.

In DSP/PCM mode, the left channel MSB is available on the 2nd rising edge of BCLK (selectable by LRP) following a rising edge of FRAME. Right channel data immediately follows left channel data. Depending on word length, BCLK frequency and sample rate, there may be unused BCLK cycles between the LSB of the right channel data and the next sample. FRAMEP should be set to 0 in this mode.
### AUDIO INTERFACE CONTROL

The register bits controlling audio format, word length and master / slave mode are summarised below. Each audio interface can be controlled individually.

Register bit MS selects audio interface operation in master or slave mode. In Master mode BCLK, and FRAME are outputs. The frequency of BCLK and FRAME in master mode are controlled with BCLKDIV. These are divided down versions of master clock. This may result in short BCLK pulses at the end of a frame if there is a non-integer ratio of BCLKs to FRAME clocks.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R6</td>
<td>0</td>
<td>MS</td>
<td>0</td>
<td>Sets the chip to be master over FRAME and BCLK 0=BCLK and FRAME clock are inputs 1=BCLK and FRAME clock are outputs generated by the WM8974 (MASTER)</td>
</tr>
</tbody>
</table>

#### Table 47 Audio Interface Control

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R4</td>
<td>1</td>
<td>ADCLRSWAP</td>
<td>0</td>
<td>Controls whether ADC data appears in 'right' or 'left' phases of FRAME clock: 0=ADC data appear in 'left' phase of FRAME 1=ADC data appears in 'right' phase of FRAME</td>
</tr>
<tr>
<td></td>
<td>2</td>
<td>DACLRSWAP</td>
<td>0</td>
<td>Controls whether DAC data appears in 'right' or 'left' phases of FRAME clock: 0=DAC data appear in 'left' phase of FRAME 1=DAC data appears in 'right' phase of FRAME</td>
</tr>
<tr>
<td></td>
<td>4:3</td>
<td>FMT</td>
<td>10</td>
<td>Audio interface Data Format Select: 00=Right Justified 01=Left Justified 10=I²S format 11= DSP/PCM mode</td>
</tr>
<tr>
<td></td>
<td>6:5</td>
<td>WL</td>
<td>10</td>
<td>Word length 00=16 bits 01=20 bits 10=24 bits 11=32 bits (see note)</td>
</tr>
<tr>
<td></td>
<td>7</td>
<td>FRAMEP</td>
<td>0</td>
<td>Frame clock polarity 0=normal 1=inverted DSP Mode control 1 = Reserved 0 = Configures interface so that MSB is available on 2nd BCLK rising edge after FRAME rising edge</td>
</tr>
<tr>
<td></td>
<td>8</td>
<td>BCP</td>
<td>0</td>
<td>BCLK polarity 0=normal 1=inverted</td>
</tr>
<tr>
<td>REGISTER ADDRESS</td>
<td>BIT</td>
<td>LABEL</td>
<td>DEFAULT</td>
<td>DESCRIPTION</td>
</tr>
<tr>
<td>------------------</td>
<td>-----</td>
<td>--------</td>
<td>---------</td>
<td>-------------</td>
</tr>
<tr>
<td></td>
<td>4:2</td>
<td>BCLKDIV</td>
<td>000</td>
<td>Configures the BCLK and FRAME output frequency, for use when the chip is master over BCLK.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>000=divide by 1 (BCLK=MCLK)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>001=divide by 2 (BCLK=MCLK/2)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>010=divide by 4</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>011=divide by 8</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>100=divide by 16</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>101=divide by 32</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>110=reserved</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>111=reserved</td>
</tr>
<tr>
<td>7:5</td>
<td>MCLKDIV</td>
<td>010</td>
<td>Sets the scaling for either the MCLK or PLL clock output (under control of CLKSEL)</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>000=divide by 1</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>001=divide by 1.5</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>010=divide by 2</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>011=divide by 3</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>100=divide by 4</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>101=divide by 6</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>110=divide by 8</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>111=divide by 12</td>
</tr>
<tr>
<td>8</td>
<td>CLKSEL</td>
<td>1</td>
<td>Controls the source of the clock for all internal operation:</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0=MCLK</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1=PLL output</td>
</tr>
</tbody>
</table>

Table 48 Clock Control

Note that the setting MCLKDIV=000 and BCLKDIV=000 must not be used simultaneously.

LOOPBACK

Setting the LOOPBACK register bit enables digital loopback. When this bit is set the output data from the ADC audio interface is fed directly into the DAC data input.

COMPANDING

The WM8974 supports A-law and µ-law companding on both transmit (ADC) and receive (DAC) sides. Companding can be enabled on the DAC or ADC audio interfaces by writing the appropriate value to the DAC_COMP or ADC_COMP register bits respectively.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>R5</td>
<td>Comanding control</td>
<td>0</td>
<td>Digital loopback function</td>
</tr>
<tr>
<td></td>
<td></td>
<td>LOOPBACK</td>
<td>0</td>
<td>0=No loopback</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1=Loopback enabled, ADC data output is fed directly into DAC data input.</td>
</tr>
<tr>
<td></td>
<td>2:1</td>
<td>ADC_COMP</td>
<td>0</td>
<td>ADC companding</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>00=off</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>01=reserved</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>10=µ-law</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>11=A-law</td>
</tr>
</tbody>
</table>
Companding involves using a piecewise linear approximation of the following equations (as set out by ITU-T G.711 standard) for data compression:

\( \mu \)-law (where \( \mu = 255 \) for the U.S. and Japan):

\[
F(x) = \ln(1 + \mu|x|) / \ln(1 + \mu) \quad \text{for} \quad -1 \leq x \leq 1
\]

A-law (where \( A = 87.6 \) for Europe):

\[
F(x) = A|x| / (1 + \ln A) \quad \text{for} \quad x \leq 1/A \\
F(x) = (1 + \ln A|x|) / (1 + \ln A) \quad \text{for} \quad 1/A \leq x \leq 1
\]

The companded data is also inverted as recommended by the G.711 standard (all 8 bits are inverted for \( \mu \)-law, all even data bits are inverted for A-law). The data will be transmitted as the first 8 MSB's of data.

Companding converts 13 bits (\( \mu \)-law) or 12 bits (A-law) to 8 bits using non-linear quantization. The input data range is separated into 8 levels, allowing low amplitude signals better precision than that of high amplitude signals. This is to exploit the operation of the human auditory system, where louder sounds do not require as much resolution as quieter sounds. The companded signal is an 8-bit word containing sign (1-bit), exponent (3-bits) and mantissa (4-bits).

<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>SIGN</td>
<td>EXPONENT</td>
<td>MANTISSA</td>
</tr>
</tbody>
</table>

Table 50 8-bit Companded Word Composition
AUDIO SAMPLE RATES

The WM8974 sample rates for the ADC and the DAC are set using the SR register bits. The cut-offs for the digital filters and the ALC attack/decay times stated are determined using these values and assume a 256fs master clock rate.

If a sample rate that is not explicitly supported by the SR register settings is required then the closest SR value to that sample rate should be chosen, the filter characteristics and the ALC attack, decay and hold times will scale appropriately.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R7</td>
<td>3:1</td>
<td>SR</td>
<td>000</td>
<td>Approximate sample rate (configures the coefficients for the internal digital filters): 000=48kHz 001=32kHz 010=24kHz 011=16kHz 100=12kHz 101=8kHz 110-111=reserved</td>
</tr>
</tbody>
</table>

Table 51 Sample Rate Control
MASTER CLOCK AND PHASE LOCKED LOOP (PLL)

The WM8974 has an on-chip phase-locked loop (PLL) circuit that can be used to:

Generate master clocks for the WM8974 audio functions from another external clock, e.g. in telecoms applications.

Generate and output (on pin CSB/GPIO) a clock for another part of the system that is derived from an existing audio master clock.

Figure 31 shows the PLL and internal clocking arrangement on the WM8974.

The PLL can be enabled or disabled by the PLLEN register bit.

Note: In order to minimise current consumption, the PLL is disabled when the VMIDSEL[1:0] bits are set to 00b. VMIDSEL[1:0] must be set to a value other than 00b to enable the PLL.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R1</td>
<td>5</td>
<td>PLLEN</td>
<td>0</td>
<td>PLL enable</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0=PLL off</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1=PLL on</td>
</tr>
</tbody>
</table>

Table 52 PLLEN Control Bit

The PLL frequency ratio $R = f_2/f_1$ (see Figure 31) can be set using the register bits PLLK and PLLN:

$$PLLN = \text{int } R$$

$$PLLK = \text{int } (2^{24} (R-PLLN))$$
EXAMPLE:

MCLK=12MHz, required clock = 12.288MHz.

R should be chosen to ensure 5 < PLLN < 13. There is a fixed divide by 4 in the PLL and a selectable divide by N after the PLL which should be set to divide by 2 to meet this requirement.

Enabling the divide by 2 sets the required f2 = 4 x 2 x 12.288MHz = 98.304MHz.

\[ R = \frac{98.304}{12} = 8.192 \]

\[ \text{PLLN} = \text{int} R = 8 \]

\[ k = \text{int} \left( \frac{2^{24} \times (8.192 - 8)}{2} \right) = 3221225 = 3126E9h \]

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R36 PLL N value</td>
<td>4</td>
<td>PLLPRESCALE</td>
<td>0</td>
<td>0 = MCLK input not divided (default)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = Divide MCLK by 2 before input to PLL</td>
</tr>
<tr>
<td></td>
<td>3:0</td>
<td>PLLN</td>
<td>1000</td>
<td>Integer (N) part of PLL input/output frequency ratio. Use values greater than 5 and less than 13.</td>
</tr>
<tr>
<td>R37 PLL K value 1</td>
<td>5:0</td>
<td>PLLK [23:18]</td>
<td>0Ch</td>
<td>Fractional (K) part of PLL input/output frequency ratio (treat as one 24-digit binary number).</td>
</tr>
<tr>
<td>R38 PLL K Value 2</td>
<td>8:0</td>
<td>PLLK [17:9]</td>
<td>093h</td>
<td></td>
</tr>
<tr>
<td>R39 PLL K Value 3</td>
<td>8:0</td>
<td>PLLK [8:0]</td>
<td>0E9h</td>
<td></td>
</tr>
</tbody>
</table>

Table 53 PLL Frequency Ratio Control

The PLL performs best when f2 is around 90MHz. Its stability peaks at N=8. Some example settings are shown in Table 54.

<table>
<thead>
<tr>
<th>MCLK (MHz)</th>
<th>DESIRED OUTPUT (MHz)</th>
<th>F2 (MHz)</th>
<th>PRESCALE DIVIDE</th>
<th>POSTSCALE DIVIDE</th>
<th>R (Hex)</th>
<th>N (Hex)</th>
<th>K (Hex)</th>
</tr>
</thead>
<tbody>
<tr>
<td>12</td>
<td>11.2896</td>
<td>90.3168</td>
<td>1 2</td>
<td></td>
<td>7</td>
<td>86C220</td>
<td></td>
</tr>
<tr>
<td>12</td>
<td>12.288</td>
<td>98.304</td>
<td>1 2</td>
<td></td>
<td>8.192</td>
<td>8</td>
<td>3126E8</td>
</tr>
<tr>
<td>13</td>
<td>11.2896</td>
<td>90.3168</td>
<td>1 2</td>
<td></td>
<td>6.947446</td>
<td>6</td>
<td>F28BD4</td>
</tr>
<tr>
<td>13</td>
<td>12.288</td>
<td>98.304</td>
<td>1 2</td>
<td></td>
<td>7.561846</td>
<td>7</td>
<td>8FD525</td>
</tr>
<tr>
<td>14.4</td>
<td>11.2896</td>
<td>90.3168</td>
<td>1 2</td>
<td></td>
<td>6.272</td>
<td>6</td>
<td>45A1CA</td>
</tr>
<tr>
<td>14.4</td>
<td>12.288</td>
<td>98.304</td>
<td>1 2</td>
<td></td>
<td>6.826667</td>
<td>6</td>
<td>D3A06E</td>
</tr>
<tr>
<td>19.2</td>
<td>11.2896</td>
<td>90.3168</td>
<td>2 2</td>
<td></td>
<td>9.408</td>
<td>9</td>
<td>6872AF</td>
</tr>
<tr>
<td>19.2</td>
<td>12.288</td>
<td>98.304</td>
<td>2 2</td>
<td></td>
<td>10.24</td>
<td>A</td>
<td>3D70A3</td>
</tr>
<tr>
<td>19.68</td>
<td>11.2896</td>
<td>90.3168</td>
<td>2 2</td>
<td></td>
<td>9.178537</td>
<td>9</td>
<td>2DB492</td>
</tr>
<tr>
<td>19.68</td>
<td>12.288</td>
<td>98.304</td>
<td>2 2</td>
<td></td>
<td>9.990243</td>
<td>9</td>
<td>FD809F</td>
</tr>
<tr>
<td>19.8</td>
<td>11.2896</td>
<td>90.3168</td>
<td>2 2</td>
<td></td>
<td>9.122909</td>
<td>9</td>
<td>1F76F7</td>
</tr>
<tr>
<td>19.8</td>
<td>12.288</td>
<td>98.304</td>
<td>2 2</td>
<td></td>
<td>9.929697</td>
<td>9</td>
<td>EE009E</td>
</tr>
<tr>
<td>24</td>
<td>11.2896</td>
<td>90.3168</td>
<td>2 2</td>
<td></td>
<td>7.5264</td>
<td>7</td>
<td>86C226</td>
</tr>
<tr>
<td>24</td>
<td>12.288</td>
<td>98.304</td>
<td>2 2</td>
<td></td>
<td>8.192</td>
<td>8</td>
<td>3126E8</td>
</tr>
<tr>
<td>26</td>
<td>11.2896</td>
<td>90.3168</td>
<td>2 2</td>
<td></td>
<td>6.947446</td>
<td>6</td>
<td>F28BD4</td>
</tr>
<tr>
<td>26</td>
<td>12.288</td>
<td>98.304</td>
<td>2 2</td>
<td></td>
<td>7.561846</td>
<td>7</td>
<td>8FD525</td>
</tr>
<tr>
<td>27</td>
<td>11.2896</td>
<td>90.3168</td>
<td>2 2</td>
<td></td>
<td>6.690133</td>
<td>6</td>
<td>BOAC93</td>
</tr>
<tr>
<td>27</td>
<td>12.288</td>
<td>98.304</td>
<td>2 2</td>
<td></td>
<td>7.281778</td>
<td>7</td>
<td>482296</td>
</tr>
</tbody>
</table>

Table 54 PLL Frequency Examples
GENERAL PURPOSE INPUT/OUTPUT

The CSB/GPIO pin can be configured to perform a variety of useful tasks by setting the GPIOSEL register bits. The GPIO is only available in 2 wire mode.

Note that SLOWCLKEN must be enabled when using the jack detect function.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R8 GPIO control</td>
<td>2:0</td>
<td>GPIOSEL</td>
<td>000</td>
<td>CSB/GPIO pin function select:</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>000=CSB input</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>001= Jack insert detect</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>010=Temp ok</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>011=Auto mute active</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>100.PLL clk o/p</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>101.PLL lock</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>110=Reserved</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>111=Reserved</td>
</tr>
<tr>
<td></td>
<td>3</td>
<td>GPIOPOL</td>
<td>0</td>
<td>GPIO Polarity invert</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0=Non inverted</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1=Inverted</td>
</tr>
<tr>
<td></td>
<td>5:4</td>
<td>OPCLKDIV</td>
<td>00</td>
<td>PLL Output clock division ratio</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>00=divide by 1</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>01=divide by 2</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>10=divide by 3</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>11=divide by 4</td>
</tr>
</tbody>
</table>

Table 55 CSB/GPIO Control

CONTROL INTERFACE

SELECTION OF CONTROL MODE AND 2-WIRE MODE ADDRESS

The control interface can operate as either a 3-wire or 2-wire MPU interface. The MODE pin determines the 2 or 3 wire mode as shown in Table 56.

The WM8974 is controlled by writing to registers through a serial control interface. A control word consists of 16 bits. The first 7 bits (B15 to B9) are address bits that select which control register is accessed. The remaining 9 bits (B8 to B0) are register bits, corresponding to the 9 bits in each control register.

<table>
<thead>
<tr>
<th>MODE</th>
<th>INTERFACE FORMAT</th>
</tr>
</thead>
<tbody>
<tr>
<td>Low</td>
<td>2 wire</td>
</tr>
<tr>
<td>High</td>
<td>3 wire</td>
</tr>
</tbody>
</table>

Table 56 Control Interface Mode Selection
3-WIRE SERIAL CONTROL MODE

In 3-wire mode, every rising edge of SCLK clocks in one data bit from the SDIN pin. A rising edge on CSB/GPIO latches in a complete control word consisting of the last 16 bits.

![Diagram of 3-Wire Serial Control Interface](image)

Figure 32 3-Wire Serial Control Interface

2-WIRE SERIAL CONTROL MODE

The WM8974 supports software control via a 2-wire serial bus. Many devices can be controlled by the same bus, and each device has a unique 7-bit device address (this is not the same as the 7-bit address of each register in the WM8974).

The WM8974 operates as a slave device only. The controller indicates the start of data transfer with a high to low transition on SDIN while SCLK remains high. This indicates that a device address and data will follow. All devices on the 2-wire bus respond to the start condition and shift in the next eight bits on SDIN (7-bit address + Read/Write bit, MSB first). If the device address received matches the address of the WM8974, then the WM8974 responds by pulling SDIN low on the next clock pulse (ACK). If the address is not recognised or the R/W bit is ‘1’ when operating in write only mode, the WM8974 returns to the idle condition and wait for a new start condition and valid address.

During a write, once the WM8974 has acknowledged a correct address, the controller sends the first byte of control data (B15 to B8, i.e. the WM8974 register address plus the first bit of register data). The WM8974 then acknowledges the first data byte by pulling SDIN low for one clock pulse. The controller then sends the second byte of control data (B7 to B0, i.e. the remaining 8 bits of register data), and the WM8974 acknowledges again by pulling SDIN low.

Transfers are complete when there is a low to high transition on SDIN while SCLK is high. After a complete sequence the WM8974 returns to the idle state and waits for another start condition. If a start or stop condition is detected out of sequence at any point during data transfer (i.e. SDIN changes while SCLK is high), the device jumps to the idle condition.

![Diagram of 2-Wire Serial Control Interface](image)

Figure 33 2-Wire Serial Control Interface

In 2-wire mode the WM8974 has a fixed device address, 0011010.
RESETTING THE CHIP

The WM8974 can be reset by performing a write of any value to the software reset register (address 0 hex). This will cause all register values to be reset to their default values. In addition to this there is a Power-On Reset (POR) circuit which ensures that the registers are set to default when the device is powered up.

POWER SUPPLIES

The WM8974 can use up to four separate power supplies:

AVDD and AGND: Analogue supply, powers all analogue functions except the speaker output and mono output drivers. AVDD can range from 2.5V to 3.6V and has the most significant impact on overall power consumption (except for power consumed in the headphone). A large AVDD slightly improves audio quality.

SPKVDD and SPKGND: Headphone and Speaker supplies, power the speaker and mono output drivers. SPKVDD can range from 2.5V to 5.5V. SPKVDD can be tied to AVDD, but it requires separate layout and decoupling capacitors to curb harmonic distortion. With a larger SPKVDD, louder headphone and speaker outputs can be achieved with lower distortion. If SPKVDD is lower than AVDD (or 1.5 x AVDD for BOOST mode), the output signal may be clipped.

DCVDD: Digital core supply, powers all digital functions except the audio and control interfaces. DCVDD can range from 1.71V to 3.6V, and has no effect on audio quality. The return path for DCVDD is DGND, which is shared with DBVDD.

DBVDD: Can range from 1.71V to 3.6V. DBVDD return path is through DGND.

It is possible to use the same supply voltage for all four supplies. However, digital and analogue supplies should be routed and decoupled separately on the PCB to keep digital switching noise out of the analogue signal paths.

Note:

- DCVDD should be greater than or equal to 1.9V when using the PLL.
- DCVDD is less than or equal to DBVDD

RECOMMENDED POWER UP/DOWN SEQUENCE

In order to minimize output pop and click noise, it is recommended that the WM8974 device is powered up and down using one of the following sequences:

Power Up When NOT Using the Output 1.5x Boost Stage:

1. Turn on external power supplies. Wait for supply voltage to settle.
2. Set BIASEN = 1, BUFIOEN = 1 and also the VMIDSEL[1:0] bits in the Power Management 1 register. * Notes 1 and 2.
3. Wait for the VMID supply to settle. * Note 2.
4. Enable DAC by setting DACEN = 1.
5. Enable mixers as required.
6. Enable output stages as required.

Power Up When Using the Output 1.5x Boost Stage:

1. Turn on external power supplies. Wait for supply voltage to settle.
2. Enable 1.5x output boost. Set MONOBOOST = 1 and SPKBOOST = 1 as required.
3. Set BIASEN = 1, BUFIOEN = 1, BUFDCOPEN = 1 and also the VMIDSEL[1:0] bits in the Power Management 1 register. * Notes 1 and 2.

4. Wait for the VMID supply to settle. * Note 2.

5. Enable DAC by setting DACEN = 1.

6. Enable mixers as required.

7. Enable output stages as required.

**Power Down (all cases):**

1. Soft mute DAC by setting DACMU = 1.

2. Disable power management register 1 by setting R1[8:0]=0x00.

3. Disable all other output stages.

4. Turn off external power supplies.

**Notes:**

1. This step enables the internal device bias buffer and the VMID buffer for unassigned inputs/outputs. This will provide a start-up reference voltage for all inputs and outputs. This will cause the inputs and outputs to ramp towards VMID (NOT using output 1.5x boost) or 1.5 x (AVDD/2) (using output 1.5x boost) in a way that is controlled and predictable (see note 2).

2. Choose the value of the VMIDSEL bits based on the start-up time (VMIDSEL=10 for slowest start-up, VMIDSEL=11 for fastest start-up). Start-up time is defined by the value of the VMIDSEL bits (the reference impedance) and the external decoupling capacitor on VMID.

In addition to the power on sequence, it is recommended that the zero cross functions are used when changing the volume in the PGAs to avoid any audible pops or clicks.
### Table 57 Typical POR Operation (typical values, not tested)

<table>
<thead>
<tr>
<th>SYMBOL</th>
<th>MIN</th>
<th>TYPICAL</th>
<th>MAX</th>
<th>UNIT</th>
</tr>
</thead>
<tbody>
<tr>
<td>t_{midrail_on}</td>
<td>500</td>
<td>ms</td>
<td></td>
<td></td>
</tr>
<tr>
<td>t_{midrail_off}</td>
<td>&gt;10</td>
<td>s</td>
<td></td>
<td></td>
</tr>
<tr>
<td>t_{adcint}</td>
<td>2/fs</td>
<td>n/fs</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

### Notes:

1. The analogue input pin charge time, \( t_{\text{midrail_on}} \), is determined by the VMID pin charge time. This time is dependent upon the value of VMID decoupling capacitor and VMID pin input resistance and AVDD power supply rise time.

2. The analogue input pin discharge time, \( t_{\text{midrail_off}} \), is determined by the analogue input coupling capacitor discharge time. The time, \( t_{\text{midrail_off}} \), is measured using a 1μF capacitor on the analogue input but will vary dependent upon the value of input coupling capacitor.

3. While the ADC is enabled there will be LSB data bit activity on the ADCDAT pin due to system noise but no significant digital output will be present.

4. The VMIDSEL and BIASEN bits must be set to enable analogue input midrail voltage and for normal ADC operation.

5. ADCDAT data output delay from power \(-p\) - with power supplies starting from \(-V\) - is determined primarily by the VMID charge time. ADC initialisation and power management bits may be set immediately after POR is released; VMID charge time will be significantly longer and will dictate when the device is stabilised for analogue input.

6. ADCDAT data output delay at power up from device standby (power supplies already applied) is determined by ADC initialisation time, 2/fs.

---

**Figure 35 DAC Power Up and Down Sequence (not to scale)**
<table>
<thead>
<tr>
<th>SYMBOL</th>
<th>MIN</th>
<th>TYPICAL</th>
<th>MAX</th>
<th>UNIT</th>
</tr>
</thead>
<tbody>
<tr>
<td>$t_{\text{line_midrail_on}}$</td>
<td>500</td>
<td>ms</td>
<td></td>
<td></td>
</tr>
<tr>
<td>$t_{\text{line_midrail_off}}$</td>
<td>1</td>
<td>s</td>
<td></td>
<td></td>
</tr>
<tr>
<td>$t_{\text{hp_midrail_on}}$</td>
<td>500</td>
<td>ms</td>
<td></td>
<td></td>
</tr>
<tr>
<td>$t_{\text{hp_midrail_off}}$</td>
<td>6</td>
<td>s</td>
<td></td>
<td></td>
</tr>
<tr>
<td>$t_{\text{dacint}}$</td>
<td>2/fs</td>
<td>$n$/fs</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Table 58 Typical POR Operation (typical values, not tested)

Notes:

1. The lineout charge time, $t_{\text{line\_midrail\_on}}$, is mainly determined by the VMID pin charge time. This time is dependent upon the value of VMID decoupling capacitor and VMID pin input resistance and AVDD power supply rise time. The values above were measured using a 4.7 μF capacitor.

2. It is not advisable to allow DACDAT data input during initialisation of the DAC. If the DAC data value is not zero at point of initialisation, then this is likely to cause a pop noise on the analogue outputs. The same is also true if the DACDAT is removed at a non-zero value, and no mute function has been applied to the signal beforehand.

3. The lineout discharge time, $t_{\text{line\_midrail\_off}}$, is dependent upon the value of the lineout coupling capacitor and the leakage resistance path to ground. The values above were measured using a 10 μF output capacitor.

4. The headphone charge time, $t_{\text{hp\_midrail\_on}}$, is dependent upon the value of VMID decoupling capacitor and VMID pin input resistance and AVDD power supply rise time. The values above were measured using a 4.7 μF VMID decoupling capacitor.

5. The headphone discharge time, $t_{\text{hp\_midrail\_off}}$, is dependent upon the value of the headphone coupling capacitor and the leakage resistance path to ground. The values above were measured using a 100 μF capacitor.

6. The VMIDSEL and BIASEN bits must be set to enable analogue output midrail voltage and for normal DAC operation.
POWER MANAGEMENT

SAVING POWER BY REDUCING OVERSAMPLING RATE

The default mode of operation of the ADC and DAC digital filters is in 64x oversampling mode. Under the control of ADCOSR and DACOSR the oversampling rate may be doubled. 64x oversampling results in a slight decrease in noise performance compared to 128x but lowers the power consumption of the device.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R10 DAC control</td>
<td>3</td>
<td>DACOSR</td>
<td>0</td>
<td>DAC oversample rate select</td>
</tr>
<tr>
<td></td>
<td></td>
<td>128</td>
<td>0</td>
<td>0 = 64x (lowest power)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>1</td>
<td>1 = 128x (best SNR)</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R14 ADC control</td>
<td>3</td>
<td>ADCOSR</td>
<td>0</td>
<td>ADC oversample rate select</td>
</tr>
<tr>
<td></td>
<td></td>
<td>128</td>
<td>0</td>
<td>0 = 64x (lowest power)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>1</td>
<td>1 = 128x (best SNR)</td>
</tr>
</tbody>
</table>

Table 59  ADC and DAC Oversampling Rate Selection

VMID

The analogue circuitry will not work when VMID is disabled (VMIDSEL[1:0] = 00b). The impedance of the VMID resistor string, together with the decoupling capacitor on the VMID pin will determine the start-up time of the VMID circuit.

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R1 Power</td>
<td>1:0</td>
<td>VMIDSEL</td>
<td>0</td>
<td>Reference string impedance to VMID pin determines</td>
</tr>
<tr>
<td>management 1</td>
<td></td>
<td></td>
<td></td>
<td>start-up time):</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>00</td>
<td>00=off (open circuit)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>01</td>
<td>01=50kΩ</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>10</td>
<td>10=500kΩ</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>11</td>
<td>11=5kΩ (for fastest start-up)</td>
</tr>
</tbody>
</table>

Table 60  VMID Impedance Control

BIASEN

<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
</tr>
</thead>
<tbody>
<tr>
<td>R1 Power</td>
<td>3</td>
<td>BIASEN</td>
<td>0</td>
<td>Analogue amplifier bias control</td>
</tr>
<tr>
<td>management 1</td>
<td></td>
<td></td>
<td></td>
<td>0=Disabled</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>1</td>
<td>1=Enabled</td>
</tr>
</tbody>
</table>

Table 61  BIASEN Control

ESTIMATED SUPPLY CURRENTS

When either the DAC or ADC are enabled it is estimated that approximately 4mA will be drawn from DCVDD when DCVDD=1.8V and fs=48kHz (This will be lower at lower sample rates). When the PLL is enabled an additional 700 microamps will be drawn from DCVDD.
Table 59 shows the estimated 3.3V AVDD current drawn by various circuits, by register bit.

<table>
<thead>
<tr>
<th>REGISTER BIT</th>
<th>AVDD CURRENT (MILLIAMPS)</th>
</tr>
</thead>
<tbody>
<tr>
<td>BUFDCOPEN</td>
<td>0.1</td>
</tr>
<tr>
<td>MONOEN</td>
<td>0.2</td>
</tr>
<tr>
<td>PLLON</td>
<td>1.4 (with clocks applied)</td>
</tr>
<tr>
<td>MICBEN</td>
<td>0.5</td>
</tr>
<tr>
<td>BIASEN</td>
<td>0.3</td>
</tr>
<tr>
<td>BUFIOEN</td>
<td>0.1</td>
</tr>
<tr>
<td>VMIDSEL</td>
<td>10K=&gt;0.3, less than 0.1 for 50k/500k</td>
</tr>
<tr>
<td>BOOSTEN</td>
<td>0.2</td>
</tr>
<tr>
<td>INPPGAEN</td>
<td>0.2</td>
</tr>
<tr>
<td>ADCEN</td>
<td>x64 (ADCOSR=0)=&gt;2.6, x128 (ADCOSR=1)=&gt;4.9</td>
</tr>
<tr>
<td>MONOEN</td>
<td>0.2</td>
</tr>
<tr>
<td>SPKPEN</td>
<td>1mA from SPKVDD + 0.2mA from AVDD in 5V mode</td>
</tr>
<tr>
<td>SPKNEN</td>
<td>1mA from SPKVDD + 0.2mA from AVDD in 5V mode</td>
</tr>
<tr>
<td>MONOMIXEN</td>
<td>0.2</td>
</tr>
<tr>
<td>SPKMIXEN</td>
<td>0.2</td>
</tr>
<tr>
<td>DACEN</td>
<td>x64 (DACOSR=0)=&gt;1.8, x128(DACOSR=1)=&gt;1.9</td>
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Table 62  AVDD Supply Current
## REGISTER MAP

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<th>B7</th>
<th>B6</th>
<th>B5</th>
<th>B4</th>
<th>B3</th>
<th>B2</th>
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<td>TSDEN</td>
<td>VROI</td>
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</table>
### REGISTER BITS BY ADDRESS

**Notes:**

1. Default values of N/A indicate non-latched data bits (e.g. software reset or volume update bits).
2. Register bits marked as “Reserved” should not be changed from the default.

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<th>DESCRIPTION</th>
<th>REFER TO</th>
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<td>[8:0]</td>
<td>RESET</td>
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<td>Resetting the Chip</td>
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<td>Dedicated buffer for DC level shifting output stages when in 1.5x gain boost configuration. 0=Buffer disabled 1=Buffer enabled (required for 1.5x gain boost)</td>
<td>Analogue Outputs</td>
</tr>
<tr>
<td>7</td>
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<td>Microphone Bias Enable 0 = OFF (high impedance output) 1 = ON</td>
<td>Microphone Biasing Circuit</td>
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<td>Unused input/output tie off buffer enable 0=Disabled 1=Enabled</td>
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<td>Reference string impedance to VMID pin: 00=off (open circuit) 01=50kΩ 10=500kΩ 11=5kΩ</td>
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<td>FMT</td>
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<td>Audio interface Data Format Select:</td>
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<td>00</td>
<td>Right Justified</td>
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<td>01</td>
<td>Left Justified</td>
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<td>DACLRSWAP</td>
<td>0</td>
<td>Controls whether DAC data appears in ‘right’ or ‘left’ phases of FRAME clock:</td>
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<tr>
<td></td>
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<td>0</td>
<td>0=DAC data appear in ‘left’ phase of FRAME</td>
<td></td>
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<tr>
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<td></td>
<td></td>
<td>1</td>
<td>1=DAC data appears in ‘right’ phase of FRAME</td>
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<td>ADCLRSWAP</td>
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<td>Controls whether ADC data appears in ‘right’ or ‘left’ phases of FRAME clock:</td>
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<td>0=ADC data appear in ‘left’ phase of FRAME</td>
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<td>1=ADC data appears in ‘right’ phase of FRAME</td>
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Rev 4.7
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<tbody>
<tr>
<td>5  (05h)</td>
<td>8:5</td>
<td>Reserved</td>
<td>0000</td>
<td>Reserved</td>
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<tr>
<td>4:3</td>
<td>DAC_COMP</td>
<td>00</td>
<td>DAC companding</td>
<td>00=off 01=reserved 10=µ-law 11=A-law</td>
<td>Digital Audio Interfaces</td>
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<td>2:1</td>
<td>ADC_COMP</td>
<td>00</td>
<td>ADC companding</td>
<td>00=off 01=reserved 10=µ-law 11=A-law</td>
<td>Digital Audio Interfaces</td>
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<tr>
<td>0</td>
<td>LOOPBACK</td>
<td>0</td>
<td>Digital loopback function</td>
<td>0=No loopback 1=Loopback enabled, ADC data output is fed directly into DAC data input.</td>
<td>Digital Audio Interfaces</td>
</tr>
<tr>
<td>6  (06h)</td>
<td>8</td>
<td>CLKSEL</td>
<td>1</td>
<td>Controls the source of the clock for all internal operation: 0=MCLK 1=PLL output</td>
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<tr>
<td>7:5</td>
<td>MCLKDIV</td>
<td>010</td>
<td>Sets the scaling for either the MCLK or PLL clock output (under control of CLKSEL) 000=divide by 1 001=divide by 1.5 010=divide by 2 011=divide by 3 100=divide by 4 101=divide by 6 110=divide by 8 111=divide by 12</td>
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<tr>
<td>4:2</td>
<td>BCLKDIV</td>
<td>000</td>
<td>Configures the BCLK and FRAME output frequency, for use when the chip is master over BCLK 000=divide by 1 (BCLK=MCLK) 001=divide by 2 (BCLK=MCLK/2) 010=divide by 4 011=divide by 8 100=divide by 16 101=divide by 32 110=reserved 111=reserved</td>
<td>Digital Audio Interfaces</td>
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<td>Reserved</td>
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<tr>
<td>0</td>
<td>MS</td>
<td>0</td>
<td>Sets the chip to be master over FRAME and BCLK 0=BCLK and FRAME clock are inputs 1=BCLK and FRAME clock are outputs generated by the WM8974 (MASTER)</td>
<td>Digital Audio Interfaces</td>
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<td>7  (07h)</td>
<td>8:4</td>
<td>Reserved</td>
<td>00000</td>
<td>Reserved</td>
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<td>3:1</td>
<td>SR</td>
<td>000</td>
<td>Approximate sample rate (configures the coefficients for the internal digital filters): 000=48kHz 001=32kHz 010=24kHz 011=16kHz 100=12kHz 101=8kHz 110-111=reserved</td>
<td>Audio Sample Rates</td>
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<td>Reserved</td>
<td>000</td>
<td>Reserved</td>
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<td></td>
<td>OPCLKDIV</td>
<td>00</td>
<td>PLL Output clock division ratio</td>
<td>General Purpose Input Output</td>
</tr>
<tr>
<td></td>
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<td></td>
<td>00:divide by 1</td>
<td>01=divide by 2</td>
<td>10=divide by 3</td>
</tr>
<tr>
<td>3</td>
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<td>GPIOPOL</td>
<td>0</td>
<td>GPIO Polarity invert</td>
<td>General Purpose Input Output</td>
</tr>
<tr>
<td></td>
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<td></td>
<td>0=Non inverted</td>
<td>1=Inverted</td>
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</tr>
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<td>2:0</td>
<td></td>
<td>GPIOSEL</td>
<td>000</td>
<td>CSB/GPIO pin function select:</td>
<td>General Purpose Input Output</td>
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<tr>
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<td>000=CSB input</td>
<td>001= Jack insert detect</td>
<td>010=Temp ok</td>
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<tr>
<td>9 (09h)</td>
<td>8:0</td>
<td>Reserved</td>
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<td>10 (0Ah)</td>
<td>8:7</td>
<td>Reserved</td>
<td>00</td>
<td>Reserved</td>
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<td>6</td>
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<td>DACMU</td>
<td>0</td>
<td>DAC soft mute enable</td>
<td>Output Signal Path</td>
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<td>0 = DACMU disabled</td>
<td>1 = DACMU enabled</td>
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<td></td>
<td>DEEMPH</td>
<td>00</td>
<td>De-Emphasis Control</td>
<td>Output Signal Path</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>00 = No de-emphasis</td>
<td>01 = 32kHz sample rate</td>
<td>10 = 44.1kHz sample rate</td>
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<td>3</td>
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<td>DACOSR128</td>
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<td>DAC oversample rate select</td>
<td>Power Management</td>
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<td>0 = 64x (lowest power)</td>
<td>1 = 128x (best SNR)</td>
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<td>AMUTE</td>
<td>0</td>
<td>DAC automute enable</td>
<td>Output Signal Path</td>
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<td>0 = automute disabled</td>
<td>1 = automute enabled</td>
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</tr>
<tr>
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<td></td>
<td>Reserved</td>
<td>0</td>
<td>Reserved</td>
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<td>DACPOL</td>
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<td>DAC Polarity Invert</td>
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<td>0 = No inversion</td>
<td>1 = DAC output inverted</td>
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<td>11 (0Bh)</td>
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<td>11111111</td>
<td>DAC Digital Volume Control</td>
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<td>0000 0000 = Digital Mute</td>
<td>0000 0001 = -127dB</td>
<td>0000 0010 = -126.5dB</td>
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<td>12 (0Ch)</td>
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<td>13 (0Dh)</td>
<td>8:0</td>
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<td>14 (0Eh)</td>
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<td>HPFEN</td>
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<td>High-Pass Filter Enable</td>
<td>Analogue to Digital Converter (ADC)</td>
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<td>LABEL</td>
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<td>DESCRIPTION</td>
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<td>HPFAPP</td>
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<td>Select audio mode or application mode</td>
<td>Analogue to Digital Converter (ADC)</td>
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<td>0=Audio mode (1st order, fc = ~3.7Hz)</td>
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<td>1=Application mode (2nd order, fc = HPFCUT)</td>
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<td>6:4</td>
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<td>HPFCUT</td>
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<td>Application mode cut-off frequency</td>
<td>Analogue to Digital Converter (ADC)</td>
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<td>See Table 11 for details.</td>
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<td>ADCOSR128</td>
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<td>0 = 64x (lowest power)</td>
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<td>1 = 128x (best SNR)</td>
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<td>ADCPOL</td>
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<td>ADC Polarity</td>
<td>Analogue to Digital Converter (ADC)</td>
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<td>0=normal</td>
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<td>1=inverted</td>
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<tr>
<td>15 (0Fh)</td>
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<td>ADCVOL</td>
<td>1111111</td>
<td>ADC Digital Volume Control</td>
<td>Analogue to Digital Converter (ADC)</td>
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<td>0000 0000 = Digital Mute</td>
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<td>0000 0001 = -127dB</td>
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<td></td>
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<td>0000 0010 = -126.5dB</td>
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<td></td>
<td>... 0.5dB steps up to 1111 1111 = 0dB</td>
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<td>18 (12h)</td>
<td>8</td>
<td>E Q MODE</td>
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<td>0 = Equaliser applied to ADC path</td>
<td>Output Signal Path</td>
</tr>
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<td>1 = Equaliser applied to DAC path</td>
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<td>0 =normal</td>
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<td>1=inverted</td>
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<td>6:5</td>
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<td>EQ1C</td>
<td>01</td>
<td>EQ Band 1 Cut-off Frequency</td>
<td>Output Signal Path</td>
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<td>00=80Hz</td>
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<td>01=105Hz</td>
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<td>10=135Hz</td>
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<td>11=175Hz</td>
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<td>EQ1G</td>
<td>01100</td>
<td>EQ Band 1 Gain Control. See Table 35 for details.</td>
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<td>E Q2 BW</td>
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<td>0=narrow bandwidth</td>
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<td></td>
<td>1=wide bandwidth</td>
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</tr>
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<td></td>
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<td>0=narrow bandwidth</td>
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<td>1=wide bandwidth</td>
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<td>EQ2C</td>
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<td>Band 2 Centre Frequency</td>
<td>Output Signal Path</td>
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<td>00=230Hz</td>
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<td>01=300Hz</td>
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<td>10=385Hz</td>
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<td>11=500Hz</td>
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<td>01100</td>
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<td>0=narrow bandwidth</td>
<td></td>
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<td></td>
<td></td>
<td></td>
<td>1=wide bandwidth</td>
<td></td>
</tr>
<tr>
<td>6:5</td>
<td></td>
<td>EQ3C</td>
<td>01</td>
<td>Band 3 Centre Frequency</td>
<td>Output Signal Path</td>
</tr>
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<td>00=650Hz</td>
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<td></td>
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<td>01=850Hz</td>
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<td>10=1.1kHz</td>
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<td></td>
<td>11=1.4kHz</td>
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<td>DESCRIPTION</td>
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<td>EQ3G</td>
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<tr>
<td>21 (15h)</td>
<td>8</td>
<td>EQ4BW</td>
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<td>Band 4 Bandwidth Control 0=narrow bandwidth 1=wide bandwidth</td>
<td>Output Signal Path</td>
</tr>
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<td>0</td>
<td>Reserved</td>
<td>Output Signal Path</td>
</tr>
<tr>
<td>6:5</td>
<td>EQ4C</td>
<td>01</td>
<td>Band 4 Centre Frequency: 00=1.8kHz 01=2.4kHz 10=3.2kHz 11=4.1kHz</td>
<td>Output Signal Path</td>
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<td>EQ4G</td>
<td>01100</td>
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</tr>
<tr>
<td>22 (16h)</td>
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<td>Output Signal Path</td>
</tr>
<tr>
<td>6:5</td>
<td>EQ5C</td>
<td>01</td>
<td>Band 5 Cut-off Frequency: 00=5.3kHz 01=6.9kHz 10=9kHz 11=11.7kHz</td>
<td>Output Signal Path</td>
<td></td>
</tr>
<tr>
<td>4:0</td>
<td>EQ5G</td>
<td>01100</td>
<td>Band 5 Gain Control. See Table 35 for details.</td>
<td>Output Signal Path</td>
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<tr>
<td>24 (18h)</td>
<td>8</td>
<td>LIMEN</td>
<td>0</td>
<td>Enable the DAC digital limiter: 0=disabled 1=enabled</td>
<td>Output Signal Path</td>
</tr>
<tr>
<td>7:4</td>
<td>LIMDCY</td>
<td>0011</td>
<td>DAC Limiter Decay time (per 6dB gain change) for 44.1kHz sampling. Note that these will scale with sample rate: 0000=750us 0001=1.5ms 0010=3ms 0011=6ms 0100=12ms 0101=24ms 0110=48ms 0111=96ms 1000=192ms 1001=384ms 1010=768ms 1011 to 1111=1.536s</td>
<td>Output Signal Path</td>
<td></td>
</tr>
<tr>
<td>3:0</td>
<td>LIMATK</td>
<td>0010</td>
<td>DAC Limiter Attack time (per 6dB gain change) for 44.1kHz sampling. Note that these will scale with sample rate: 0000=94us 0001=188s 0010=375us 0011=750us 0100=1.5ms 0101=3ms 0110=6ms 0111=12ms 1000=24ms 1001=48ms 1010=96ms 1011 to 1111=192ms</td>
<td>Output Signal Path</td>
<td></td>
</tr>
<tr>
<td>REGISTER ADDRESS</td>
<td>BIT</td>
<td>LABEL</td>
<td>DEFAULT</td>
<td>DESCRIPTION</td>
<td>REFER TO</td>
</tr>
<tr>
<td>------------------</td>
<td>-----</td>
<td>-------</td>
<td>---------</td>
<td>-------------</td>
<td>----------</td>
</tr>
<tr>
<td>25 (19h)</td>
<td>8:7</td>
<td>0</td>
<td>Reserved</td>
<td>DAC Limiter Programable signal threshold level (determines level at which the limiter starts to operate) 0000=-1dB 0001=-2dB 0010=-3dB 0011=-4dB 0100=-5dB 101 to 111=-6dB</td>
<td>Output Signal Path</td>
</tr>
<tr>
<td>6:4</td>
<td>0</td>
<td>LIMLVL</td>
<td>000</td>
<td>DAC Limiter volume boost (can be used as a stand-alone volume boost when LIMEN=0): 0000=0dB 0001=+1dB 0010=+2dB … (1dB steps) 1011=+11dB 1100=+12dB 1101 to 1111=reserved</td>
<td>Output Signal Path</td>
</tr>
<tr>
<td>3:0</td>
<td>0</td>
<td>LIMBOOST</td>
<td>0000</td>
<td>DAC Limiter volume boost (can be used as a stand-alone volume boost when LIMEN=0): 0000=0dB 0001=+1dB 0010=+2dB … (1dB steps) 1011=+11dB 1100=+12dB 1101 to 1111=reserved</td>
<td>Output Signal Path</td>
</tr>
<tr>
<td>27 (1Bh)</td>
<td>8</td>
<td>NFU</td>
<td>0</td>
<td>Notch filter update. The notch filter values used internally only update when one of the NFU bits is set high.</td>
<td>Analogue to Digital Converter (ADC)</td>
</tr>
<tr>
<td>7</td>
<td>0</td>
<td>NFEN</td>
<td>0</td>
<td>Notch filter enable: 0=Disabled 1=Enabled</td>
<td>Analogue to Digital Converter (ADC)</td>
</tr>
<tr>
<td>6:0</td>
<td>0</td>
<td>NFA0[13:7]</td>
<td>0000000</td>
<td>Notch Filter a0 coefficient, bits [13:7]</td>
<td>Analogue to Digital Converter (ADC)</td>
</tr>
<tr>
<td>28 (1Ch)</td>
<td>8</td>
<td>NFU</td>
<td>0</td>
<td>Notch filter update. The notch filter values used internally only update when one of the NFU bits is set high.</td>
<td>Analogue to Digital Converter (ADC)</td>
</tr>
<tr>
<td>7</td>
<td>0</td>
<td></td>
<td>0</td>
<td>Reserved</td>
<td></td>
</tr>
<tr>
<td>6:0</td>
<td>0</td>
<td>NFA0[6:0]</td>
<td>0000000</td>
<td>Notch Filter a0 coefficient, bits [6:0]</td>
<td>Analogue to Digital Converter (ADC)</td>
</tr>
<tr>
<td>29 (1Dh)</td>
<td>8</td>
<td>NFU</td>
<td>0</td>
<td>Notch filter update. The notch filter values used internally only update when one of the NFU bits is set high.</td>
<td>Analogue to Digital Converter (ADC)</td>
</tr>
<tr>
<td>7</td>
<td>0</td>
<td></td>
<td>0</td>
<td>Reserved</td>
<td></td>
</tr>
<tr>
<td>30 (1Eh)</td>
<td>8</td>
<td>NFU</td>
<td>0</td>
<td>Notch filter update. The notch filter values used internally only update when one of the NFU bits is set high.</td>
<td>Analogue to Digital Converter (ADC)</td>
</tr>
<tr>
<td>7</td>
<td>0</td>
<td></td>
<td>0</td>
<td>Reserved</td>
<td></td>
</tr>
<tr>
<td>6:0</td>
<td>0</td>
<td>NFA1[6:0]</td>
<td>0000000</td>
<td>Notch Filter a1 coefficient, bits [6:0]</td>
<td>Analogue to Digital Converter (ADC)</td>
</tr>
<tr>
<td>32 (20h)</td>
<td>8</td>
<td>ALCSEL</td>
<td>0</td>
<td>ALC function select: 0=ALC off (PGA gain set by INPPGAVOL register bits) 1=ALC on (ALC controls PGA gain)</td>
<td>Input Limiter / Automatic Level Control (ALC)</td>
</tr>
<tr>
<td>7:6</td>
<td></td>
<td></td>
<td></td>
<td>Reserved</td>
<td></td>
</tr>
<tr>
<td>REGISTER ADDRESS</td>
<td>BIT</td>
<td>LABEL</td>
<td>DEFAULT</td>
<td>DESCRIPTION</td>
<td>REFER TO</td>
</tr>
<tr>
<td>------------------</td>
<td>-----</td>
<td>--------</td>
<td>----------</td>
<td>-----------------------------------------------------------------------------</td>
<td>--------------------------------------------------------------------------</td>
</tr>
<tr>
<td>5:3</td>
<td>5:3</td>
<td>ALCMAX</td>
<td>111</td>
<td>Set Maximum Gain of PGA when using ALC:</td>
<td>Input Limiter / Automatic Level Control (ALC)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>111=+35.25dB</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>110=+29.25dB</td>
<td></td>
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<td></td>
<td></td>
<td></td>
<td></td>
<td>101=+23.25dB</td>
<td></td>
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<td></td>
<td></td>
<td></td>
<td></td>
<td>100=+17.25dB</td>
<td></td>
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<td></td>
<td></td>
<td></td>
<td></td>
<td>011=+11.25dB</td>
<td></td>
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<td></td>
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<td></td>
<td></td>
<td>010=+5.25dB</td>
<td></td>
</tr>
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<td></td>
<td></td>
<td></td>
<td></td>
<td>001=+0.75dB</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>000= -6.75dB</td>
<td></td>
</tr>
<tr>
<td>2:0</td>
<td>2:0</td>
<td>ALCMIN</td>
<td>000</td>
<td>Set minimum gain of PGA when using ALC:</td>
<td>Input Limiter / Automatic Level Control (ALC)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>000= -12dB</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>001= -6dB</td>
<td></td>
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<td></td>
<td></td>
<td></td>
<td></td>
<td>010=0dB</td>
<td></td>
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<td></td>
<td></td>
<td></td>
<td></td>
<td>011= +6dB</td>
<td></td>
</tr>
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<td></td>
<td></td>
<td></td>
<td></td>
<td>100= +12dB</td>
<td></td>
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<td></td>
<td>101= +18dB</td>
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<td></td>
<td>110= +24dB</td>
<td></td>
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<td></td>
<td></td>
<td></td>
<td></td>
<td>111= +30dB</td>
<td></td>
</tr>
<tr>
<td>33 (21h)</td>
<td>8</td>
<td>ALCZC</td>
<td>0</td>
<td>ALC zero cross detection.</td>
<td>Input Limiter / Automatic Level Control (ALC)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = disabled</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = enabled</td>
<td></td>
</tr>
<tr>
<td>7:4</td>
<td>7:4</td>
<td>ALCHLD</td>
<td>000</td>
<td>ALC hold time before gain is increased.</td>
<td>Input Limiter / Automatic Level Control (ALC)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0000 = 0ms</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0001 = 2.67ms</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0010 = 5.33ms</td>
<td></td>
</tr>
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<td></td>
<td></td>
<td></td>
<td></td>
<td>… (time doubles with every step)</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1111 = 43.691s</td>
<td></td>
</tr>
<tr>
<td>3:0</td>
<td>3:0</td>
<td>ALCLEVEL</td>
<td>1011</td>
<td>ALC target – sets signal level at ADC input</td>
<td>Input Limiter / Automatic Level Control (ALC)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0000 = -28.5dB FS</td>
<td></td>
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<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0001 = -27.0dB FS</td>
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<td></td>
<td></td>
<td></td>
<td>… (1.5dB steps)</td>
<td></td>
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<td></td>
<td></td>
<td>1110 = -7.5dB FS</td>
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<td></td>
<td></td>
<td></td>
<td></td>
<td>1111 = -6dB FS</td>
<td></td>
</tr>
<tr>
<td>34 (22h)</td>
<td>8</td>
<td>ALCMODE</td>
<td>0</td>
<td>Determines the ALC mode of operation:</td>
<td>Input Limiter / Automatic Level Control (ALC)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = ALC mode</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = Limiter mode</td>
<td></td>
</tr>
<tr>
<td>7:4</td>
<td>7:4</td>
<td>ALCDCY</td>
<td>0011</td>
<td>Decay (gain ramp-up) time (ALCMODE =0)</td>
<td>Input Limiter / Automatic Level Control (ALC)</td>
</tr>
<tr>
<td></td>
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<td></td>
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<td></td>
<td></td>
<td></td>
<td>Per step</td>
<td>Per 6dB</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0000</td>
<td>410us</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0001</td>
<td>820us</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0010</td>
<td>1.64ms</td>
</tr>
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<td></td>
<td></td>
<td></td>
<td>… (time doubles with every step)</td>
<td></td>
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<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1010 or higher</td>
<td>420ms</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0011</td>
<td>90.8us</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0001</td>
<td>181.6us</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0010</td>
<td>363.2us</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>… (time doubles with every step)</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1010</td>
<td>93ms</td>
</tr>
<tr>
<td>REGISTER ADDRESS</td>
<td>BIT</td>
<td>LABEL</td>
<td>DEFAULT</td>
<td>DESCRIPTION</td>
<td>REFER TO</td>
</tr>
<tr>
<td>------------------</td>
<td>-----</td>
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<td>---------</td>
<td>-------------</td>
<td>-----------------------------------------------</td>
</tr>
<tr>
<td>3:0</td>
<td></td>
<td>ALCATK</td>
<td>0010</td>
<td>ALC attack (gain ramp-down) time</td>
<td>Input Limiter / Automatic Level Control (ALC)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>(ALCMODE = 0)</td>
<td></td>
</tr>
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<td></td>
<td></td>
<td></td>
<td></td>
<td>Per step</td>
<td>Per 6dB</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>0000</td>
<td>104us</td>
<td>832us</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>0001</td>
<td>208us</td>
<td>1,664ms</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>0010</td>
<td>416us</td>
<td>3,328ms</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>… (time doubles with every step)</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>1010 or higher</td>
<td>106ms</td>
<td>852ms</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>0010</td>
<td>ALC attack (gain ramp-down) time</td>
<td>Input Limiter / Automatic Level Control (ALC)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>(ALCMODE = 1)</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>Per step</td>
<td>Per 6dB</td>
<td>90% of range</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>0000</td>
<td>22.7us</td>
<td>182.4us</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>0001</td>
<td>45.4us</td>
<td>363.2us</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>0010</td>
<td>90.8us</td>
<td>726.4us</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>… (time doubles with every step)</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>1010</td>
<td>23.2ms</td>
<td>186ms</td>
</tr>
<tr>
<td>35 (23h)</td>
<td>8:4</td>
<td></td>
<td>000000</td>
<td>Reserved</td>
<td></td>
</tr>
<tr>
<td>3</td>
<td></td>
<td>NGEN</td>
<td>0</td>
<td>ALC Noise gate function enable</td>
<td>Input Limiter / Automatic Level Control (ALC)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = enable</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = disable</td>
<td></td>
</tr>
<tr>
<td>2:0</td>
<td></td>
<td>NGTH</td>
<td>000</td>
<td>ALC Noise gate threshold:</td>
<td>Input Limiter / Automatic Level Control (ALC)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>000</td>
<td>000 = -39dB</td>
<td></td>
</tr>
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<td></td>
<td></td>
<td></td>
<td></td>
<td>001 = -45dB</td>
<td></td>
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<td></td>
<td></td>
<td></td>
<td></td>
<td>010 = -51dB</td>
<td></td>
</tr>
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<td></td>
<td></td>
<td></td>
<td></td>
<td>111 = -81dB</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>… (6dB steps)</td>
<td>262ms</td>
<td>317ms</td>
</tr>
<tr>
<td>36 (24h)</td>
<td>8:5</td>
<td></td>
<td>0000</td>
<td>Reserved</td>
<td></td>
</tr>
<tr>
<td>4</td>
<td></td>
<td>PLLPRESCALE</td>
<td>0</td>
<td>0 = MCLK input not divided (default)</td>
<td>Master Clock and Phase Locked Loop (PLL)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>1 = Divide MCLK by 2 before input PLL</td>
<td></td>
<td></td>
</tr>
<tr>
<td>3:0</td>
<td></td>
<td>PLLN[3:0]</td>
<td>1000</td>
<td>Integer (N) part of PLL input/output frequency ratio. Use values greater than 5 and less than 13.</td>
<td>Master Clock and Phase Locked Loop (PLL)</td>
</tr>
<tr>
<td>37 (25h)</td>
<td>8:6</td>
<td></td>
<td>000</td>
<td>Reserved</td>
<td></td>
</tr>
<tr>
<td>5:0</td>
<td></td>
<td>PLLK[23:18]</td>
<td>001100</td>
<td>Fractional (K) part of PLL1 input/output frequency ratio (treat as one 24-digit binary number).</td>
<td>Master Clock and Phase Locked Loop (PLL)</td>
</tr>
<tr>
<td>38 (26h)</td>
<td>8:0</td>
<td></td>
<td>0100010011</td>
<td>Fractional (K) part of PLL1 input/output frequency ratio (treat as one 24-digit binary number).</td>
<td>Master Clock and Phase Locked Loop (PLL)</td>
</tr>
<tr>
<td>39 (27h)</td>
<td>8:0</td>
<td></td>
<td>011101001</td>
<td>Fractional (K) part of PLL1 input/output frequency ratio (treat as one 24-digit binary number).</td>
<td>Master Clock and Phase Locked Loop (PLL)</td>
</tr>
<tr>
<td>40 (28h)</td>
<td>8:3</td>
<td></td>
<td>00000000</td>
<td>Reserved</td>
<td></td>
</tr>
<tr>
<td>2</td>
<td></td>
<td>MONOATTN</td>
<td>0</td>
<td>Attenuation control for bypass path (output of input boost stage) to mono mixer input</td>
<td>Analogue Outputs</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = 0dB</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = -10dB</td>
<td></td>
</tr>
<tr>
<td>1</td>
<td></td>
<td>SPKATTN</td>
<td>0</td>
<td>Attenuation control for bypass path (output of input boost stage) to speaker mixer input</td>
<td>Analogue Outputs</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 = 0dB</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1 = -10dB</td>
<td></td>
</tr>
<tr>
<td>0</td>
<td></td>
<td></td>
<td>0</td>
<td>Reserved</td>
<td></td>
</tr>
<tr>
<td>REGISTER ADDRESS</td>
<td>BIT</td>
<td>LABEL</td>
<td>DEFAULT</td>
<td>DESCRIPTION</td>
<td>REFER TO</td>
</tr>
<tr>
<td>------------------</td>
<td>-----</td>
<td>-------------</td>
<td>---------</td>
<td>------------------------------------------------------------------------------</td>
<td>-------------------</td>
</tr>
</tbody>
</table>
| 44 (2Ch)         | 8   | MBVSEL      | 0       | Microphone Bias Voltage Control  
|                  |     |             |         | 0 = 0.9 x AVDD  
|                  |     |             |         | 1 = 0.75 x AVDD                                                              |                   |
| 7:4              |     |             | 0000    | Reserved                                                                     |                   |
| 3                |     | AUXMODE     | 0       | Auxiliary Input Mode  
|                  |     |             |         | 0 = inverting buffer  
|                  |     |             |         | 1 = mixer (on-chip input resistor bypassed)                                  |                   |
| 2                |     | AUX2INPPGA  | 0       | Select AUX amplifier output as input PGA signal source.  
|                  |     |             |         | 0=AUX not connected to input PGA  
|                  |     |             |         | 1=AUX connected to input PGA amplifier negative terminal.                   |                   |
| 1                |     | MICN2INPPGA | 1       | Connect MICN to input PGA negative terminal.  
|                  |     |             |         | 0=MICN not connected to input PGA  
|                  |     |             |         | 1=MICN connected to input PGA amplifier negative terminal.                  |                   |
| 0                |     | MICP2INPPGA | 1       | Connect input PGA amplifier positive terminal to MICP or VMID.  
|                  |     |             |         | 0 = input PGA amplifier positive terminal connected to VMID  
|                  |     |             |         | 1 = input PGA amplifier positive terminal connected to MICP through variable resistor string |                   |
| 45 (2Dh)         | 8   |             | 0       | Reserved                                                                     |                   |
| 7                |     | INPPGAZC    | 0       | Input PGA zero cross enable:  
|                  |     |             |         | 0=Update gain when gain register changes  
|                  |     |             |         | 1=Update gain on 1st zero cross after gain register write.                 |                   |
| 6                |     | INPPGAMUTE  | 0       | Mute control for input PGA:  
|                  |     |             |         | 0=Input PGA not muted, normal operation  
|                  |     |             |         | 1=Input PGA muted (and disconnected from the following input BOOST stage). |                   |
| 5:0              |     | INPPGAVOL   | 010000  | Input PGA volume  
|                  |     |             |         | 000000 = -12dB  
|                  |     |             |         | 000001 = -11.25db  
|                  |     |             |         | 010000 = 0dB  
|                  |     |             |         | 111111 = 35.25dB                                                       |                   |
| 47 (2Fh)         | 8   | PGABOOST    | 0       | Input Boost  
|                  |     |             |         | 0 = PGA output has +0dB gain through input BOOST stage.  
|                  |     |             |         | 1 = PGA output has +20dB gain through input BOOST stage.                  |                   |
| 7                |     |             | 0       | Reserved                                                                     |                   |
| 6:4              |     | MICP2BOOSTVOL | 000   | Controls the MICP pin to the input boost stage (NB, when using this path set MICP2INPPGA=0):  
|                  |     |             |         | 000=Path disabled (disconnected)  
|                  |     |             |         | 001=-12dB gain through boost stage  
|                  |     |             |         | 010=-9dB gain through boost stage  
|                  |     |             |         | 100=-6dB gain through boost stage  
<p>|                  |     |             |         | 111=+6dB gain through boost stage                                             |                   |
| 3                |     |             | 0       | Reserved                                                                     |                   |</p>
<table>
<thead>
<tr>
<th>REGISTER ADDRESS</th>
<th>BIT</th>
<th>LABEL</th>
<th>DEFAULT</th>
<th>DESCRIPTION</th>
<th>REFER TO</th>
</tr>
</thead>
<tbody>
<tr>
<td>2:0</td>
<td></td>
<td>AUX2BOOSTVOL</td>
<td>000</td>
<td>Controls the auxiliary amplifier to the input boost stage: 000=Path disabled (disconnected) 001=–12dB gain through boost stage 010=–9dB gain through boost stage ... 111=+6dB gain through boost stage</td>
<td>Input Signal Path</td>
</tr>
<tr>
<td>49 (31h)</td>
<td>8:4</td>
<td></td>
<td>00000</td>
<td>Reserved</td>
<td></td>
</tr>
<tr>
<td>3</td>
<td></td>
<td>MONOBOOST</td>
<td>0</td>
<td>Mono output boost stage control (see Table 37 for details) 0=No boost (output is inverting buffer) 1=1.5x gain boost</td>
<td>Analogue Outputs</td>
</tr>
<tr>
<td>2</td>
<td></td>
<td>SPKBOOST</td>
<td>0</td>
<td>Speaker output boost stage control (see Table 37 for details) 0=No boost (outputs are inverting buffers) 1=1.5x gain boost</td>
<td>Analogue Outputs</td>
</tr>
<tr>
<td>1</td>
<td></td>
<td>TSDEN</td>
<td>1</td>
<td>Thermal Shutdown Enable 0 : thermal shutdown disabled 1 : thermal shutdown enabled</td>
<td>Output Switch</td>
</tr>
<tr>
<td>0</td>
<td></td>
<td>VROI</td>
<td>0</td>
<td>VREF (AVDD/2 or 1.5xAVDD/2) to analogue output resistance 0: approx 1kΩ 1: approx 30 kΩ</td>
<td>Analogue Outputs</td>
</tr>
<tr>
<td>50 (32h)</td>
<td>8:6</td>
<td></td>
<td>000</td>
<td>Reserved</td>
<td></td>
</tr>
<tr>
<td>5</td>
<td></td>
<td>AUX2SPK</td>
<td>0</td>
<td>Output of auxiliary amplifier to speaker mixer input 0 = not selected 1 = selected</td>
<td>Analogue Outputs</td>
</tr>
<tr>
<td>4:2</td>
<td></td>
<td></td>
<td>000</td>
<td>Reserved</td>
<td></td>
</tr>
<tr>
<td>1</td>
<td></td>
<td>BYP2SPK</td>
<td>0</td>
<td>Bypass path (output of input boost stage) to speaker mixer input 0 = not selected 1 = selected</td>
<td>Analogue Outputs</td>
</tr>
<tr>
<td>0</td>
<td></td>
<td>DAC2SPK</td>
<td>0</td>
<td>Output of DAC to speaker mixer input 0 = not selected 1 = selected</td>
<td>Analogue Outputs</td>
</tr>
<tr>
<td>54 (36h)</td>
<td>8</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>7</td>
<td></td>
<td>SPKZC</td>
<td>0</td>
<td>Speaker Volume control zero cross enable: 1 = Change gain on zero cross only 0 = Change gain immediately</td>
<td>Analogue Outputs</td>
</tr>
<tr>
<td>6</td>
<td></td>
<td>SPKMUTE</td>
<td>0</td>
<td>Speaker output mute enable 0=Speaker output enabled 1=Speaker output muted (VMIDOP)</td>
<td>Analogue Outputs</td>
</tr>
<tr>
<td>5:0</td>
<td></td>
<td>SPKVOL</td>
<td>111001</td>
<td>Speaker Volume Adjust 111111 = +6dB 111110 = +5dB ... (1.0 dB steps) 111001=0dB ... 000000=–57dB</td>
<td>Analogue Outputs</td>
</tr>
<tr>
<td>56 (38h)</td>
<td>8:7</td>
<td></td>
<td>0</td>
<td>Reserved</td>
<td></td>
</tr>
<tr>
<td>REGISTER ADDRESS</td>
<td>BIT</td>
<td>LABEL</td>
<td>DEFAULT</td>
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<td>REFER TO</td>
</tr>
<tr>
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<td>-----</td>
<td>-------------</td>
<td>---------</td>
<td>-----------------------------------------------------------------------------</td>
<td>---------------------</td>
</tr>
<tr>
<td>6</td>
<td>0</td>
<td>MONOMUTE</td>
<td>MONOOUT</td>
<td>MONOOUT Mute Control</td>
<td>analogue Outputs</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>0</td>
<td>0=No mute</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>1</td>
<td>1=Output muted. During mute the mono output will output VMID which can be</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>used as a DC reference for a headphone out.</td>
<td></td>
</tr>
<tr>
<td>5:3</td>
<td>0</td>
<td>Reserved</td>
<td>Reserved</td>
<td>Reserved</td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>0</td>
<td>AUX2MONO</td>
<td>0</td>
<td>Output of Auxiliary amplifier to mono mixer input:</td>
<td>analogue Outputs</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>0</td>
<td>0 = not selected</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>1</td>
<td>1 = selected</td>
<td></td>
</tr>
<tr>
<td>1</td>
<td>0</td>
<td>BYP2MONO</td>
<td>0</td>
<td>Bypass path (output of input boost stage) to mono mixer input</td>
<td>analogue Outputs</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>0</td>
<td>0 = non selected</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>1</td>
<td>1 = selected</td>
<td></td>
</tr>
<tr>
<td>0</td>
<td>0</td>
<td>DAC2MONO</td>
<td>0</td>
<td>Output of DAC to mono mixer input</td>
<td>analogue Outputs</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>0</td>
<td>0 = not selected</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>1</td>
<td>1 = selected</td>
<td></td>
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</table>
### DIGITAL FILTER CHARACTERISTICS

<table>
<thead>
<tr>
<th>PARAMETER</th>
<th>TEST CONDITIONS</th>
<th>MIN</th>
<th>TYP</th>
<th>MAX</th>
<th>UNIT</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>ADC Filter</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Passband</td>
<td>+/- 0.025dB</td>
<td>0</td>
<td>0.454fs</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>-6dB</td>
<td>0.5fs</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Passband Ripple</td>
<td>+/- 0.025dB</td>
<td>0.546fs</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Stopband</td>
<td>0.546fs</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Stopband Attenuation</td>
<td>f &gt; 0.546fs</td>
<td>-60</td>
<td></td>
<td></td>
<td>dB</td>
</tr>
<tr>
<td>Group Delay</td>
<td>21/fs</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

| **ADC High-Pass Filter** | | | | |
| High-Pass Filter Corner Frequency | -3dB | 3.7 | | Hz |
|                               | -0.5dB | 10.4 | | |
|                               | -0.1dB | 21.6 | | |

| **DAC Filter** | | | | |
| Passband      | +/- 0.035dB     | 0    | 0.454fs |      |      |
|               | -6dB            | 0.5fs|      |      |      |
| Passband Ripple | +/- 0.035dB     | 0.546fs|      |      |      |
| Stopband      | 0.546fs         |      |      |      |      |
| Stopband Attenuation | f > 0.546fs     | -80  |      |      | dB   |
| Group Delay   | 29/fs           |      |      |      |      |

Table 63 Digital Filter Characteristics

### TERMINOLOGY

1. **Stop Band Attenuation (dB)** – the degree to which the frequency spectrum is attenuated (outside audio band)
2. **Pass-band Ripple** – any variation of the frequency response in the pass-band region
3. Note that this delay applies only to the filters and does not include additional delays through other digital circuits. See Table 64 for the total delay.

<table>
<thead>
<tr>
<th>PARAMETER</th>
<th>TEST CONDITIONS</th>
<th>MIN</th>
<th>TYP</th>
<th>MAX</th>
<th>UNIT</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>ADC Path Group Delay</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Total Delay (ADC analogue input to digital audio interface output)</td>
<td>EQ disabled</td>
<td>26/fs</td>
<td>28/fs</td>
<td>30/fs</td>
<td>fs</td>
</tr>
<tr>
<td></td>
<td>EQ enabled</td>
<td>27/fs</td>
<td>29/fs</td>
<td>31/fs</td>
<td>fs</td>
</tr>
</tbody>
</table>

| **DAC Path Group Delay** | | | | |
| Total Delay (Audio interface input to DAC analogue output) | EQ disabled | 34/fs | 36/fs | 38/fs | fs |
|                     | EQ enabled      | 35/fs| 37/fs| 39/fs| fs |

Table 64 Total Group Delay

**Notes:**

1. Wind noise filter is disabled.
DAC FILTER RESPONSES

Figure 36 DAC Digital Filter Frequency Response

Figure 37 DAC Digital Filter Ripple

ADC FILTER RESPONSES

Figure 38 ADC Digital Filter Frequency Response

Figure 39 ADC Digital Filter Ripple
DE-EMPHASIS FILTER RESPONSES

Figure 40  De-emphasis Frequency Response (32kHz)

Figure 41  De-emphasis Error (32kHz)

Figure 42  De-emphasis Frequency Response (44.1kHz)

Figure 43  De-emphasis Error (44.1kHz)

Figure 44  De-emphasis Frequency Response (48kHz)

Figure 45  De-emphasis Error (48kHz)
HIGH-PASS FILTER

The WM8974 has a selectable digital high-pass filter in the ADC filter path. This filter has two modes, audio and applications. In audio mode the filter is a 1st order IIR with a cut-off of around 3.7Hz. In applications mode the filter is a 2nd order high-pass filter with a selectable cut-off frequency.

Figure 46 ADC High-pass Filter Response, HPFAPP=0

Figure 47 ADC High-pass Filter Responses (48kHz), HPFAPP=1, all cut-off settings shown.

Figure 48 ADC High-pass Filter Responses (24kHz), HPFAPP=1, all cut-off settings shown.

Figure 49 ADC High-pass Filter Responses (12kHz), HPFAPP=1, all cut-off settings shown.
The WM8974 has a 5-band equaliser which can be applied to either the ADC path or the DAC path. The plots from Figure 50 to Figure 63 show the frequency responses of each filter with a sampling frequency of 48kHz, firstly showing the different cut-off/centre frequencies with a gain of ±12dB, and secondly a sweep of the gain from -12dB to +12dB for the lowest cut-off/centre frequency of each filter.

Figure 50 EQ Band 1 – Low Frequency Shelf Filter Cut-offs
Figure 51 EQ Band 1 – Gains for Lowest Cut-off Frequency
Figure 52 EQ Band 2 – Peak Filter Centre Frequencies, EQ2BW=0
Figure 53 EQ Band 2 – Peak Filter Gains for Lowest Cut-off Frequency, EQ2BW=0
Figure 54 EQ Band 2 – EQ2BW=0, EQ2BW=1
Figure 55  EQ Band 3 – Peak Filter Centre Frequencies, EQ3BW=0

Figure 56  EQ Band 3 – Peak Filter Gains for Lowest Cut-off Frequency, EQ3BW=0

Figure 57  EQ Band 3 – EQ3BW=0, EQ3BW=1
Figure 58  EQ Band 4 – Peak Filter Centre Frequencies, EQ3BW=0

Figure 59  EQ Band 4 – Peak Filter Gains for Lowest Cut-off Frequency, EQ4BW=0

Figure 60  EQ Band 4 – EQ3BW=0, EQ3BW=1

Figure 61  EQ Band 5 – High Frequency Shelf Filter Cut-offs

Figure 62  EQ Band 5 – Gains for Lowest Cut-off Frequency
Figure 63 shows the result of having the gain set on more than one channel simultaneously. The blue traces show each band (lowest cut-off/centre frequency) with ±12dB gain. The red traces show the cumulative effect of all bands with +12dB gain and all bands -12dB gain, with EQxBW=0 for the peak filters.

Figure 63  Cumulative Frequency Boost/Cut
Figure 64  Recommended External Components

Notes:
1. AGND and GND should be connected as close to the WM8974 as possible.
2. C1, C2, C3, C4, C8 and C9 should be positioned as close to WM8974 as possible.
3. Capacitor types should be carefully chosen. Capacitors with very low ESR are recommended for optimum performance.
4. R3 can be populated with other values to remove common mode noise on the microphone.
# PACKAGE DIAGRAM

**FL: 24 PIN QFN PLASTIC PACKAGE** 4 x 4 x 0.9 mm BODY, 0.50 mm LEAD PITCH

**DIMENSIONS (mm)**

<table>
<thead>
<tr>
<th>Symbol</th>
<th>MIN</th>
<th>NOM</th>
<th>MAX</th>
<th>NOTE</th>
</tr>
</thead>
<tbody>
<tr>
<td>A</td>
<td>0.80</td>
<td>0.85</td>
<td>0.90</td>
<td></td>
</tr>
<tr>
<td>A1</td>
<td>0</td>
<td>0.035</td>
<td>0.05</td>
<td></td>
</tr>
<tr>
<td>A3</td>
<td>0.203 REF</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>b</td>
<td>0.20</td>
<td>0.25</td>
<td>0.30</td>
<td>1</td>
</tr>
<tr>
<td>D</td>
<td>4.00 BSC</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>D2</td>
<td>2.40</td>
<td>2.50</td>
<td>2.60</td>
<td>2</td>
</tr>
<tr>
<td>E</td>
<td>4.00 BSC</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>E2</td>
<td>2.40</td>
<td>2.50</td>
<td>2.60</td>
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</tr>
<tr>
<td>e</td>
<td>0.50 BSC</td>
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</tr>
<tr>
<td>G</td>
<td>0.20</td>
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<td></td>
<td></td>
</tr>
<tr>
<td>H</td>
<td>0.10</td>
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<td></td>
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</tr>
<tr>
<td>L</td>
<td>0.35</td>
<td>0.40</td>
<td>0.45</td>
<td></td>
</tr>
<tr>
<td>T</td>
<td>0.103</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>W</td>
<td>0.15</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Tolerances of Form and Position**

aaa | 0.10
bbb | 0.10
ccc | 0.10

**REF:** JEDEC, MO-220, VARIATION VGGD-8.

**NOTES:**
1. DIMENSION b APPLIES TO METALLIZED TERMINAL AND IS MEASURED BETWEEN 0.15 mm AND 0.30 mm FROM TERMINAL TIP.
2. FALLS WITHIN JEDEC, MO-220, VARIATION VGGD-8.
3. ALL DIMENSIONS ARE IN MILLIMETRES.
4. THE TERMINAL #1 IDENTIFIER AND TERMINAL NUMBERING CONVENTION SHALL CONFORM TO JEDEC 95-1 SPP-002.
5. COPLANARITY APPLIES TO THE EXPOSED HEAT SINK SLUG AS WELL AS THE TERMINALS.
6. REFER TO APPLICATIONS NOTE WAN.0118 FOR FURTHER INFORMATION REGARDING PCB FOOTPRINTS AND QFN PACKAGE SOLDERING.
7. THIS DRAWING IS SUBJECT TO CHANGE WITHOUT NOTICE.
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Contacting Cirrus Logic Support
For all product questions and inquiries, contact a Cirrus Logic Sales Representative.
To find one nearest you, go to www.cirrus.com.

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## REVISION HISTORY

<table>
<thead>
<tr>
<th>DATE</th>
<th>REV</th>
<th>ORIGINATOR</th>
<th>CHANGES</th>
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<tbody>
<tr>
<td>26/09/11</td>
<td>4.6</td>
<td>JMacD</td>
<td>Order codes changed from WM8974GEFL/V and WM8974GEFL/RV to WM8974C2GEFL/V and WM8974C2GEFL/RV to reflect change to copper wire bonding.</td>
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<tr>
<td>26/09/11</td>
<td>4.6</td>
<td>JMacD</td>
<td>Package diagram updated to DM102.C</td>
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<tr>
<td>12/08/16</td>
<td>4.7</td>
<td>PH</td>
<td>MICBIAS voltage (MBVSEL=1) updated to 0.75 x AVDD.</td>
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