

Stereo CODEC with Speaker Driver

DESCRIPTION

The WM8976 is a low power, high quality CODEC designed for portable applications such as multimedia phone, digital still camera or digital camcorder.

The device integrates a preamp for differential microphone, and includes drivers for speakers, headphone and differential or stereo line output. External component requirements are reduced as no separate microphone or headphone amplifiers are required.

Advanced on-chip digital signal processing includes a 5-band equaliser, a mixed signal Automatic Level Control for the microphone or line input through the ADC as well as a purely digital limiter function for record or playback. Additional digital filtering options are available in the ADC path, to cater for application filtering such as 'wind noise reduction'.

The WM8976 digital audio interface can operate as a master or a slave. An internal PLL can generate all required audio clocks for the CODEC from common reference clock frequencies, such as 12MHz and 13MHz.

The WM8976 operates at analogue supply voltages from 2.5V to 3.3V, although the digital core can operate at voltages down to 1.71V to save power. The speaker outputs and OUT3/4 line outputs can run from a 5V supply if increased output power is required. Individual sections of the chip can also be powered down under software control.

FEATURES

Stereo CODEC:

- DAC SNR 98dB, THD -84dB ('A' weighted @ 48kHz)
- ADC SNR 95dB, THD -84dB ('A' weighted @ 48kHz)
- On-chip Headphone Driver with 'capless' option
 - 40mW per channel into 16Ω / 3.3V SPKVDD
- 1W output power into 8Ω BTL speaker / 5V SPKVDD
 - Capable of driving piezo speakers
 - Stereo speaker drive configuration

Mic Preamps:

- Differential or single-ended microphone interfaces
 - Programmable preamp gain
 - Pseudo differential input with common mode rejection
 - Programmable ALC / Noise Gate in ADC path
- Low-noise bias supplied for electret microphone

Other Features:

- Enhanced 3-D function for improved stereo separation
- Digital playback limiter
- 5-band Equaliser (record or playback)
- Programmable ADC High Pass Filter (wind noise reduction)
- Programmable ADC Notch Filter
- Aux inputs for stereo analogue input signals or 'beep'
- On-chip PLL supporting 12, 13, 19.2MHz and other clocks
- Support for 8, 11.025, 12, 16, 22.05, 24, 32, 44.1 and 48kHz sample rates
- Low power, low voltage
 - 2.5V to 3.6V (digital: 1.71V to 3.6V)
- 5x5mm 32-lead QFN package

APPLICATIONS

- Stereo Camcorder or DSC
- Multimedia Phone

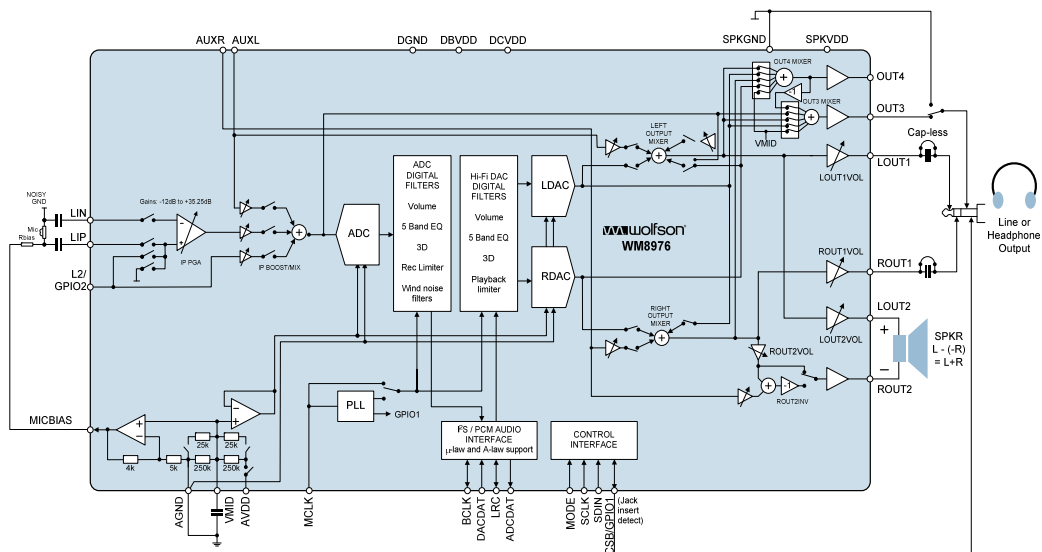
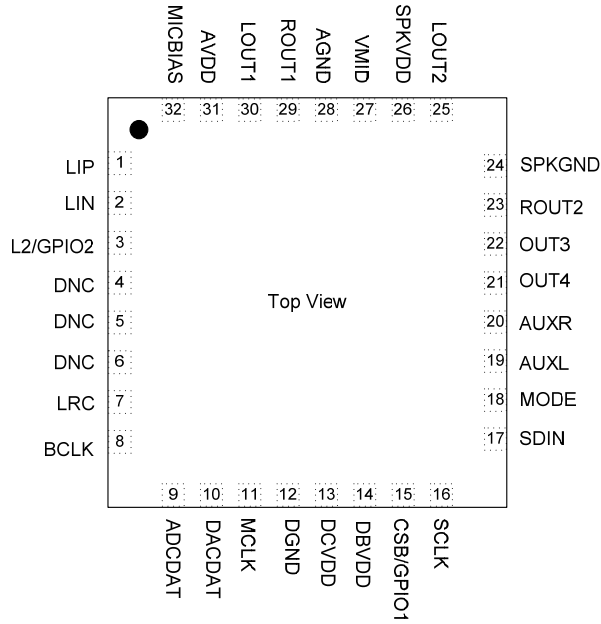


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PIN CONFIGURATION



ORDERING INFORMATION

| ORDER CODE | TEMPERATURE RANGE | PACKAGE | MOISTURE SENSITIVITY LEVEL | PEAK SOLDERING TEMPERATURE |
|----------------|-------------------|--|----------------------------|----------------------------|
| WM8976CGEFL/V | -25°C to +85°C | 32-lead QFN (5 x 5 mm) (Pb-free) | MSL3 | 260°C |
| WM8976CGEFL/RV | -25°C to +85°C | 32-lead QFN (5 x 5 mm) (Pb-free, tape and reel) | MSL3 | 260°C |

Note:

Reel quantity = 3,500

PIN DESCRIPTION

| PIN | NAME | TYPE | DESCRIPTION |
|-----|-----------|------------------------|---|
| 1 | LIP | Analogue input | Mic Pre-amp positive input |
| 2 | LIN | Analogue input | Mic Pre-amp negative input |
| 3 | L2/GPIO2 | Analogue input | Line input/secondary mic pre-amp positive input/GPIO2 pin |
| 4 | DNC | Do not connect | Leave this pin floating |
| 5 | DNC | Do not connect | Leave this pin floating |
| 6 | DNC | Do not connect | Leave this pin floating |
| 7 | LRC | Digital Input / Output | DAC and ADC Sample Rate Clock |
| 8 | BCLK | Digital Input / Output | Digital Audio Port Clock |
| 9 | ADCDAT | Digital Output | ADC Digital Audio Data Output |
| 10 | DACDAT | Digital Input | DAC Digital Audio Data Input |
| 11 | MCLK | Digital Input | Master Clock Input |
| 12 | DGND | Supply | Digital ground |
| 13 | DCVDD | Supply | Digital core logic supply |
| 14 | DBVDD | Supply | Digital buffer (I/O) supply |
| 15 | CSB/GPIO1 | Digital Input / Output | 3-Wire Control Interface Chip Select / GPIO1 pin |
| 16 | SCLK | Digital Input | 3-Wire Control Interface Clock Input / 2-Wire Control Interface Clock Input |
| 17 | SDIN | Digital Input / Output | 3-Wire Control Interface Data Input / 2-Wire Control Interface Data Input |
| 18 | MODE | Digital Input | Control Interface Selection |
| 19 | AUXL | Analogue input | Left Auxiliary input |
| 20 | AUXR | Analogue input | Right Auxiliary input |
| 21 | OUT4 | Analogue Output | Buffered midrail Headphone pseudo-ground, or Right line output or MONO mix output |
| 22 | OUT3 | Analogue Output | Buffered midrail Headphone pseudo-ground, or Left line output |
| 23 | ROUT2 | Analogue Output | Second right output, or BTL speaker driver positive output |
| 24 | SPKGND | Supply | Speaker ground (feeds speaker amp and OUT3/OUT4) |
| 25 | LOUT2 | Analogue Output | Second left output, or BTL speaker driver negative output |
| 26 | SPKVDD | Supply | Speaker supply (feed speaker amp only) |
| 27 | VMID | Reference | Decoupling for ADC and DAC reference voltage |
| 28 | AGND | Supply | Analogue ground (feeds ADC and DAC) |
| 29 | ROUT1 | Analogue Output | Headphone or Line Output Right |
| 30 | LOUT1 | Analogue Output | Headphone or Line Output Left |
| 31 | AVDD | Supply | Analogue supply (feeds ADC and DAC) |
| 32 | MICBIAS | Analogue Output | Microphone Bias |

Note:

It is recommended that the QFN ground paddle should be connected to analogue ground on the application PCB. Refer to the application note WAN_0118 on "Guidelines on How to Use QFN Packages and Create Associated PCB Footprints"

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.

Wolfson 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.

| CONDITION | MIN | MAX |
|---|------------|------------|
| DBVDD, DCVDD, AVDD supply voltages | -0.3V | +4.5V |
| SPKVDD supply voltage | -0.3V | +7V |
| Voltage range digital inputs | DGND -0.3V | DVDD +0.3V |
| Voltage range analogue inputs | AGND -0.3V | AVDD +0.3V |
| Operating temperature range, T _A | -25°C | +85°C |
| Storage temperature after soldering | -65°C | +150°C |

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, i.e. not internally connected.

RECOMMENDED OPERATING CONDITIONS

| PARAMETER | SYMBOL | TEST CONDITIONS | MIN | TYP | MAX | UNIT |
|-------------------------------|-----------------------|-----------------|-------------------|-----|-----|------|
| Digital supply range (Core) | DCVDD | | 1.71 ¹ | | 3.6 | V |
| Digital supply range (Buffer) | DBVDD | | 1.71 ² | | 3.6 | V |
| Analogue core supply range | AVDD | | 2.5 | | 3.6 | V |
| Analogue output supply range | SPKVDD | | 2.5 | | 5.5 | V |
| Ground | DGND, AGND, SPKGND | | | 0 | | V |

Notes:

1. When using the PLL, DCVDD must not be less than 1.9V.
2. DBVDD must be greater than or equal to DCVDD.
3. Analogue supplies have to be \geq to digital supplies.
4. In non-boosted mode, SPKVDD should = AVDD, if boosted SPKVDD should be $\geq 1.5x$ AVDD.

ELECTRICAL CHARACTERISTICS

Test Conditions

DCVDD=1.8V, AVDD=DBVDD=SPKVDD= 3.3V, T_A = +25°C, 1kHz signal, fs = 48kHz, 24-bit audio data unless otherwise stated.

| PARAMETER | SYMBOL | TEST CONDITIONS | MIN | TYP | MAX | UNIT |
|--|--------------------|---|--|---------------|-------|-------------|
| Microphone Preamp Inputs (LIP, LIN) | | | | | | |
| Full-scale Input Signal Level – note this changes in proportion to AVDD (Note 1) | V _{INFS} | PGABOOST = 0dB INPPGAVOL = 0dB | | 1.0 0 | | Vrms dBV |
| Mic PGA equivalent input noise | At 35.25dB gain | 0 to 20kHz | | 150 | | uV |
| Input resistance | R _{MICIN} | Gain set to 35.25dB | | 1.6 | | kΩ |
| | R _{MICIN} | Gain set to 0dB | | 47 | | kΩ |
| | R _{MICIN} | Gain set to -12dB | | 75 | | kΩ |
| | R _{MICIP} | LIP2INPPGA = 1 | | 94 | | kΩ |
| | C _{MICIN} | | | 10 | | pF |
| MIC Programmable Gain Amplifier (PGA) | | | | | | |
| Maximum Programmable Gain | | | | 35.25 | | dB |
| Minimum Programmable Gain | | | | -12 | | dB |
| Programmable Gain Step Size | | Guaranteed monotonic | | 0.75 | | dB |
| Mute Attenuation | | | | 120 | | dB |
| Selectable Input Gain Boost (0/+20dB) | | | | | | |
| Gain Boost on PGA input | | Boost disabled | | 0 | | dB |
| | | Boost enabled | | 20 | | dB |
| Maximum Gain from AUXL or L2 input to boost/mixer | | | | +6 | | dB |
| Minimum Gain from AUXL or L2 input to boost/mixer | | | | -12 | | dB |
| Gain step size to boost/mixer | | Guaranteed monotonic | | 3 | | dB |
| Auxiliary Analogue Inputs (AUXL, AUXR) | | | | | | |
| Full-scale Input Signal Level (0dB) – note this changes in proportion to AVDD | V _{INFS} | | | AVDD/3.3 0 | | Vrms dBV |
| Input Capacitance | C _{MICIN} | | | 10 | | pF |
| Automatic Level Control (ALC) | | | | | | |
| Target Record Level | | | -22.5 | | -1.5 | dB |
| Programmable gain | | | -12 | | 35.25 | dB |
| Gain Hold Time (Note 3,5) | t _{HOLD} | MCLK = 12.288MHz (Note 3) | 0, 2.67, 5.33, 10.67, ... , 43691 (time doubles with each step) | | | ms |
| Gain Ramp-Up (Decay) Time (Note 4,5) | t _{DCY} | ALCMODE=0 (ALC), MCLK=12.288MHz (Note 3) | 3.3, 6.6, 13.1, ... , 3360 (time doubles with each step) | | | ms |
| | | ALCMODE=1 (limiter), MCLK=12.288MHz (Note 3) | 0.73, 1.45, 2.91, ... , 744 (time doubles with each step) | | | |
| Gain Ramp-Down (Attack) Time (Note 4,5) | t _{ATK} | ALCMODE=0 (ALC), MCLK=12.288MHz (Note 3) | 0.83, 1.66, 3.33, ... , 852 (time doubles with each step) | | | ms |
| | | ALCMODE=1 (limiter), MCLK=12.288MHz (Note 3) | 0.18, 0.36, 0.73, ... , 186 (time doubles with each step) | | | |
| Mute Attenuation | | | | 120 | | dB |

Test ConditionsDCVDD=1.8V, AVDD=DBVDD=SPKVDD= 3.3V, T_A = +25°C, 1kHz signal, fs = 48kHz, 24-bit audio data unless otherwise stated.

| PARAMETER | SYMBOL | TEST CONDITIONS | MIN | TYP | MAX | UNIT |
|---|----------------|---|-----|------------------|-----|---------|
| Analogue to Digital Converter (ADC) | | | | | | |
| Signal to Noise Ratio (Note 6) | SNR | A-weighted, 0dB gain | 85 | 95 | | dB |
| Total Harmonic Distortion (Note 7) | THD | -3dBFS input | | -84 | -74 | dB |
| Digital to Analogue Converter (DAC) to Line-Out (LOUT1, ROUT1 with 10kΩ / 50pF load) | | | | | | |
| Full-scale output | | PGA gains set to 0dB, OUT34BOOST=0 | | AVDD/3.3 | | Vrms |
| | | PGA gains set to 0dB, OUT34BOOST=1 | | 1.5x (AVDD/3.3) | | |
| Signal to Noise Ratio (Note 6) | SNR | A-weighted | 90 | 98 | | dB |
| Total Harmonic Distortion (Note 7) | THD | R _L = 10kΩ full-scale signal | | -84 | -76 | dB |
| Channel Separation (Note 9) | | 1kHz signal | | 110 | | dB |
| Output Mixers (LMX1, RMX1) | | | | | | |
| Maximum PGA gain into mixer | | | | +6 | | dB |
| Minimum PGA gain into mixer | | | | -15 | | dB |
| PGA gain step into mixer | | Guaranteed monotonic | | 3 | | dB |
| Analogue Outputs (LOUT1, ROUT1, LOOUT2, ROUT2) | | | | | | |
| Maximum Programmable Gain | | | | +6 | | dB |
| Minimum Programmable Gain | | | | -57 | | dB |
| Programmable Gain step size | | Guaranteed monotonic | | 1 | | dB |
| Mute attenuation | | 1kHz, full scale signal | | 85 | | dB |
| Headphone Output (LOUT1, ROUT1 with 32Ω load) | | | | | | |
| 0dB full scale output voltage | | | | AVDD/3.3 | | Vrms |
| Signal to Noise Ratio | SNR | A-weighted | | 102 | | dB |
| Total Harmonic Distortion | THD | R _L = 16Ω, P _o =20mW AVDD=3.3V | | 0.003 -92 | | % dB |
| | | R _L = 32 Ω, P _o =20mW AVDD=3.3V | | 0.008 - 82 | | % dB |
| Speaker Output (LOUT2, ROUT2 with 8Ω bridge tied load, INVROUT2=1) | | | | | | |
| Full scale output voltage, 0dB gain. (Note 9) | | SPKBOOST=0 | | SPKVDD/3.3 | | Vrms |
| | | SPKBOOST=1 | | (SPKVDD/3.3)*1.5 | | |
| Output Power | P _O | Output power is very closely correlated with THD; see below | | | | |
| Total Harmonic Distortion | THD | P _O =200mW, R _L = 8Ω, SPKVDD=3.3V | | 0.04 -68 | | % dB |
| | | P _O =320mW, R _L = 8Ω, SPKVDD=3.3V | | 1.0 -40 | | % dB |
| | | P _O =500mW, R _L = 8Ω, SPKVDD=5V | | 0.02 -74 | | % dB |
| | | P _O =860mW, R _L = 8Ω, SPKVDD=5V | | 1.0 -40 | | % dB |
| Signal to Noise Ratio | SNR | SPKVDD=3.3V, R _L = 8Ω | | 90 | | dB |
| | | SPKVDD=5V, R _L = 8Ω | | 90 | | dB |
| Power Supply Rejection Ratio (50Hz-22kHz) | PSRR | R _L = 8Ω BTL | | 80 | | dB |
| | | R _L = 8Ω BTL SPKVDD=5V (boost) | | 69 | | dB |

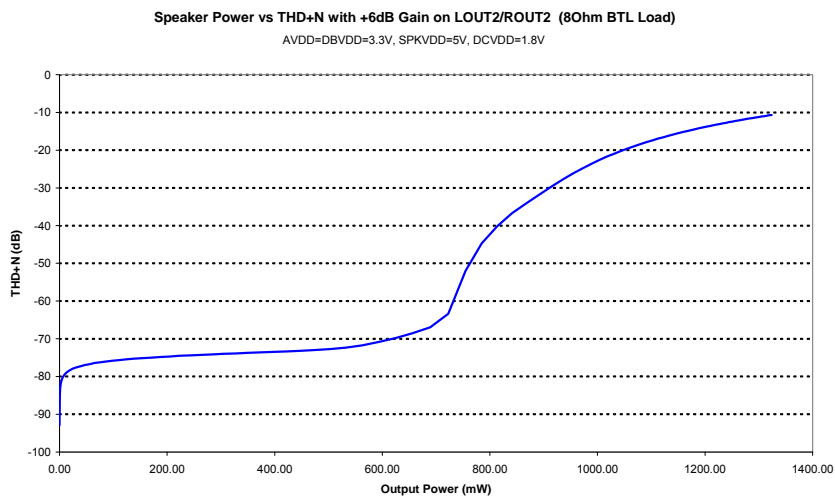
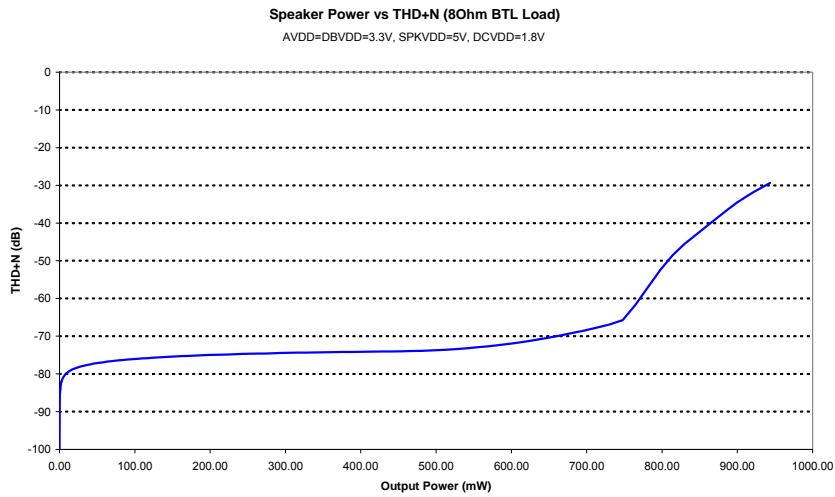
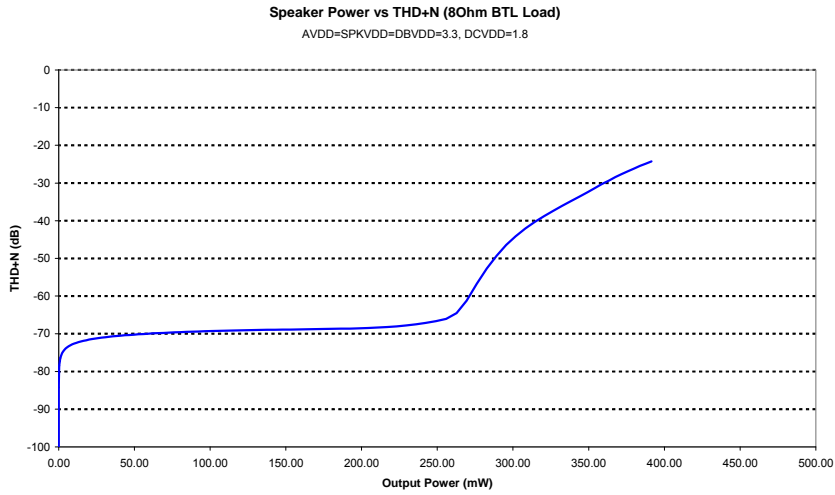
Test ConditionsDCVDD=1.8V, AVDD=DBVDD=SPKVDD= 3.3V, T_A = +25°C, 1kHz signal, fs = 48kHz, 24-bit audio data unless otherwise stated.

| PARAMETER | SYMBOL | TEST CONDITIONS | MIN | TYP | MAX | UNIT |
|--|----------------------|---|-----------|------------------|-----------|--------|
| OUT3/OUT4 outputs (with 10kΩ / 50pF load) | | | | | | |
| Full-scale output voltage, 0dB gain (Note 9) | | OUT3BOOST=0/ OUT4BOOST=0 | | SPKVDD/3.3 | | Vrms |
| | | OUT3BOOST=1 OUT4BOOST=1 | | (SPKVDD/3.3)*1.5 | | Vrms |
| Signal to Noise Ratio (Note 6) | SNR | A-weighted | | 98 | | dB |
| Total Harmonic Distortion (Note 7) | THD | R _L = 10 kΩ full-scale signal | | -84 | | dB |
| Channel Separation (Note 8) | | 1kHz signal | | 100 | | dB |
| Power Supply Rejection Ratio (50Hz-22kHz) | PSRR | R _L = 10kΩ | | 52 | | dB |
| | | R _L = 10kΩ SPKVDD=5V (boost) | | 56 | | dB |
| Microphone Bias | | | | | | |
| Bias Voltage | V _{MICBIAS} | MBVSEL=0 | | 0.9*AVDD | | V |
| | | MBVSEL=1 | | 0.65*AVDD | | V |
| Bias Current Source | I _{MICBIAS} | | | | 3 | mA |
| Output Noise Voltage | V _n | 1K to 20kHz | | 15 | | nV/√Hz |
| Digital Input / Output | | | | | | |
| Input HIGH Level | V _{IH} | | 0.7×DBVDD | | | V |
| Input LOW Level | V _{IL} | | | | 0.3×DBVDD | V |
| Output HIGH Level | V _{OH} | I _{OL} =1mA | 0.9×DBVDD | | | V |
| Output LOW Level | V _{OL} | I _{OH} =1mA | | | 0.1×DBVDD | V |
| Input capacitance | | | | 10 | | pF |
| Input leakage | | | | 50 | | pA |

TERMINOLOGY

- Input level to LIP is limited to a maximum of -3dB or THD+N performance will be reduced.
- Note when BEEP path is not enabled then AUXL and AUXR have the same input impedances.
- 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.
- Ramp-up and Ramp-Down times are defined as the time it takes for the PGA to sweep across 90% of its gain range.
- All hold, ramp-up and ramp-down times scale proportionally with MCLK
- 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).
- THD+N (dB) – THD+N is a ratio, of the rms values, of (Noise + Distortion)/Signal.
- Channel Separation (dB) – Also known as Cross-Talk. This is a measure of the amount one channel is isolated from the other. Measured by applying a full scale signal to one channel input and measuring the level of signal apparent at the other channel output.
- The maximum output voltage can be limited by the speaker power supply. If OUT3BOOST, OUT4BOOST or SPKBOOST is set then SPKVDD should be 1.5xAVDD to prevent clipping taking place in the output stage (when PGA gains are set to 0dB).

SPEAKER OUTPUT THD VERSUS POWER



POWER CONSUMPTION

Typical current consumption for various scenarios is shown below.

| MODE | AVDD (3.0V) (mA) | DCVDD (1.8V) (mA) | DBVDD ¹ (3.0V) (mA) | TOTAL POWER (mW) |
|--|------------------------|-------------------------|--------------------------------------|------------------------|
| Off | 0.04 ³ | 0.0008 | <0.0001 | 0.12 |
| Sleep (VREF maintained, no clocks) | 0.04 | 0.0008 | <0.0001 | 0.12 |
| MIC Record (8kHz) ² | 4.1 | 1.0 | 0.001 | 14.1 |
| Stereo 16Ω HP Playback (48kHz, quiescent) ² | 3.3 | 6.2 | 0.004 | 21.1 |
| Stereo 16Ω HP Playback (48kHz, white noise) ² | 5.4 | 7.3 | 0.004 | 29.4 |
| Stereo 16Ω HP Playback (48kHz, sine wave) ² | 18 | 6.7 | 0.004 | 66.1 |

Notes:

1. DBVDD Current will increase with greater loading on digital I/O pins.
2. 5 Band EQ is enabled.
3. AVDD standby current will fall to nearer 15uA when thermal shutdown sensor is disabled.

Table 1 Power Consumption

ESTIMATING SUPPLY CURRENT

When either the DAC or ADC is enabled approximately 7mA will be drawn from DCVDD when DCVDD=1.8V and fs=48kHz. When the PLL is enabled approximately 1.5mA additional current will be drawn from DCVDD.

As a general rule, digital supply currents will scale in proportion to sample rates. Supply current for analogue and digital blocks will also be lower at lower supply voltages.

Power consumed by the output drivers will depend greatly on the signal characteristics. A quiet signal, or a signal with long periods of silence will consume less power than a signal which is continuously loud.

Estimated supply current for the analogue blocks is shown in Table 2. Note that power dissipated in the load is not shown.

| REGISTER BIT | AVDD CURRENT (mA) AVDD=3.3V |
|--------------|---|
| BUFDCOPEN | 0.1 |
| OUT4MIXEN | 0.2 |
| OUT3MIXEN | 0.2 |
| PLLEN | 1.2 (with clocks applied) |
| MICBEN | 0.5 |
| BIASEN | 0.3 |
| BUFIOEN | 0.1 |
| VMIDSEL | 5K Ω = >0.3, less than 0.1 for 75K Ω 300K Ω settings |
| ROUT1EN | 0.4 |
| LOUT1EN | 0.4 |
| BOOSTENL | 0.2 |
| INPPGAENL | 0.2 |
| ADCENL | 2.6 (x64, ADCOSR=0) 4.9 (x128, ADCOSR=1) |
| OUT4EN | 0.2 |
| OUT3EN | 0.2 |
| LOUT2EN | 1mA from SPKVDD + 0.2mA from AVDD in 5V mode |
| ROUT2EN | 1mA from SPKVDD + 0.2mA from AVDD in 5V mode |
| RMIXEN | 0.2 |
| LMIXEN | 0.2 |
| DACENR | 1.8 (x64, DACOSR=0) 1.9 (x128, DACOSR=1) |
| DACENL | 1.8 (x64, DACOSR=0) 1.9 (x128, DACOSR=1) |

Table 2 AVDD Supply Current (AVDD=3.3V)

AUDIO PATHS OVERVIEW

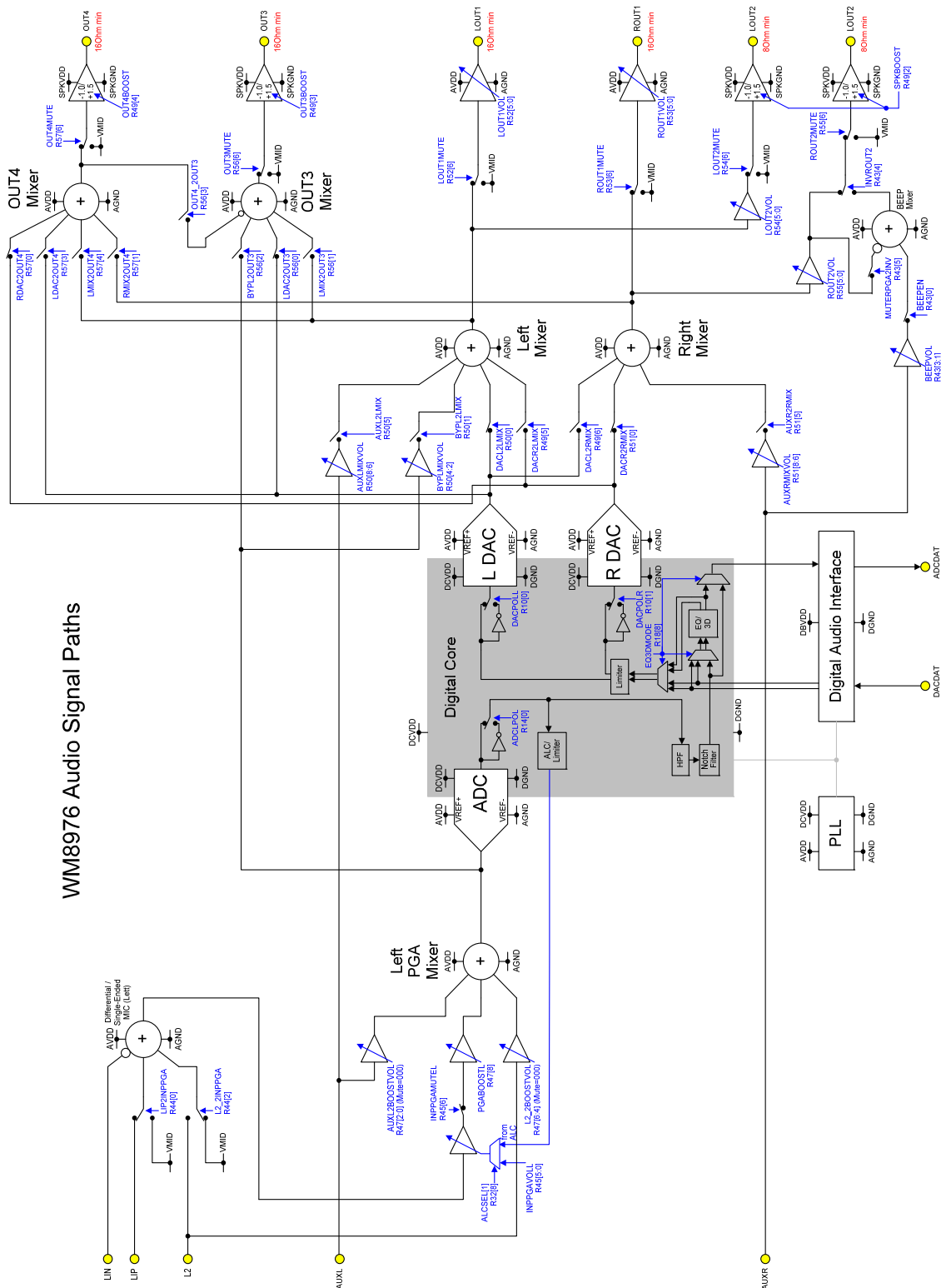


Figure 1 WM8976 Audio Signal Paths

SIGNAL TIMING REQUIREMENTS
SYSTEM CLOCK TIMING

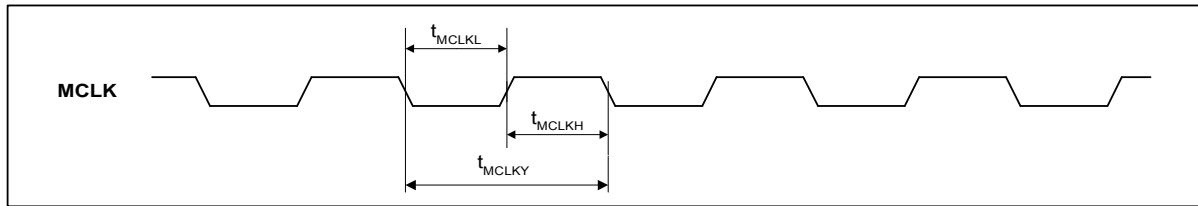


Figure 2 System Clock Timing Requirements

Test Conditions

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

| PARAMETER | SYMBOL | CONDITIONS | MIN | TYP | MAX | UNIT |
|--|---------------------|-------------------------------------|-------|-----|-------|------|
| System Clock Timing Information | | | | | | |
| MCLK cycle time | T _{MCLKY} | MCLK=SYSCLK (=256fs) | 81.38 | | | ns |
| | | MCLK input to PLL ^{Note 1} | 20 | | | ns |
| MCLK duty cycle | T _{MCLKDS} | | 60:40 | | 40:60 | |

Note 1:

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

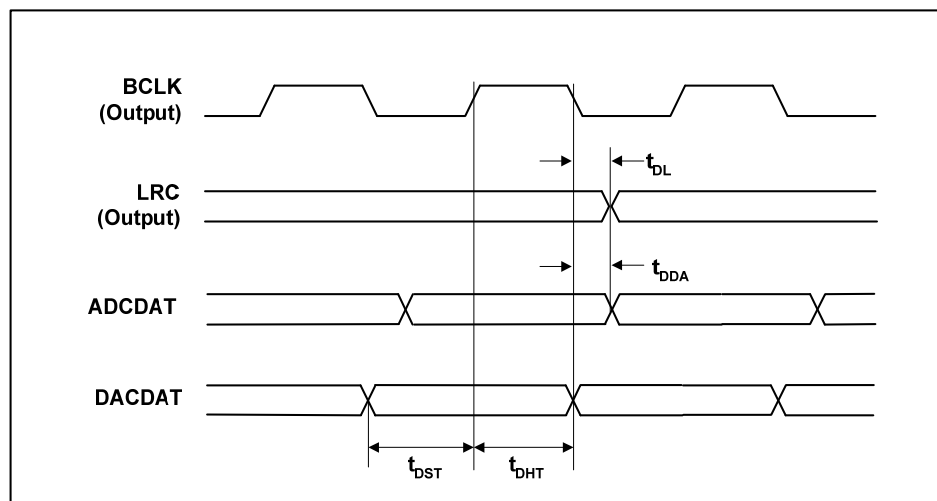


Figure 3 Digital Audio Data Timing – Master Mode (see Control Interface)

Test Conditions

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

| PARAMETER | SYMBOL | MIN | TYP | MAX | UNIT |
|---|------------------|-----|-----|-----|------|
| Audio Data Input Timing Information | | | | | |
| LRC propagation delay from BCLK falling edge | t _{DL} | | | 10 | ns |
| ADCDAT propagation delay from BCLK falling edge | t _{DDA} | | | 10 | ns |
| DACDAT setup time to BCLK rising edge | t _{DST} | 10 | | | ns |
| DACDAT hold time from BCLK rising edge | t _{DHT} | 10 | | | ns |

AUDIO INTERFACE TIMING – SLAVE MODE

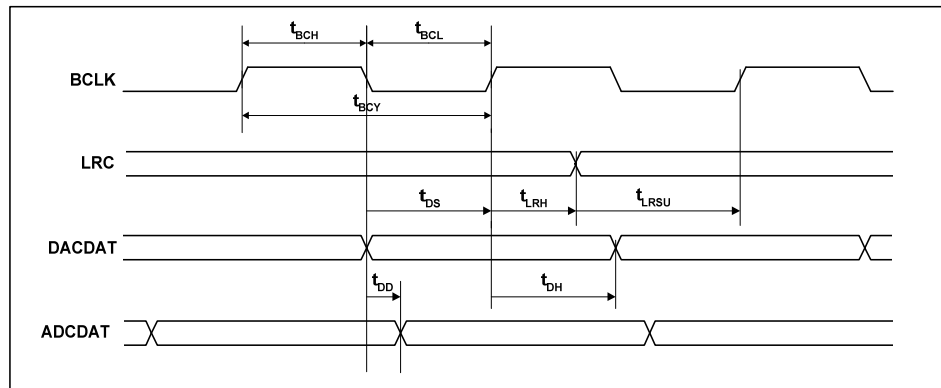


Figure 4 Digital Audio Data Timing – Slave Mode

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.

| PARAMETER | SYMBOL | MIN | TYP | MAX | UNIT |
|---|-------------------|-----|-----|-----|------|
| Audio Data Input Timing Information | | | | | |
| BCLK cycle time | t _{BCY} | 50 | | | ns |
| BCLK pulse width high | t _{BCH} | 20 | | | ns |
| BCLK pulse width low | t _{BCL} | 20 | | | ns |
| LRC set-up time to BCLK rising edge | t _{LRSU} | 10 | | | ns |
| LRC hold time from BCLK rising edge | t _{LRH} | 10 | | | ns |
| DACDAT hold time from BCLK rising edge | t _{DH} | 10 | | | ns |
| DACDAT setup time to BCLK rising edge | t _{DS} | 10 | | | ns |
| ADCDAT propagation delay from BCLK falling edge | t _{DD} | | | 10 | ns |

Note:

BCLK period should always be greater than or equal to MCLK period.

CONTROL INTERFACE TIMING – 3-WIRE MODE

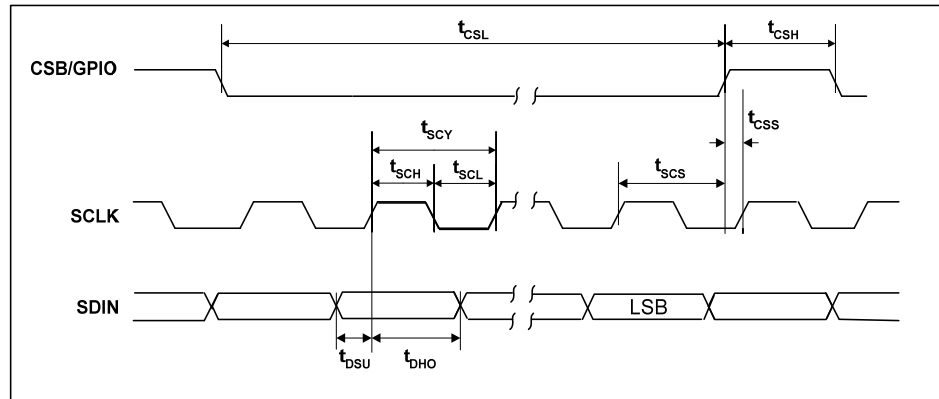
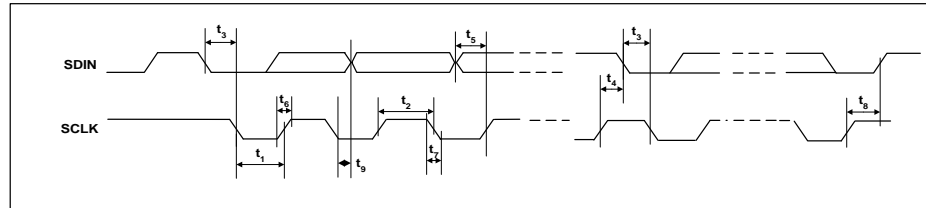


Figure 5 Control Interface Timing – 3-Wire Serial Control Mode

Test Conditions

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

| PARAMETER | SYMBOL | MIN | TYP | MAX | UNIT |
|---|-----------|-----|-----|-----|------|
| Program Register Input Information | | | | | |
| SCLK rising edge to CSB rising edge | t_{SCS} | 80 | | | ns |
| SCLK pulse cycle time | t_{SCY} | 200 | | | ns |
| SCLK pulse width low | t_{SCL} | 80 | | | ns |
| SCLK pulse width high | t_{SCH} | 80 | | | ns |
| SDIN to SCLK set-up time | t_{DSU} | 40 | | | ns |
| SCLK to SDIN hold time | t_{DHO} | 40 | | | ns |
| CSB pulse width low | t_{CSL} | 40 | | | ns |
| CSB pulse width high | t_{CSH} | 40 | | | ns |
| CSB rising to SCLK rising | t_{CSS} | 40 | | | ns |
| Pulse width of spikes that will be suppressed | t_{ps} | 0 | | 5 | ns |

CONTROL INTERFACE TIMING – 2-WIRE MODE**Figure 6 Control Interface Timing – 2-Wire Serial Control Mode****Test Conditions**

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

| PARAMETER | SYMBOL | MIN | TYP | MAX | UNIT |
|---|----------|-----|-----|-----|------|
| Program Register Input Information | | | | | |
| SCLK Frequency | | 0 | | 526 | kHz |
| SCLK Low Pulse-Width | t_1 | 1.3 | | | us |
| SCLK High Pulse-Width | t_2 | 600 | | | ns |
| Hold Time (Start Condition) | t_3 | 600 | | | ns |
| Setup Time (Start Condition) | t_4 | 600 | | | ns |
| Data Setup Time | t_5 | 100 | | | ns |
| SDIN, SCLK Rise Time | t_6 | | | 300 | ns |
| SDIN, SCLK Fall Time | t_7 | | | 300 | ns |
| Setup Time (Stop Condition) | t_8 | 600 | | | ns |
| Data Hold Time | t_9 | | | 900 | ns |
| Pulse width of spikes that will be suppressed | t_{ps} | 0 | | 5 | ns |

INTERNAL POWER ON RESET CIRCUIT

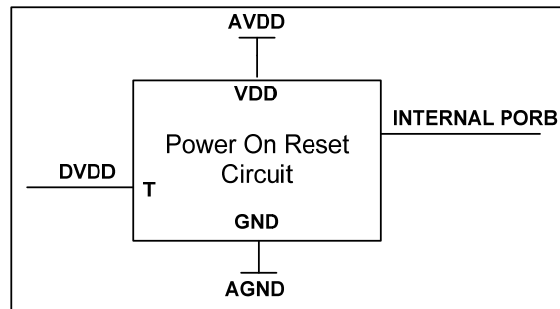


Figure 7 Internal Power on Reset Circuit Schematic

The WM8980 includes an internal Power-On-Reset Circuit, as shown in Figure 7, which is used reset the digital logic into a default state after power up. The POR circuit is powered from AVDD and monitors DVDD. It asserts PORB low if AVDD or DVDD is below a minimum threshold.

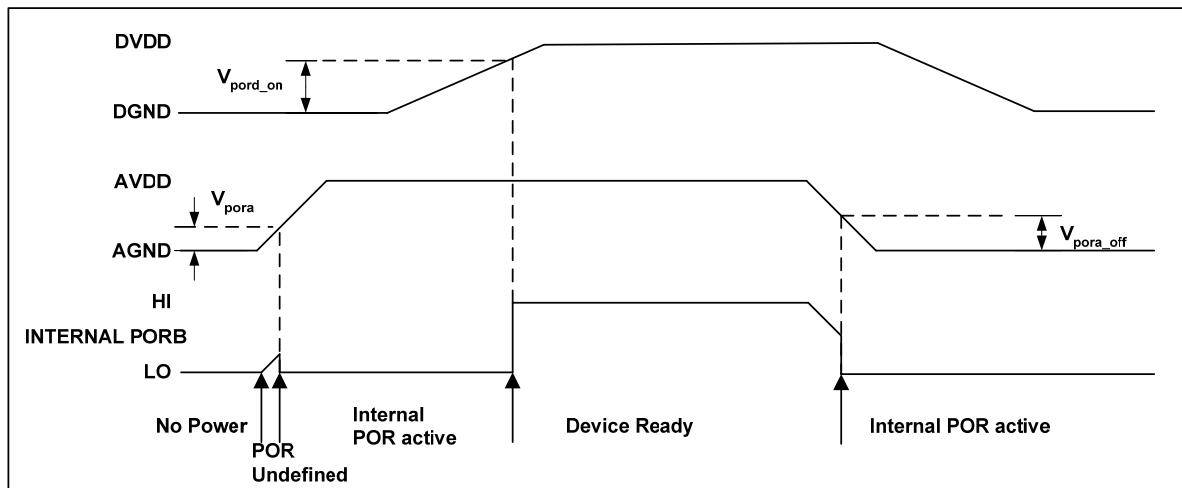


Figure 8 Typical Power up Sequence where AVDD is Powered before DVDD

Figure 8 shows a typical power-up sequence where AVDD comes up first. When AVDD goes above the minimum threshold, V_{pora} , there is enough voltage for the circuit to guarantee PORB is asserted low and the chip is held in reset. In this condition, all writes to the control interface are ignored. Now AVDD is at full supply level. Next DVDD rises to V_{pord_on} and PORB is released high and all registers are in their default state and writes to the control interface may take place.

On power down, where AVDD falls first, PORB is asserted low whenever AVDD drops below the minimum threshold V_{pora_off} .

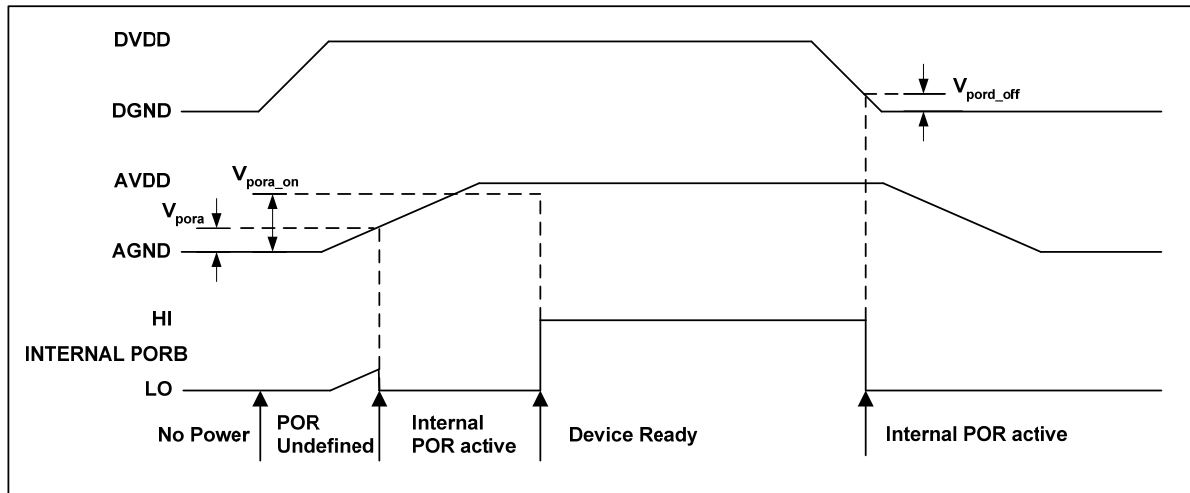


Figure 9 Typical Power up Sequence where DVDD is Powered before AVDD

Figure 9 shows a typical power-up sequence where DVDD comes up first. First it is assumed that DVDD is already up to specified operating voltage. When AVDD goes above the minimum threshold, V_{pora} , there is enough voltage for the circuit to guarantee PORB is asserted low and the chip is held in reset. In this condition, all writes to the control interface are ignored. When AVDD rises to V_{pora_on} , PORB is released high and all registers are in their default state and writes to the control interface may take place.

On power down, where DVDD falls first, PORB is asserted low whenever DVDD drops below the minimum threshold V_{pord_off} .

| SYMBOL | MIN | TYP | MAX | UNIT |
|-----------------|-----|-----|-----|------|
| V_{pora} | 0.4 | 0.6 | 0.8 | V |
| V_{pora_on} | 0.9 | 1.2 | 1.6 | V |
| V_{pora_off} | 0.4 | 0.6 | 0.8 | V |
| V_{pord_on} | 0.5 | 0.7 | 0.9 | V |
| V_{pord_off} | 0.4 | 0.6 | 0.8 | V |

Table 3 Typical POR Operation (typical values, not tested)

Notes:

1. If AVDD and DVDD suffer a brown-out (i.e. drop below the minimum recommended operating level but do not go below V_{pora_off} or V_{pord_off}) then the chip will not reset and will resume normal operation when the voltage is back to the recommended level again.
2. The chip will enter reset at power down when AVDD or DVDD falls below V_{pora_off} or V_{pord_off} . This may be important if the supply is turned on and off frequently by a power management system.
3. The minimum t_{por} period is maintained even if DVDD and AVDD have zero rise time. This specification is guaranteed by design rather than test.

DEVICE DESCRIPTION

INTRODUCTION

The WM8976 is a low power audio CODEC combining a high quality stereo audio DAC and mono ADC, with flexible line and microphone input and output processing. Applications for this device include multimedia phones, digital camcorders, and digital still cameras with record and playback capability.

FEATURES

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

MICROPHONE INPUT

A microphone input is provided, allowing a microphone to be pseudo-differentially connected, with user defined gain using internal resistors. The provision of the common mode input pin allows for rejection of common mode noise on the microphone input (level depends on gain setting chosen). A microphone bias is output from the chip which can be used to bias the microphone. The signal routing can be configured to allow manual adjustment of mic level, or to allow the ALC loop to control the level of mic signal that is transmitted.

Total gain through the microphone path 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.

LINE INPUTS (AUXL, AUXR)

The inputs, AUXL and AUXR, can be used as a stereo line input or as an input for warning tones (or 'beeps') etc. The left input can be summed into the record path, along with the microphone preamp output, so allowing for mixing of audio with 'backing music' etc as required.

ADC

The ADC uses a 24-bit delta sigma oversampling architecture to deliver optimum performance with low power consumption.

HI-FI DAC

The hi-fi DAC provides high quality audio playback suitable for all portable audio hi-fi type applications, including MP3 players and portable disc players of all types.

OUTPUT MIXERS

Flexible mixing is provided on the outputs of the device. A stereo mixer is provided for the stereo headphone or line outputs, LOUT1/ROUT1, and additional summers on the OUT3/OUT4 outputs allow for an optional differential or stereo line output on these pins. Gain adjustment PGAs are provided for the LOUT1/ROUT1 and LOUT2/ROUT2 outputs, and signal switching is provided to allow for all possible signal combinations. The output buffers can be configured in several ways, allowing support of up to three sets of external transducers; ie stereo headphone, BTL speaker, and BTL earpiece may be connected simultaneously. Thermal implications should be considered before simultaneous full power operation of all outputs is attempted.

Alternatively, if a speaker output is not required, the LOUT2 and ROUT2 pins might be used as a stereo headphone driver, (disable output invert buffer on ROUT2). In that case two sets of headphones might be driven, or the LOUT2 and ROUT2 pins used as a line output driver.

OUT3 and OUT4 can be configured to provide an additional stereo lineout from the output of the DACs, the mixers or the input microphone boost stages. Alternatively OUT4 can be configured as a mono mix of left and right DACs or mixers, or simply a buffered version of the chip midrail reference voltage. OUT3 can also be configured as a buffered VMID output. This voltage may then be used as a headphone 'pseudo ground' allowing removal of the large AC coupling capacitors often used in the output path.

AUDIO INTERFACES

The WM8976 has a standard audio interface, to support the transmission of data to and from the chip. This interface is a 3 wire standard audio interface which supports a number of audio data formats including I2S, DSP/PCM Mode (a burst mode in which LRC sync plus 2 data packed words are transmitted), 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 features, the WM8976 offers a choice of 2 or 3 wire control interface. It is fully compatible and an ideal partner for a wide range of industry standard microprocessors, controllers and DSPs.

Selection between the modes is via the MODE pin. In 2 wire mode the address of the device is fixed as 0011010.

CLOCKING SCHEMES

WM8976 offers the normal audio DAC clocking scheme operation, where 256fs MCLK is provided to the DAC and ADC.

A PLL is included which may be used to generate these clocks in the event that they are not available from the system controller. This PLL uses an input clock, typically the 12MHz USB or iLink clock, to generate high quality audio clocks. If this PLL is not required for generation of these clocks, it can be reconfigured to generate alternative clocks which may then be output on the GPIO pins and used elsewhere in the system.

POWER CONTROL

The design of the WM8976 has given much attention to power consumption without compromising performance. It operates at very low voltages, and includes the ability to power off any unused parts of the circuitry under software control, and includes standby and power off modes.

OPERATION SCENARIOS

Flexibility in the design of the WM8976 allows for a wide range of operational scenarios, some of which are proposed below:

Multimedia phone; High quality playback to a stereo headset, a mono ear speaker or a loudspeaker is supported, allowing hi-fi playback to be mixed with voice and other analogue inputs while simultaneously transmitting a differential output from the microphone amplifier. A 5-band EQ enables hi-fi playback to be customised to suit the user's preferences and the music style, while programmable filtering allows fixed-frequency noise (e.g. 217Hz) to be reduced in the digital domain.

Camcorder; the provision of a microphone preamplifier allows support for both internal and external microphones. All drivers for speaker, headphone and line output connections are integrated. The

selectable 'application filters' after the ADC provide for features such as 'wind noise' reduction, or mechanical noise reducing filters.

Digital still camera recording; Support for digital recording is similar to the camcorder case. However, additionally if the DSC supports MP3 playback, and perhaps recording, the ability of the ADC to support full 48ks/s high quality recording increases device flexibility.

AUXILIARY ANALOGUE INPUTS

An analogue stereo FM tuner or other auxiliary analogue input can be connected to the AUX inputs of WM8976, and the stereo signal listened to via headphones.

INPUT SIGNAL PATH

The WM8976 has flexible analogue inputs. An input PGA stage is followed by a boost/mix stage which drives into the hi-fi ADC. The input path has three input pins which can be configured in a variety of ways to accommodate single-ended or differential microphones. There is an auxiliary input pin which can be fed into the input boost/mix stage as well as driving into the output path. A bypass path exists from the output of the boost/mix stage into the output left/right mixers.

MICROPHONE INPUTS

The WM8976 can accommodate a variety of microphone configurations including single ended and differential inputs. The inputs to the differential input PGA are LIN, LIP and L2.

In single-ended microphone input configuration the microphone signal should be input to LIN and the internal NOR gate configured to clamp the non-inverting input of the input PGA to VMID.

In differential mode the larger signal should be input to LIP and the smaller (e.g. noisy ground connections) should be input to LIN.

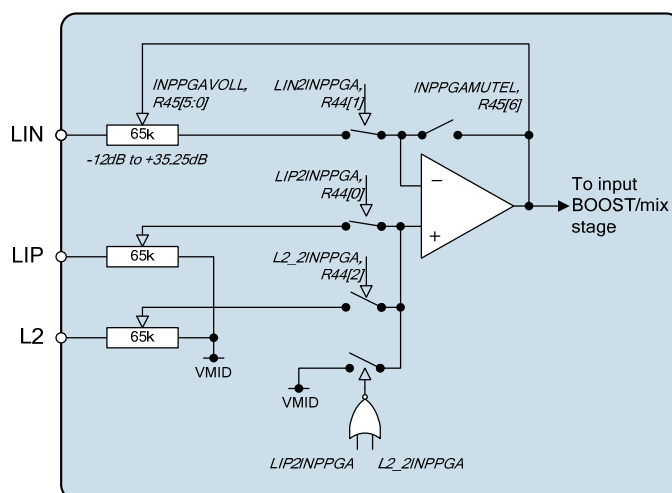


Figure 10 Microphone Input PGA Circuit

The input PGA is enabled by the IPPGAENL register bits.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|--------------------------------|-----|-----------|---------|---|
| R2 Power Management 2 | 2 | INPPGAENL | 0 | Input PGA enable 0 = disabled 1 = enabled |

Table 4 Input PGA Enable Register Settings

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|-------------------------|-----|------------|---------|--|
| R44 Input Control | 0 | LIP2INPPGA | 1 | Connect LIP pin to input PGA amplifier positive terminal. 0 = LIP not connected to input PGA 1 = input PGA amplifier positive terminal connected to LIP (constant input impedance) |
| | 1 | LIN2INPPGA | 1 | Connect LIN pin to input PGA negative terminal. 0=LIN not connected to input PGA 1=LIN connected to input PGA amplifier negative terminal. |
| | 2 | L2_2INPPGA | 0 | Connect L2 pin to input PGA positive terminal. 0=L2 not connected to input PGA 1=L2 connected to input PGA amplifier positive terminal (constant input impedance). |

Table 5 Input PGA Control

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 LIN input to the PGA output and from the L2 amplifier to the PGA output is always common and controlled by the register bits INPPGAVOLL[5:0]. These register bits also affect the LIP pin when LIP2INPPGA=1, the L2 pin when L2_2INPPGA=1 and the L2 pin when L2_2INPPGA=1.

When the Automatic Level Control (ALC) is enabled the input PGA gains are controlled automatically and the INPPGAVOLL bits should not be used.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|---------------------------------------|-----|--------------|-------------|--|
| R45 Input PGA volume control | 5:0 | INPPGAVOLL | 010000 | Input PGA volume 000000 = -12dB 000001 = -11.25db . 010000 = 0dB . 111111 = 35.25dB |
| | 6 | INPPGAMUTEL | 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). |
| | 7 | INPPGAZCL | 0 | Input PGA zero cross enable: 0=Update gain when gain register changes 1=Update gain on 1 st zero cross after gain register write. |
| | 8 | INPPGAUPDATE | Not latched | INPPGAVOLL volume does not update until a 1 is written to INPPGAUPDATE |
| R32 ALC control 1 | 8 | ALCSEL | 0 | ALC function select: 0=ALC off 1=ALC on |

Table 6 Input PGA Volume Control

VOLUME UPDATES

Volume settings will not be applied to the PGAs until a '1' is written to one of the INPPGAUPDATE bits. This is to allow left and right channels to be updated at the same time, as shown in Figure 11.

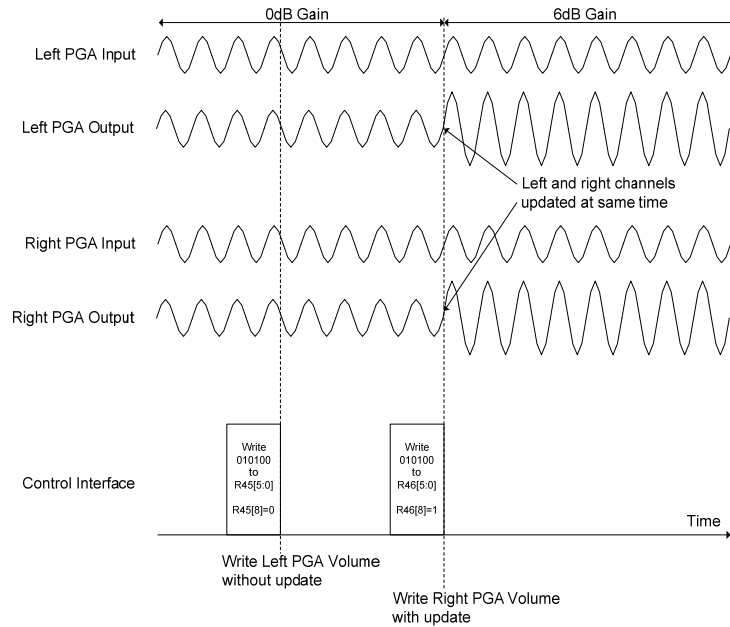


Figure 11 Simultaneous Left and Right Volume Updates

If the volume is adjusted while the signal is a non-zero value, an audible click can occur as shown in Figure 12.

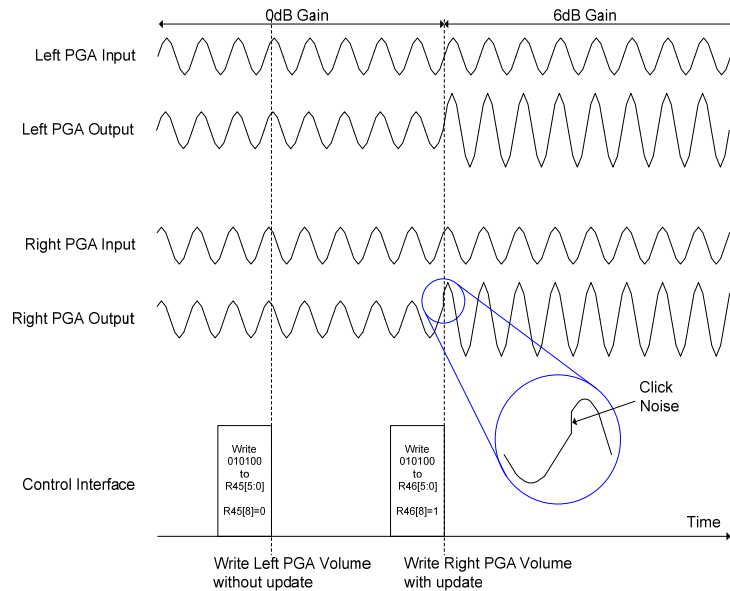


Figure 12 Click Noise During Volume Update

In order to prevent this click noise, a zero cross function is provided. When enabled, this will cause the PGA volume to update only when a zero crossing occurs, minimising click noise as shown in Figure 13.

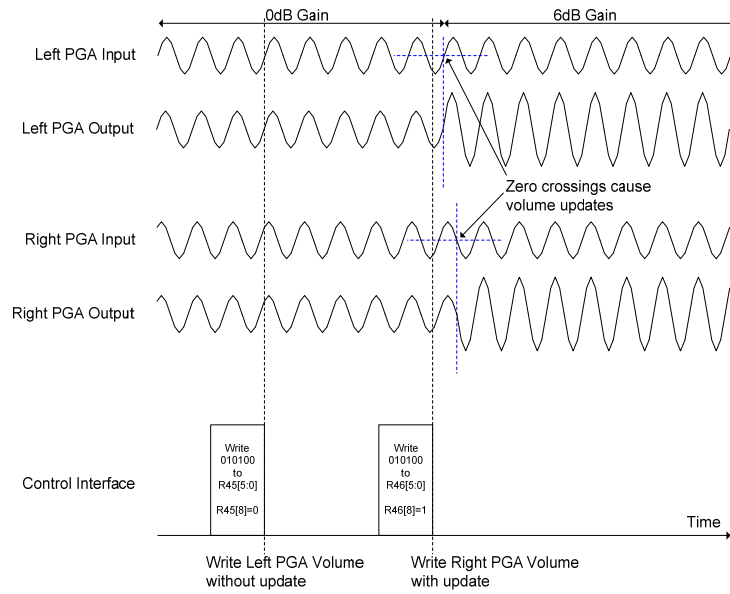


Figure 13 Volume Update Using Zero Cross Detection

If there is a long period where no zero-crossing occurs, a timeout circuit in the WM8980 will automatically update the volume. The volume updates will occur between one and two timeout periods, depending on when the INPPGAUPDATE bit is set as shown in Figure 14.

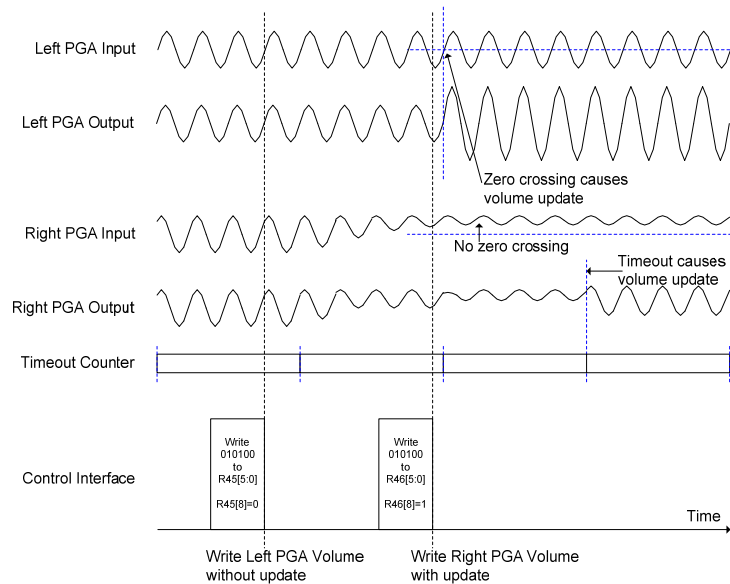


Figure 14 Volume Update after Timeout

AUXILIARY INPUTS

There are two auxiliary inputs, AUXL and AUXR which can be used for a variety of purposes such as stereo line inputs or as a 'beep' input signal to be mixed with the outputs.

The AUXL input can be used as a line input to the input BOOST stage which has gain adjust of -12dB to +6dB in 3dB steps (plus off). See the INPUT BOOST section for further details.

The AUXL/R inputs can also be mixed into the output channel mixers, with a gain of -15dB to +6dB plus off.

In addition the AUXR input can be summed into the Right speaker output path (ROUT2) with a gain adjust of -15 to +6dB. This allows a 'beep' input to be output on the speaker outputs only without affecting the headphone or lineout signals.

INPUT BOOST

The input PGA stage is followed by an input BOOST circuit. The input BOOST circuit has 3 selectable inputs: the input microphone PGA output, the AUX amplifier output and the L2 input pin (can be used as a line input, bypassing the input PGA). These three inputs can be mixed together and have individual gain boost/adjust as shown in Figure 15.

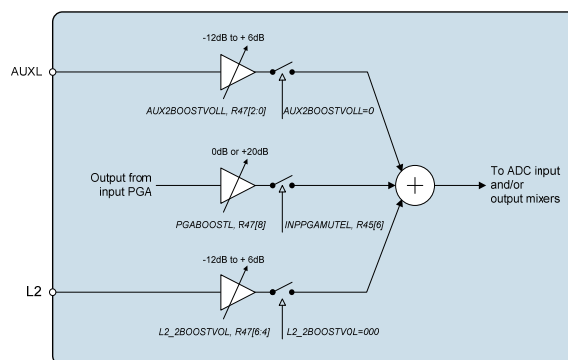


Figure 15 Input Boost Stage

The input PGA paths can have a +20dB boost (PGABOOSTL=1), a 0dB pass through (PGABOOSTL=0) or be completely isolated from the input boost circuit (INPPGAMUTEL=1).

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|----------------------------|-----|-----------|---------|--|
| R47 Input BOOST control | 8 | PGABOOSTL | 1 | Boost enable for input PGA: 0 = PGA output has +0dB gain through input BOOST stage. 1 = PGA output has +20dB gain through input BOOST stage. |

Table 7 Input BOOST Stage Control

The Auxiliary amplifier path to the BOOST stage is controlled by the AUXL2BOOSTVOL[2:0] register bits. When AUXL2BOOSTVOL=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 L2 path to the BOOST stage is controlled by the LIP2BOOSTVOL[2:0] register bits. When L2_2BOOSTVOL=000 the L2 input pin is completely disconnected from the BOOST stage. Settings 001 through to 111 control the gain in 3dB steps from -12dB to +6dB.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|----------------------------|-----|---------------|---------|---|
| R47 Input BOOST control | 2:0 | AUXL2BOOSTVOL | 000 | 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 |
| | 6:4 | L2_2BOOSTVOL | 000 | Controls the L2 pin 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 |

Table 8 Input BOOST Stage Control

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

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|--------------------------|-----|----------|---------|---|
| R2 Power management 2 | 4 | BOOSTENL | 0 | Input BOOST enable 0 = Boost stage OFF 1 = Boost stage ON |

Table 9 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. When MBVSEL=0, MICBIAS=0.9*AVDD and when MBVSEL=1, MICBIAS=0.65*AVDD. The output can be enabled or disabled using the MICBEN control bit.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|--------------------------|-----|--------|---------|---|
| R1 Power management 1 | 4 | MICBEN | 0 | Microphone Bias Enable 0 = OFF (high impedance output) 1 = ON |

Table 10 Microphone Bias Enable Control

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|----------------------|-----|--------|---------|--|
| R44 Input control | 8 | MBVSEL | 0 | Microphone Bias Voltage Control 0 = 0.9 * AVDD 1 = 0.65 * AVDD |

Table 11 Microphone Bias Voltage Control

The internal MICBIAS circuitry is shown in Figure 16. 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.

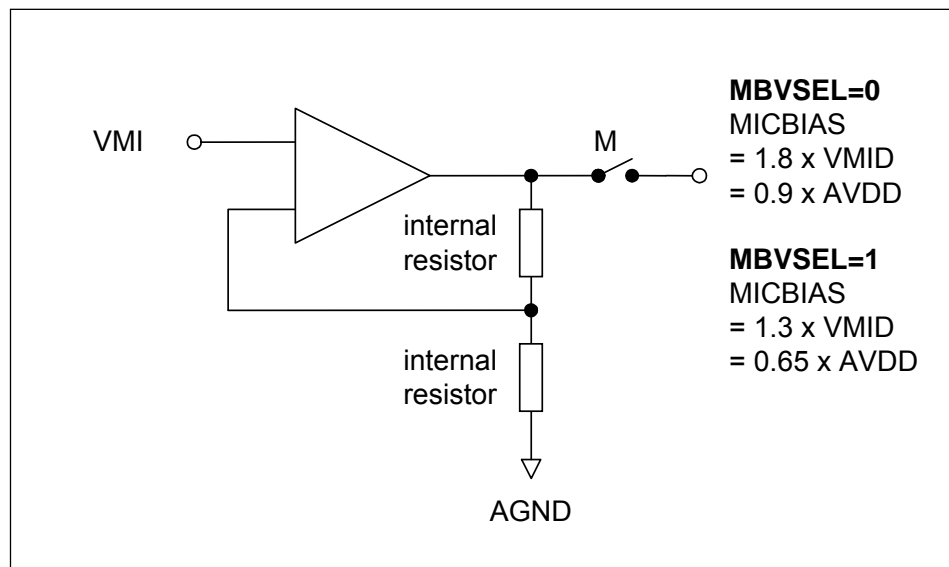


Figure 16 Microphone Bias Schematic

ANALOGUE TO DIGITAL CONVERTER (ADC)

The WM8976 uses a multi-bit, oversampled sigma-delta ADC. 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 1.0V_{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 for each ADC channel is illustrated in Figure 17.

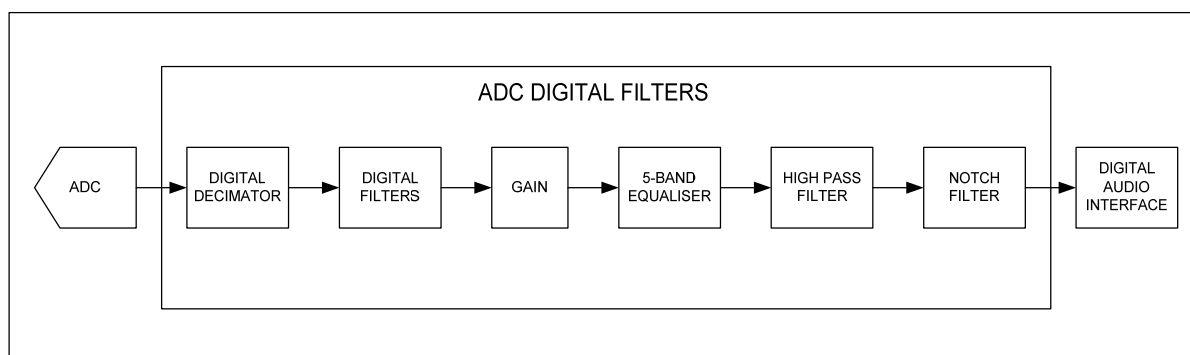


Figure 17 ADC Digital Filter Path

The ADC is enabled by the ADCENL/R register bit.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|--------------------------|-----|--------|---------|--|
| R2 Power management 2 | 0 | ADCENL | 0 | Enable ADC: 0 = ADC disabled 1 = ADC enabled |

Table 12 ADC Enable Control

The polarity of the output signal can also be changed under software control using the ADCLPOL 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.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|--------------------|-----|---------|---------|---|
| R14 ADC Control | 0 | ADCLPOL | 0 | ADC polarity adjust: 0=normal 1=inverted |
| | 3 | ADCOSR | 0 | ADC oversample rate select: 0=64x (lower power) 1=128x (best performance) |

Table 13 ADC Control

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 15.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|--------------------|-----|--------|---------|--|
| R14 ADC Control | 8 | HPFEN | 1 | High Pass Filter Enable 0=disabled 1=enabled |
| | 7 | HPFAPP | 0 | Select audio mode or application mode 0=Audio mode (1 st order, $f_c \approx 3.7\text{Hz}$) 1=Application mode (2 nd order, $f_c = \text{HPFCUT}$) |
| | 6:4 | HPFCUT | 000 | Application mode cut-off frequency See Table 15 for details. |

Table 14 ADC Enable Control

| HPFCUT [2:0] | SR=101/100 | | | SR=011/010 | | | SR=001/000 | | |
|-----------------|------------|--------|-----|------------|-------|-----|------------|------|-----|
| | fs (kHz) | | | | | | | | |
| | 8 | 11.025 | 12 | 16 | 22.05 | 24 | 32 | 44.1 | 48 |
| 000 | 82 | 113 | 122 | 82 | 113 | 122 | 82 | 113 | 122 |
| 001 | 102 | 141 | 153 | 102 | 141 | 153 | 102 | 141 | 153 |
| 010 | 131 | 180 | 196 | 131 | 180 | 196 | 131 | 180 | 196 |
| 011 | 163 | 225 | 245 | 163 | 225 | 245 | 163 | 225 | 245 |
| 100 | 204 | 281 | 306 | 204 | 281 | 306 | 204 | 281 | 306 |
| 101 | 261 | 360 | 392 | 261 | 360 | 392 | 261 | 360 | 392 |
| 110 | 327 | 450 | 490 | 327 | 450 | 490 | 327 | 450 | 490 |
| 111 | 408 | 563 | 612 | 408 | 563 | 612 | 408 | 563 | 612 |

Table 15 High Pass Filter Cut-off Frequencies (HPFAPP=1). Values in Hz.

Note that the High Pass filter values (when HPFAPP=1) are calculated with the assumption that the SR register bits are set correctly for the actual sample rate as shown in Table 15.

PROGRAMMABLE NOTCH FILTER

A programmable notch filter is provided. This filter has a variable centre frequency and bandwidth, programmable via two coefficients, a_0 and a_1 . The coefficients must be entered in 2's complement notation. A_0 and a_1 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.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|-----------------------|-----|------------|---------|--|
| R27 Notch Filter 1 | 6:0 | NFA0[13:7] | 0 | Notch filter a_0 coefficient, bits [13:7] |
| | 7 | NFEN | 0 | Notch filter enable: 0=Disabled 1=Enabled |
| | 8 | NFU | 0 | Notch filter update. The notch filter values used internally only update when one of the NFU bits is set high. |
| R28 Notch Filter 2 | 6:0 | NFA0[6:0] | 0 | Notch filter a_0 coefficient, bits [6:0] |
| | 8 | NFU | 0 | Notch filter update. The notch filter values used internally only update when one of the NFU bits is set high. |
| R29 Notch Filter 3 | 6:0 | NFA1[13:7] | 0 | Notch filter a_1 coefficient, bits [13:7] |
| | 8 | NFU | 0 | Notch filter update. The notch filter values used internally only update when one of the NFU bits is set high. |
| R30 Notch Filter 4 | 0-6 | NFA1[6:0] | 0 | Notch filter a_1 coefficient, bits [6:0] |
| | 8 | NFU | 0 | Notch filter update. The notch filter values used internally only update when one of the NFU bits is set high. |

Table 16 Notch Filter Function

The coefficients are calculated as follows:

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

$$a_1 = -(1 + a_0) \cos(w_0)$$

Where:

$$w_0 = 2\pi f_c / f_s$$

$$w_b = 2\pi f_b / f_s$$

f_c = centre frequency in Hz, f_b = -3dB bandwidth in Hz, f_s = sample frequency in Hz

The actual register values can be determined from the coefficients as follows:

$$\text{NFA0} = -a_0 \times 2^{13}$$

$$\text{NFA1} = -a_1 \times 2^{12}$$

NOTCH FILTER WORKED EXAMPLE

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

$$f_c = 1000 \text{ Hz}$$

$$f_b = 100 \text{ Hz}$$

$$f_s = 48000 \text{ Hz}$$

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

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

$$a_0 = \frac{1 - \tan(w_b / 2)}{1 + \tan(w_b / 2)} = \frac{1 - \tan(0.01308996939 / 2)}{1 + \tan(0.01308996939 / 2)} = 0.9869949627$$

$$a_1 = -(1 + a_0) \cos(w_0) = -(1 + 0.9869949627) \cos(0.1308996939) = -1.969995945$$

$$\text{NFA0} = -a_0 \times 213 = -8085 \text{ (rounded to nearest whole number)}$$

$$\text{NFA1} = -a_1 \times 212 = 8069 \text{ (rounded to nearest whole number)}$$

These values are then converted to a 2's complement notation:

$$\text{NfnA0}[12:0] = 13'h1F95; \text{ Converting to 2's complement NFA0} = 14'h4000 - 14'h1F95 = 14'h206B$$

$$\text{NfnA1}[12:0] = 13'h1F85; \text{ Converting to 2's complement NFA0} = 14'h1F85$$

DIGITAL ADC VOLUME CONTROL

The output of the ADC 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:

$$0.5 \times (G-255) \text{ dB for } 1 \leq G \leq 255; \quad \text{MUTE for } G = 0$$

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|---------------------------|-----|------------------|---------------------|--|
| R15 ADC Digital Volume | 7:0 | ADCVOLL [7:0] | 11111111 (0dB) | ADC Digital Volume Control 0000 0000 = Digital Mute 0000 0001 = -127dB 0000 0010 = -126.5dB ... 0.5dB steps up to 1111 1111 = 0dB |
| | 8 | ADCVU | Not latched | ADC volume does not update until a 1 is written to ADCVU |

Table 17 ADC Digital Volume Control

INPUT LIMITER / AUTOMATIC LEVEL CONTROL (ALC)

The WM8976 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.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|-------------------------------|-----|-----------------|-------------------|--|
| R32 (20h) ALC Control 1 | 2:0 | ALCMIN [2:0] | 000 (-12dB) | Set minimum gain of PGA 000 = -12dB 001 = -6dB 010 = 0dB 011 = +6dB 100 = +12dB 101 = +18dB 110 = +24dB 111 = +30dB |
| | 5:3 | ALCMAX [2:0] | 111 (+35.25dB) | Set Maximum Gain of PGA 111 = +35.25dB 110 = +29.25dB 101 = +23.25dB 100 = +17.25dB 011 = +11.25dB 010 = +5.25dB 001 = -0.75dB 000 = -6.75dB |
| | 8:7 | ALCSEL | 00 | ALC function select 00 = ALC disabled 01 = Right channel ALC enabled 10 = Left channel ALC enabled 11 = Both channels ALC enabled |
| R33 (21h) ALC Control 2 | 3:0 | ALCLVL [3:0] | 1011 (-6dB) | ALC target – sets signal level at ADC input 1111 = -1.5dBFS 1110 = -1.5dBFS 1101 = -3dBFS 1100 = -4.5dBFS 1011 = -6dBFS 1010 = -7.5dBFS 1001 = -9dBFS 1000 = -10.5dBFS 0111 = -12dBFS 0110 = -13.5dBFS 0101 = -15dBFS 0100 = -16.5dBFS 0011 = -18dBFS 0010 = -19.5dBFS 0001 = -21dBFS 0000 = -22.5dBFS |
| | 8 | Reserved | 0 | Reserved. Set to 0. |

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION | | | |
|------------------------------------|-----------------|---------------------|--|---|--|--------------|--------------|
| | 7:4 | ALCHLD [3:0] | 0000 (0ms) | ALC hold time before gain is increased. 0000 = 0ms 0001 = 2.67ms 0010 = 5.33ms 0011 = 10.66ms 0100 = 21.32ms 0101 = 42.64ms 0110 = 85.28ms 0111 = 0.17s 1000 = 0.34s 1001 = 0.68s 1010 or higher = 1.36s | | | |
| R34 (22h) ALC Control 3 | 8 | ALCMODE | 0 | Determines the ALC mode of operation: 0 = ALC mode (Normal Operation) 1 = Limiter mode. | | | |
| | 7:4 | ALCDCY [3:0] | 0011 (26ms/6dB) | Decay (gain ramp-up) time (ALCMODE ==0) | | | |
| | | | | | Per step | Per 6dB | 90% of range |
| | | | | 0000 | 410us | 3.28ms | 23.6ms |
| | | | | 0001 | 820us | 6.56ms | 47.2ms |
| | | | | 0010 | 1.64ms | 13.1ms | 94.5ms |
| | | | | ... (time doubles with every step) | | | |
| | | | | 1010 or higher | 420ms | 3.36s | 24.2s |
| | | | | 0011 (5.8ms/6dB) | Decay (gain ramp-up) time (ALCMODE ==1) | | |
| | | | | | Per step | Per 6dB | 90% of range |
| | | | | 0000 | 90.8us | 726us | 5.23ms |
| | 0001 | 182us | 1.45ms | 10.5ms | | | |
| 0010 | 363us | 2.91ms | 20.9ms | | | | |
| ... (time doubles with every step) | | | | | | | |
| 1010 | 93ms | 744ms | 5.36s | | | | |
| 3:0 | ALCATK [3:0] | 0010 (3.3ms/6dB) | ALC attack (gain ramp-down) time (ALCMODE == 0) | | | | |
| | | | | Per step | Per 6dB | 90% of range | |
| | | | 0000 | 104us | 832us | 6ms | |
| | | | 0001 | 208us | 1.66ms | 12ms | |
| | | | 0010 | 416us | 3.33ms | 24ms | |
| | | | ... (time doubles with every step) | | | | |
| | | | 1010 or higher | 106ms | 852ms | 6.13s | |
| | | | 0010 (726us/6dB) | ALC attack (gain ramp-down) time (ALCMODE == 1) | | | |
| | | | | Per step | Per 6dB | 90% of range | |
| | | | 0000 | 22.7us | 182us | 1.31ms | |
| 0001 | 45.4us | 363us | 2.62ms | | | | |

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION | | | |
|------------------|-----|-------|---------|-------------|--------|-------|--------|
| | | | | 0010 | 90.8us | 726us | 5.23ms |
| 1010 or higher | | | | 23.2ms | 186ms | 1.34s | |

Table 18 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.

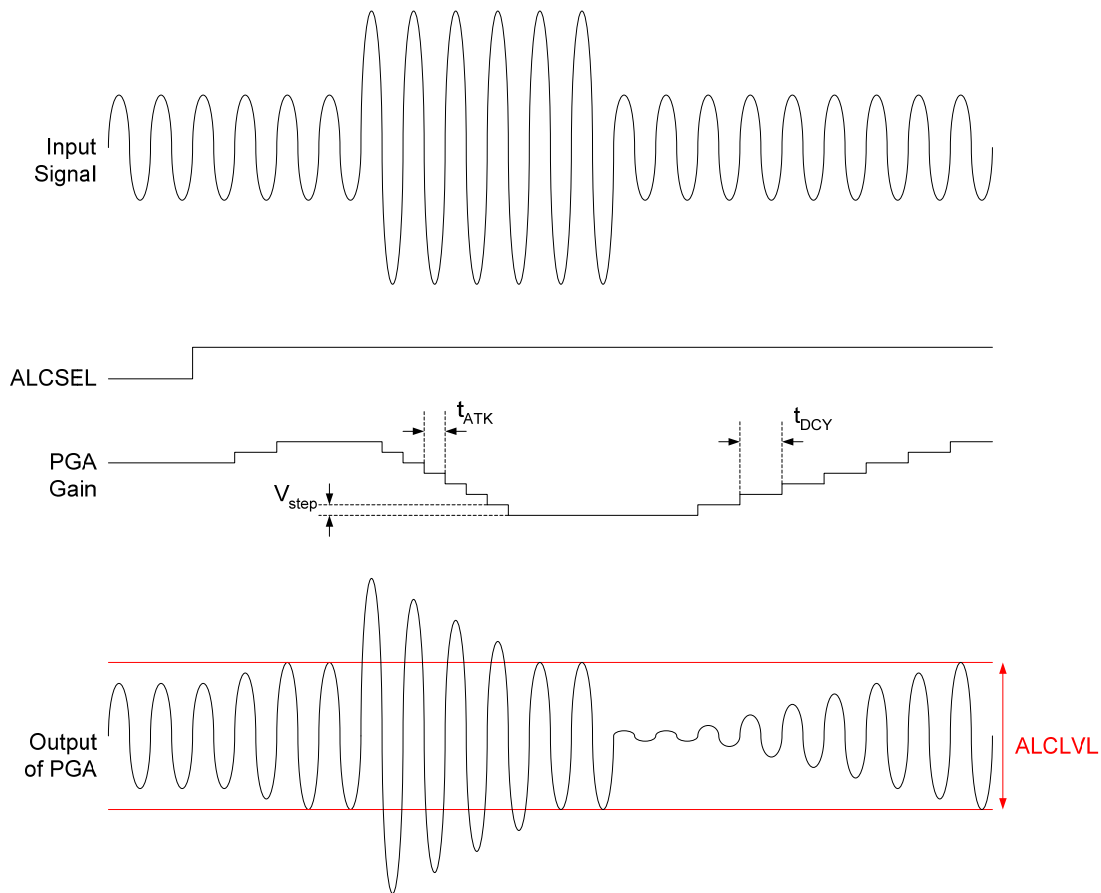


Figure 18 ALC Normal Mode Operation

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.

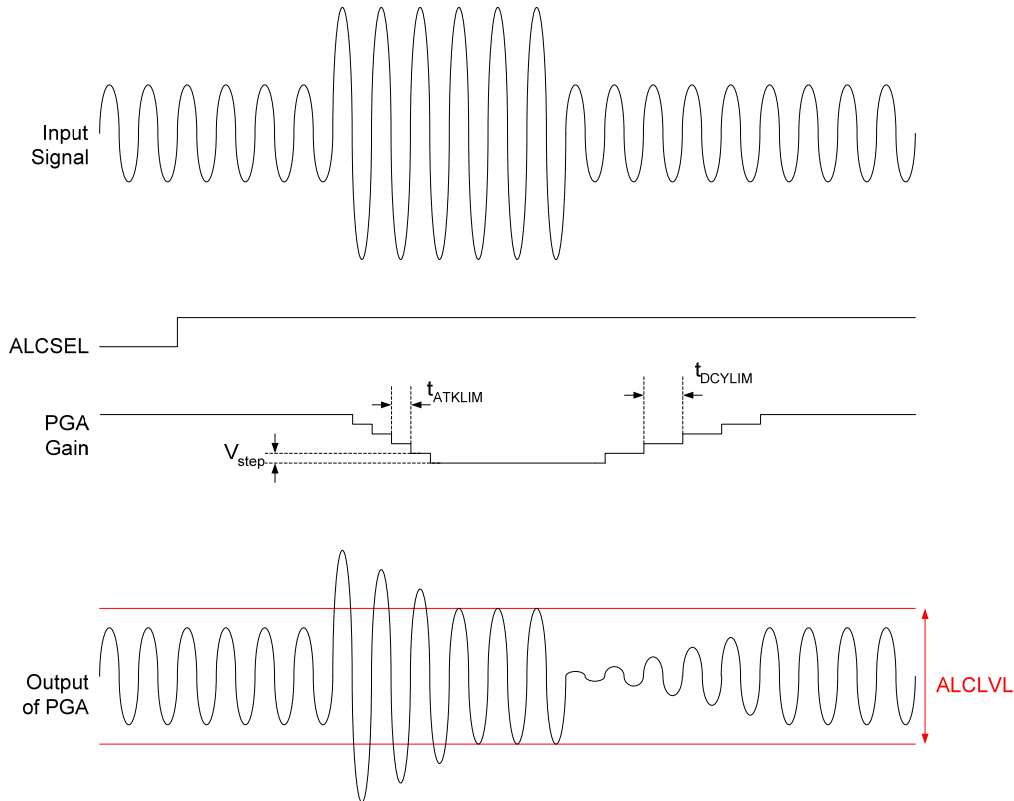


Figure 19 ALC Limiter Mode Operation

ALC LIMITER MODE INITIALISATION SEQUENCE

In order to properly initialise the ALC function, the following sequence of register writes is required:

1. Set INPPGAVOLL to the required input PGA gain (R45[5:0]).
2. Enable analogue inputs (R44[2:0]) as required.
3. Disable INPPGAENL (R2[2] =0).
4. Set ALCMAXGAIN (R32[5:3]) and ALCMINGAIN (R32[2:0]) to the required level for operation.
5. Set ALCLVL (R33[3:0]) to the required level for operation.
6. Set R34 to 0x000.
7. Wait for 1ms to allow the input PGA gain to update by the limiter circuit.
8. Enable Limiter mode (R34[8]=1).
9. Wait for 1ms to allow the input PGA gain to update by the limiter circuit.
10. Enable INPPGAENL (R2[2] =1).

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) | | | |
|---------------------------|------------------|---------------------|---------------------|
| Attack Time (s) | | | |
| ALCATK | t _{ATK} | t _{ATK6dB} | t _{ATK90%} |
| 0000 | 104µs | 832µs | 6ms |
| 0001 | 208µs | 1.66ms | 12ms |
| 0010 | 416µs | 3.33ms | 24ms |
| 0011 | 832µs | 6.66ms | 48ms |
| 0100 | 1.66ms | 13.3ms | 96ms |
| 0101 | 3.33ms | 26.6ms | 192ms |
| 0110 | 6.66ms | 53.2ms | 384ms |
| 0111 | 13.3ms | 106ms | 767ms |
| 1000 | 26.6ms | 213.2ms | 1.53s |
| 1001 | 53.2ms | 426ms | 3.07s |
| 1010 | 106ms | 852ms | 6.13s |

| ALCMODE = 0 (Normal Mode) | | | |
|---------------------------|------------------|---------------------|---------------------|
| Decay Time (s) | | | |
| ALCDCY | t _{DCY} | t _{DCY6dB} | t _{DCY90%} |
| 0000 | 410µs | 3.28ms | 23.6ms |
| 0001 | 820µs | 6.56ms | 47.2ms |
| 0010 | 1.64ms | 13.1ms | 94.5ms |
| 0011 | 3.28ms | 26.2ms | 189ms |
| 0100 | 6.56ms | 52.5ms | 378ms |
| 0101 | 13.1ms | 105ms | 756ms |
| 0110 | 26.2ms | 210ms | 1.51s |
| 0111 | 52.5ms | 420ms | 3.02s |
| 1000 | 105ms | 840ms | 6.05s |
| 1001 | 210ms | 1.68s | 12.1s |
| 1010 | 420ms | 3.36s | 24.2s |

Table 19 ALC Normal Mode (Attack and Decay times)

LIMITER MODE

| ALCMODE = 1 (Limiter Mode) | | | |
|----------------------------|---------------------|------------------------|------------------------|
| Attack Time (s) | | | |
| ALCATK | t _{ATKLIM} | t _{ATKLIM6dB} | t _{ATKLIM90%} |
| 0000 | 22.7µs | 182µs | 1.31ms |
| 0001 | 45.4µS | 363µs | 2.62ms |
| 0010 | 90.8µS | 726µs | 5.23ms |
| 0011 | 182µS | 1.45ms | 10.5ms |
| 0100 | 363µS | 2.91ms | 20.9ms |
| 0101 | 726µS | 5.81ms | 41.8ms |
| 0110 | 1.45ms | 11.6ms | 83.7ms |
| 0111 | 2.9ms | 23.2ms | 167ms |
| 1000 | 5.81ms | 46.5ms | 335ms |
| 1001 | 11.6ms | 93ms | 669ms |
| 1010 | 23.2ms | 186ms | 1.34s |

| ALCMODE = 1 (Limiter Mode) | | | |
|----------------------------|---------------------|------------------------|------------------------|
| Attack Time (s) | | | |
| ALCDCY | t _{DCYLIM} | t _{DCYLIM6dB} | t _{DCYLIM90%} |
| 0000 | 90.8µs | 726µs | 5.23ms |
| 0001 | 182µS | 1.45ms | 10.5ms |
| 0010 | 363µS | 2.91ms | 20.9ms |
| 0011 | 726µS | 5.81ms | 41.8ms |
| 0100 | 1.45ms | 11.6ms | 83.7ms |
| 0101 | 2.91ms | 23.2ms | 167ms |
| 0110 | 5.81ms | 46.5ms | 335ms |
| 0111 | 11.6ms | 93ms | 669ms |
| 1000 | 23.2ms | 186ms | 1.34s |
| 1001 | 46.5ms | 372ms | 2.68s |
| 1010 | 93ms | 744ms | 5.36s |

Table 20 ALC Limiter Mode (Attack and Decay times)

MINIMUM 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.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|------------------|-----|--------|---------|-------------------------|
| R32 | 5:3 | ALCMAX | 111 | Set Maximum Gain of PGA |
| ALC Control 1 | 2:0 | ALCMIN | 000 | Set minimum gain of PGA |

Table 21 ALC Max/Min Gain

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.

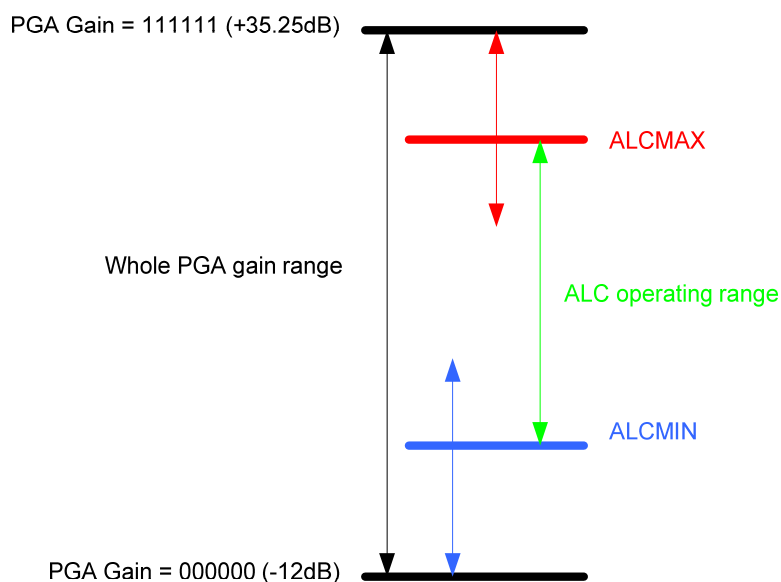


Figure 20 ALC Min/Max Gain

| ALCMAX | Maximum Gain (dB) |
|--------|-------------------|
| 111 | 35.25 |
| 110 | 29.25 |
| 101 | 23.25 |
| 100 | 17.25 |
| 011 | 11.25 |
| 010 | 5.25 |
| 001 | -0.75 |
| 000 | -6.75 |

Table 22 ALC Max Gain Values

| ALCMIN | Minimum Gain (dB) |
|--------|-------------------|
| 000 | -12 |
| 001 | -6 |
| 010 | 0 |
| 011 | 6 |
| 100 | 12 |
| 101 | 18 |
| 110 | 24 |
| 111 | 30 |

Table 23 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.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|-------------------------|-----|--------|---------|---|
| R33 ALC Control 2 | 7:4 | ALCHLD | 0000 | ALC hold time before gain is increased. |

Table 24 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.

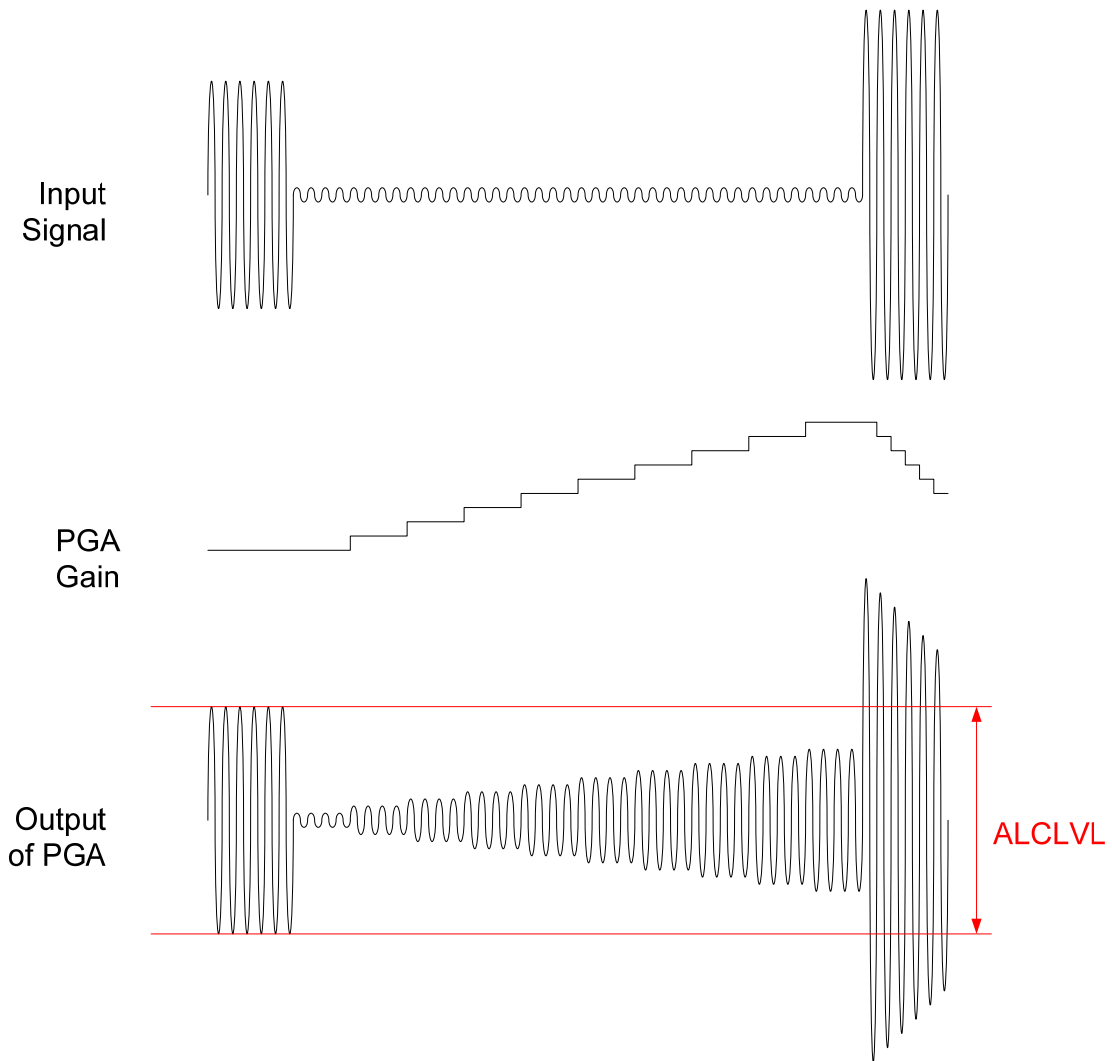


Figure 21 ALCLVL

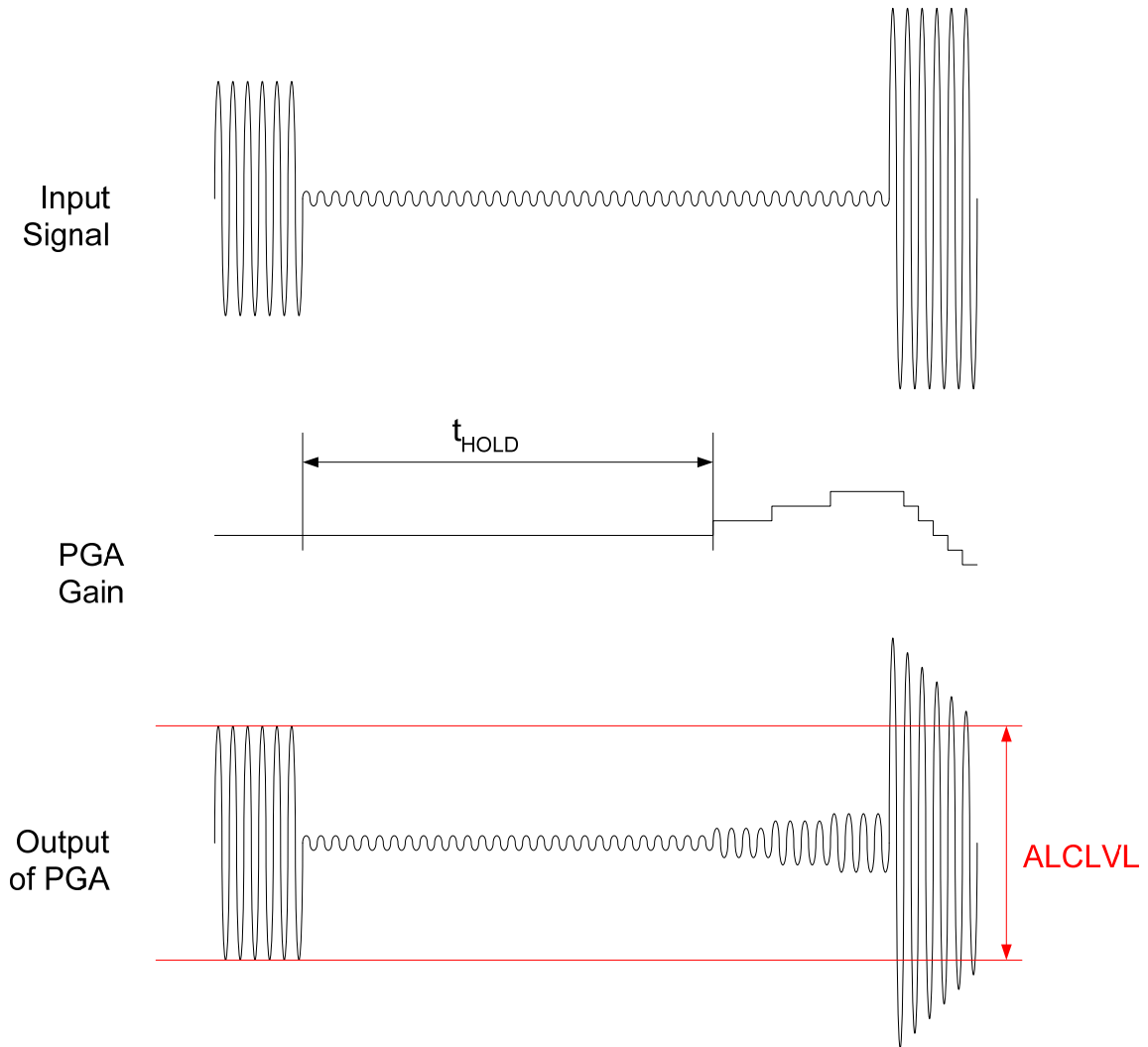


Figure 22 ALC Hold Time

| ALCHLD | t_{HOLD} (s) |
|--------|-----------------------|
| 0000 | 0 |
| 0001 | 2.67ms |
| 0010 | 5.34ms |
| 0011 | 10.7ms |
| 0100 | 21.4ms |
| 0101 | 42.7ms |
| 0110 | 85.4ms |
| 0111 | 171ms |
| 1000 | 342ms |
| 1001 | 684ms |
| 1010 | 1.37s |

Table 25 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 $\text{ALCATK} = 0000$), until the signal level falls below 87.5% of full scale. This function is automatically enabled whenever the ALC is enabled.

Note: If $\text{ALCATK} = 0000$, 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 WM8976 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.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|-------------------------------------|-----|--------|---------|---|
| 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 26 ALC Noise Gate Control

The diagrams below show the response of the system to the same signal with and without noise gate.

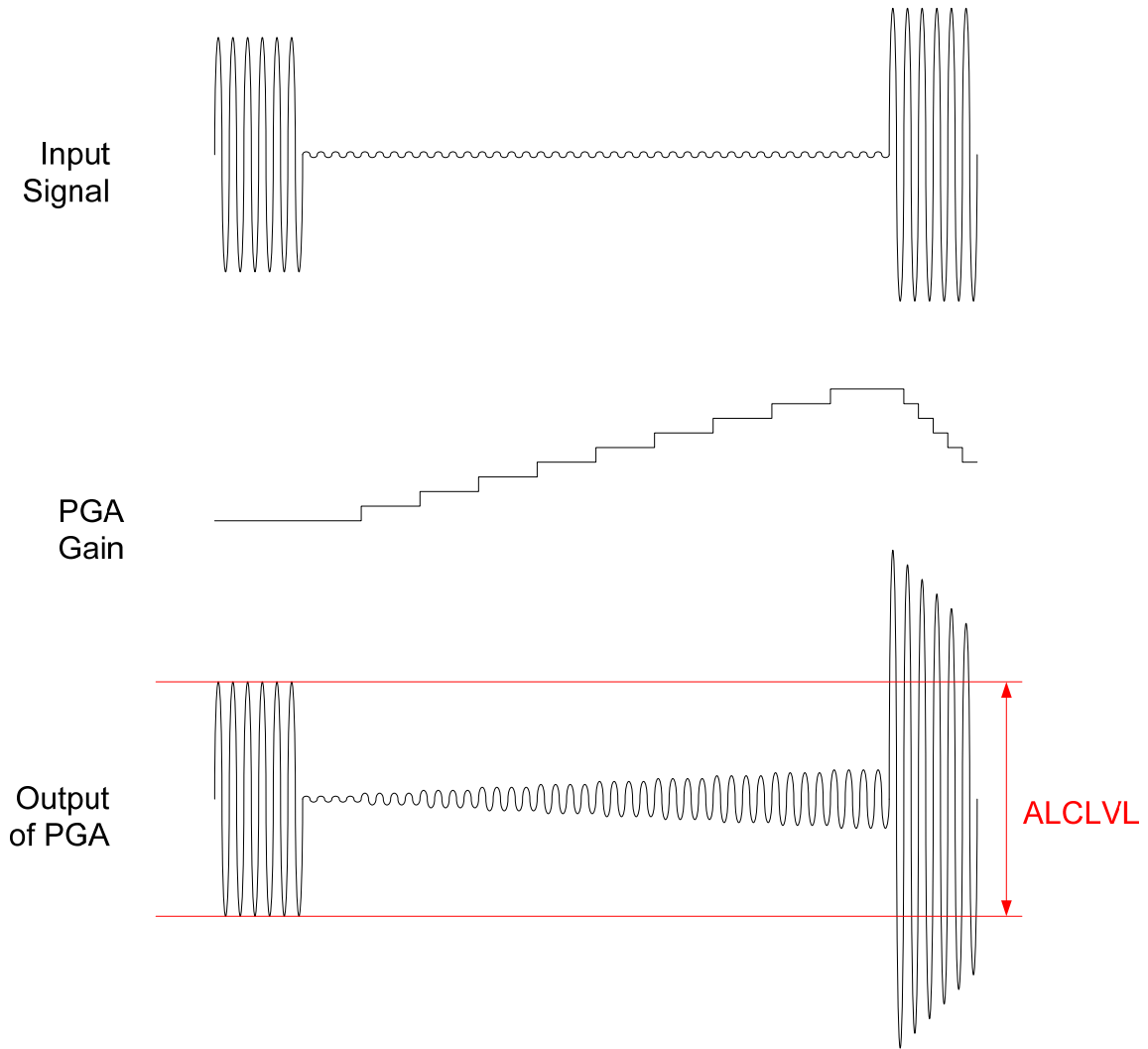


Figure 23 ALC Operation Above Noise Gate Threshold

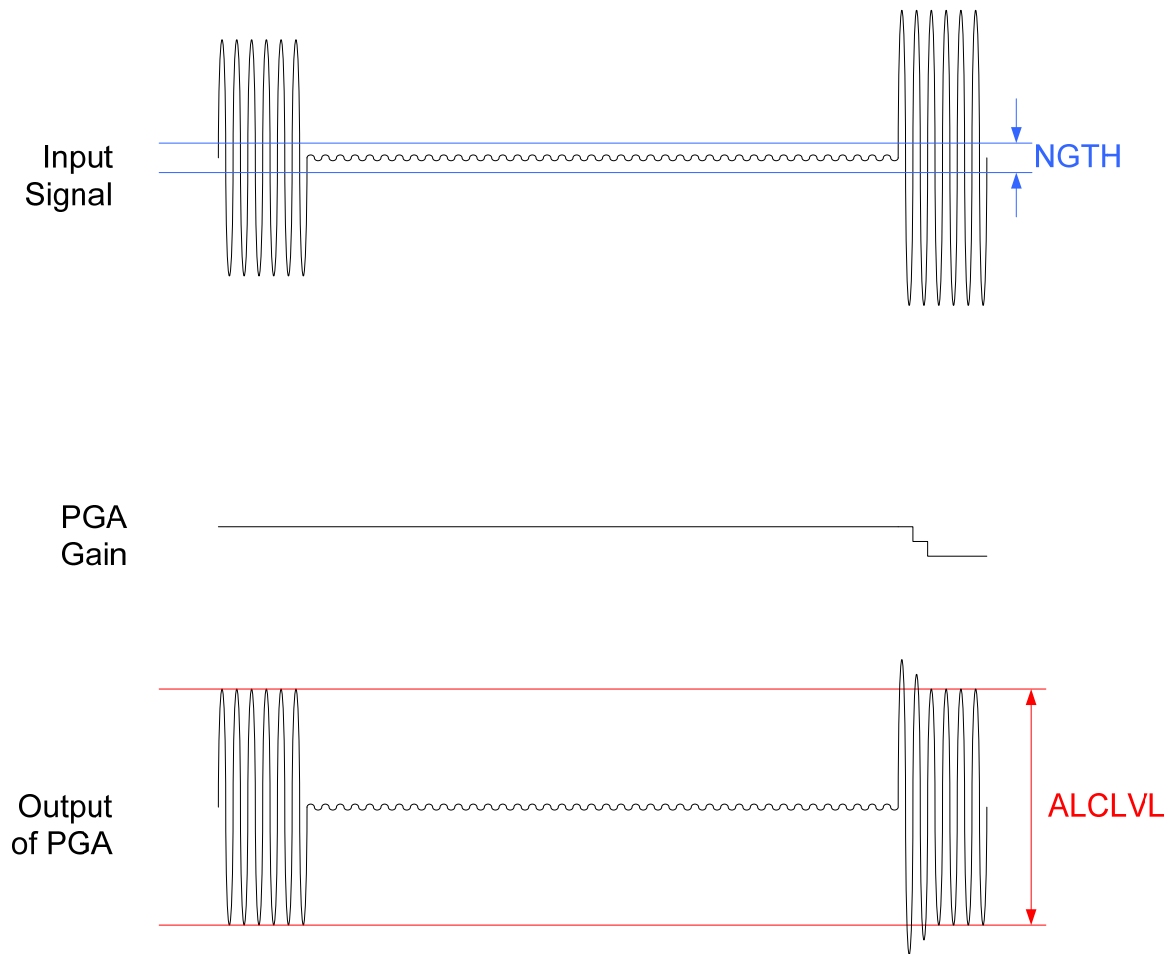


Figure 24 Noise Gate Operation

OUTPUT SIGNAL PATH

The WM8976 output signal paths consist of digital application filters, up-sampling filters, stereo hi-fi DACs, analogue mixers, speaker, stereo headphone and stereo line/mono/midrail output drivers. The digital filters and DAC are enabled by register bits DACENL And DACENR. 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 WM8976, irrespective of whether the DACs are enabled or not.

The WM8976 DACs receive digital input data on the DACDAT pin. The digital filter block processes the data to provide the following functions:

- Digital volume control
- Graphic equaliser
- Digital peak limiter
- Sigma-Delta Modulation

High performance sigma-delta 24-bit audio DAC converts the digital data into an analogue signal.

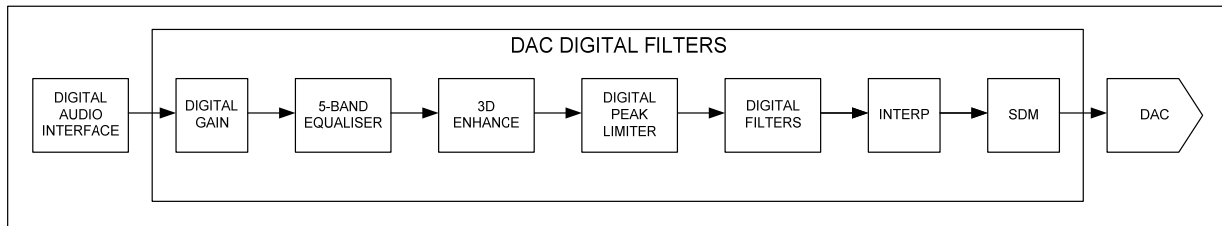


Figure 25 DAC Digital Filter Path

The analogue outputs from the DACs can then be mixed with the aux analogue inputs and the ADC analogue inputs. The mix is fed to the output drivers for headphone (LOUT1/ROUT1), speaker (LOUT2/ROUT2) or line (OUT3/OUT4). OUT3 and OUT4 have additional mixers which allow them to output different signals to the headphone and speaker outputs.

DIGITAL PLAYBACK (DAC) PATH

Digital data is passed to the WM8976 via the flexible audio interface and is then passed through a variety of advanced digital filters (as shown in Figure 25) to the hi-fi DACs. The DACs are enabled by the DACENL/R register bits.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|--------------------------|-----|--------|---------|---|
| R3 Power Management 3 | 0 | DACENL | 0 | Left channel DAC enable 0 = DAC disabled 1 = DAC enabled |
| | 1 | DACENR | 0 | Right channel DAC enable 0 = DAC disabled 1 = DAC enabled |

Table 27 DAC Enable Control

The WM8976 also has a Soft Mute function, which, when enabled, gradually attenuates the volume of the digital signal to zero. When disabled, the gain will ramp back up to the digital gain setting. This function is enabled by default. To play back an audio signal, this function must first be disabled by setting the SOFTMUTE bit to zero.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|--------------------|-----|----------|---------|--|
| R10 DAC Control | 0 | DACPOLL | 0 | Left DAC output polarity: 0 = non-inverted 1 = inverted (180 degrees phase shift) |
| | 1 | DACPOLR | 0 | Right DAC output polarity: 0 = non-inverted 1 = inverted (180 degrees phase shift) |
| | 2 | AMUTE | 0 | Automute enable 0 = Amute disabled 1 = Amute enabled |
| | 3 | DACOSR | 0 | DAC oversampling rate: 0=64x (lowest power) 1=128x (best performance) |
| | 6 | SOFTMUTE | 0 | Softmute enable: 0=Enabled 1=Disabled |

Table 28 DAC Control Register

The digital audio data is converted to oversampled bit streams in the on-chip, true 24-bit digital interpolation filters. The bitstream data enters the multi-bit, sigma-delta DACs, which convert it to a high quality analogue audio signal. The multi-bit DAC architecture reduces high frequency noise and sensitivity to clock jitter. It also uses a Dynamic Element Matching technique for high linearity and low distortion.

The DAC output phase defaults to non-inverted. Setting DACPOLL will invert the DAC output phase on the left channel and DACPOLR inverts the phase on the right channel.

AUTO-MUTE

The DAC has an auto-mute 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.

DIGITAL HI-FI DAC VOLUME (GAIN) 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:

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

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|------------------------------------|-----|------------------|---------------------|--|
| R11 Left DAC Digital Volume | 7:0 | DACVOLL [7:0] | 11111111 (0dB) | Left DAC Digital Volume Control 0000 0000 = Digital Mute 0000 0001 = -127dB 0000 0010 = -126.5dB ... 0.5dB steps up to 1111 1111 = 0dB |
| | 8 | DACVU | Not latched | DAC left and DAC right volume do not update until a 1 is written to DACVU (in reg 11 or 12) |
| R12 Right DAC Digital Volume | 7:0 | DACVOLR [7:0] | 11111111 (0dB) | Right DAC Digital Volume Control 0000 0000 = Digital Mute 0000 0001 = -127dB 0000 0010 = -126.5dB ... 0.5dB steps up to 1111 1111 = 0dB |
| | 8 | DACVU | Not latched | DAC left and DAC right volume do not update until a 1 is written to DACVU (in reg 11 or 12) |

Table 29 DAC Digital Volume Control

Note: An additional gain of up to +12dB can be added using the gain block embedded in the digital peak limiter circuit (see DAC OUTPUT LIMITER section).

5-BAND EQUALISER

A 5-band graphic equaliser function which can be used to change the output frequency levels to suit the environment. This can be applied to the ADC or DAC path and is described in the 5-BAND EQUALISER section for further details on this feature.

3-D ENHANCEMENT

The WM8976 has an advanced digital 3-D enhancement feature which can be used to vary the perceived stereo separation of the left and right channels. See the 3-D STEREO ENHANCEMENT section for further details on this feature.

DAC DIGITAL OUTPUT LIMITER

The WM8976 has a digital output limiter function. The operation of this is shown in Figure 26. In this diagram the upper graph shows the envelope of the input/output signals and the lower graph shows the gain characteristic.

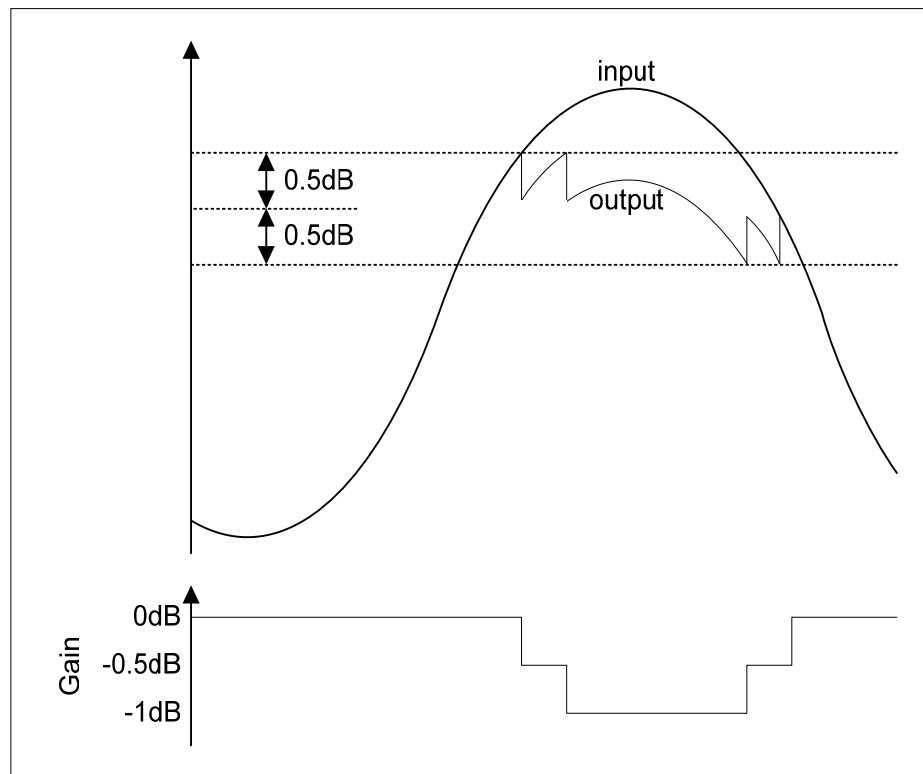


Figure 26 DAC Digital Limiter Operation

The limiter has a programmable upper threshold which is close to 0dB. Referring to Figure 26, 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.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|--------------------------------------|-----|----------|---------|---|
| R24 DAC digital limiter control 1 | 3:0 | LIMATK | 0010 | Limiter Attack time (per 6dB gain change) for 44.1kHz sampling. Note that these will scale proportionally 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 |
| | 7:4 | LIMDCY | 0011 | Limiter Decay time (per 6dB gain change) for 44.1kHz sampling. Note that these will scale proportionally 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 |
| | 8 | LIMEN | 0 | Enable the DAC digital limiter: 0=disabled 1=enabled |
| R25 DAC digital limiter control 2 | 3:0 | LIMBOOST | 0000 | 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 |

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|------------------|-----|--------|---------|--|
| | 6:4 | LIMLVL | 000 | 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 |

Table 30 DAC Digital Limiter Control

5-BAND GRAPHIC EQUALISER

A 5-band graphic equaliser (EQ) is provided, which can be applied to the ADC or DAC path, together with 3D enhancement, under control of the EQ3DMODE register bit.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|---------------------|-----|----------|---------|---|
| R18 EQ Control 1 | 8 | EQ3DMODE | 1 | 0 = Equaliser applied to ADC path 1 = Equaliser and 3D Enhancement applied to DAC path |

Table 31 EQ and 3D Enhancement 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.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|-----------------------------|-----|-------|----------------|--|
| R18 EQ Band 1 Control | 4:0 | EQ1G | 01100 (0dB) | Band 1 Gain Control. See Table 37 for details. |
| | 6:5 | EQ1C | 01 | Band 1 Cut-off Frequency: 00=80Hz 01=105Hz 10=135Hz 11=175Hz |

Table 32 EQ Band 1 Control

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|-----------------------------|-----|-------|----------------|--|
| R19 EQ Band 2 Control | 4:0 | EQ2G | 01100 (0dB) | Band 2 Gain Control. See Table 37 for details. |
| | 6:5 | EQ2C | 01 | Band 2 Centre Frequency: 00=230Hz 01=300Hz 10=385Hz 11=500Hz |
| | 8 | EQ2BW | 0 | Band 2 Bandwidth Control 0=narrow bandwidth 1=wide bandwidth |

Table 33 EQ Band 2 Control

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|-----------------------------|-----|-------|----------------|--|
| R20 EQ Band 3 Control | 4:0 | EQ3G | 01100 (0dB) | Band 3 Gain Control. See Table 37 for details. |
| | 6:5 | EQ3C | 01 | Band 3 Centre Frequency: 00=650Hz 01=850Hz 10=1.1kHz 11=1.4kHz |
| | 8 | EQ3BW | 0 | Band 3 Bandwidth Control 0=narrow bandwidth 1=wide bandwidth |

Table 34 EQ Band 3 Control

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|-----------------------------|-----|-------|----------------|--|
| R21 EQ Band 4 Control | 4:0 | EQ4G | 01100 (0dB) | Band 4 Gain Control. See Table 37 for details |
| | 6:5 | EQ4C | 01 | Band 4 Centre Frequency: 00=1.8kHz 01=2.4kHz 10=3.2kHz 11=4.1kHz |
| | 8 | EQ4BW | 0 | Band 4 Bandwidth Control 0=narrow bandwidth 1=wide bandwidth |

Table 35 EQ Band 4 Control

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|----------------------------------|-----|-------|----------------|--|
| R22 EQ Band 5 Gain Control | 4:0 | EQ5G | 01100 (0dB) | Band 5 Gain Control. See Table 37 for details. |
| | 6:5 | EQ5C | 01 | Band 5 Cut-off Frequency: 00=5.3kHz 01=6.9kHz 10=9kHz 11=11.7kHz |

Table 36 EQ Band 5 Control

| GAIN REGISTER | GAIN |
|------------------|----------|
| 00000 | +12dB |
| 00001 | +11dB |
| 00010 | +10dB |
| (1dB steps) | |
| 01100 | 0dB |
| 01101 | -1dB |
| 11000 | -12dB |
| 11001 to 11111 | Reserved |

Table 37 Gain Register Table

3D STEREO ENHANCEMENT

The WM8976 has a digital 3D enhancement option to increase the perceived separation between the left and right channels. Selection of 3D for playback is controlled by register bit EQ3DMODE. Switching this bit from record to playback or from playback to record may only be done when ADC and DAC are disabled. The WM8976 control interface will only allow EQ3DMODE to be changed when ADC and DAC are disabled (ie ADCENL = 0, ADCENR = 0, DACENL = 0 and DACENR = 0).

The DEPTH3D setting controls the degree of stereo expansion.

When 3D enhancement is used, it may be necessary to attenuate the signal by 6dB to avoid limiting.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|------------------|-----|--------------|---------|--|
| R41 (29h) 3D | 3:0 | DEPTH3D[3:0] | 0000 | Stereo depth 0000: 0% (minimum 3D effect) 0001: 6.67% 1110: 93.3% 1111: 100% (maximum 3D effect) |

Table 38 3D Stereo Enhancement Function

ANALOGUE OUTPUTS

The WM8976 has three sets of stereo analogue outputs. These are:

- LOUT1 and ROUT1 which are normally used to drive a headphone load.
- LOUT2 and ROUT2 – normally used to drive an 8Ω BTL speaker.
- OUT3 and OUT4 – can be configured as a stereo line out (OUT3 is left output and OUT4 is right output). OUT4 can also be used to provide a mono mix of left and right channels.

LOUT2, ROUT2, OUT3 and OUT4 are supplied from SPKVDD and are capable of driving up to 1.5Vrms signals as shown in Figure 27. LOUT1 and ROUT1 are supplied from AVDD and can only drive out a 1V rms signal (AVDD/3.3).

LOUT1, ROUT1, LOUT2 and ROUT2 have individual analogue volume PGAs with -57dB to +6dB ranges.

There are four output mixers in the output signal path, the left and right channel mixers which control the signals to speaker, headphone (and optionally the line outputs) and also dedicated OUT3 and OUT4 mixers.

LEFT AND RIGHT OUTPUT CHANNEL MIXERS

The left and right output channel mixers are shown in Figure 27. These mixers allow the AUX inputs, the ADC bypass and the DAC left and right channels to be combined as desired. This allows a mono mix of the DAC channels to be done as well as mixing in external line-in from the AUX or speech from the input bypass path.

The AUX and bypass inputs have individual volume control from -15dB to +6dB and the DAC volume can be adjusted in the digital domain if required. The output of these mixers is connected to both the headphone (LOUT1 and ROUT1) and speaker (LOUT2 and ROUT2) and can optionally be connected to the OUT3 and OUT4 mixers.

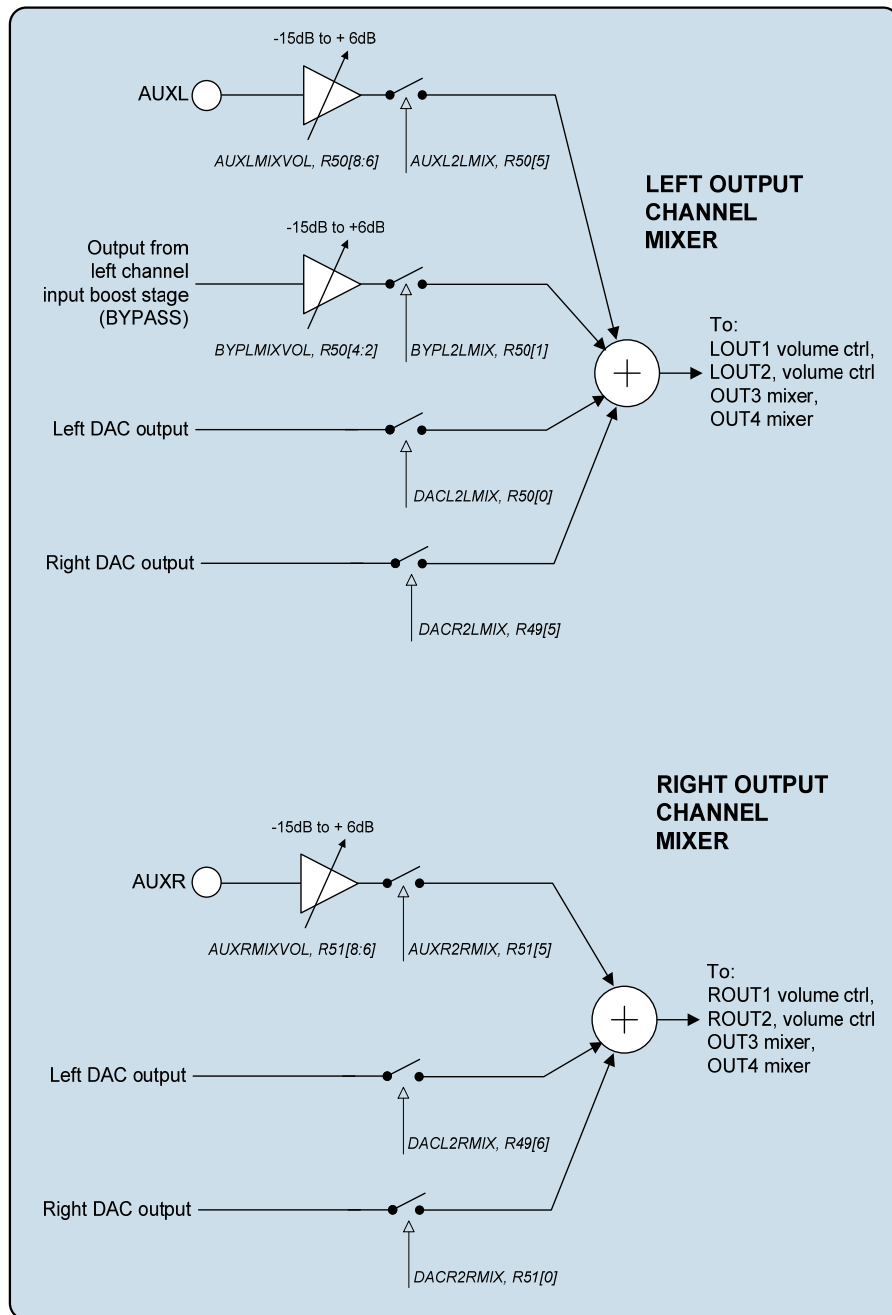


Figure 27 Left/Right Output Channel Mixers

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|--|-----|------------|---------|--|
| R49 Output mixer control | 5 | DACR2LMIX | 0 | Right DAC output to left output mixer 0 = not selected 1 = selected |
| | 6 | DACL2RMIX | 0 | Left DAC output to right output mixer 0 = not selected 1 = selected |
| R50 Left channel output mixer control | 0 | DACL2LMIX | 1 | Left DAC output to left output mixer 0 = not selected 1 = selected |
| | 1 | BYPL2LMIX | 0 | Bypass path (from the input boost output) to left output mixer 0 = not selected 1 = selected |
| | 4:2 | BYPLMIXVOL | 000 | Bypass volume control to output channel mixer: 000 = -15dB 001 = -12dB ... 101 = 0dB 110 = +3dB 111 = +6dB |
| | 5 | AUXL2LMIX | 0 | Left Auxiliary input to left channel output mixer: 0 = not selected 1 = selected |
| | 8:6 | AUXLMIXVOL | 000 | Aux left channel input to left mixer volume control: 000 = -15dB 001 = -12dB ... 101 = 0dB 110 = +3dB 111 = +6dB |

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|---|-----|------------|---------|--|
| R51 Right channel output mixer control | 0 | DACR2RMIX | 1 | Right DAC output to right output mixer 0 = not selected 1 = selected |
| | 4:2 | BYPRMIXVOL | 000 | Right bypass volume control to output channel mixer: 000 = -15dB 001 = -12dB ... 101 = 0dB 110 = +3dB 111 = +6dB |
| | 5 | AUXR2RMIX | 0 | Right Auxiliary input to right channel output mixer: 0 = not selected 1 = selected |
| | 8:6 | AUXRMIXVOL | 000 | Aux right channel input to right mixer volume control: 000 = -15dB 001 = -12dB ... 101 = 0dB 110 = +3dB 111 = +6dB |
| R3 Power management 3 | 2 | LMIXEN | 0 | Left output channel mixer enable: 0 = disabled 1 = enabled |
| | 3 | RMIXEN | 0 | Right output channel mixer enable: 0 = disabled 1 = enabled |

Table 39 Left and Right Output Mixer Control

HEADPHONE OUTPUTS (LOUT1 AND ROUT1)

The headphone outputs, LOUT1 and ROUT1 can drive a 16Ω or 32Ω headphone load, either through DC blocking capacitors, or DC coupled without any capacitor. Each headphone output has an analogue volume control PGA with a gain range of -57dB to +6dB as shown in Figure 30.

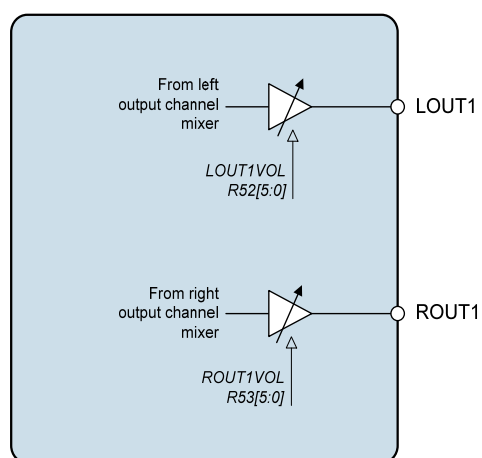
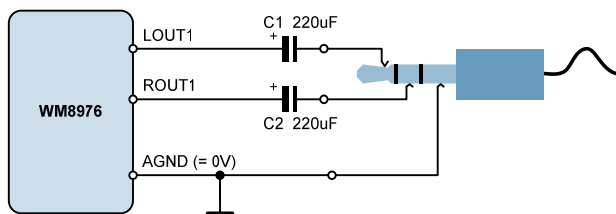


Figure 28 Headphone Outputs LOUT1 and ROUT1

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|--------------------------------|-----|-----------|-------------|--|
| R52 LOUT1 Volume control | 7 | LOUT1ZC | 0 | Headphone volume zero cross enable: 1 = Change gain on zero cross only 0 = Change gain immediately |
| | 6 | LOUT1MUTE | 0 | Left headphone output mute: 0 = Normal operation 1 = Mute |
| | 5:0 | LOUT1VOL | 111001 | Left headphone output volume: 000000 = -57dB ... 111001 = 0dB ... 111111 = +6dB |
| | 8 | HPVU | Not latched | LOUT1 and ROUT1 volumes do not update until a 1 is written to HPVU (in reg 52 or 53) |
| R53 ROUT1 Volume control | 7 | ROUT1ZC | 0 | Headphone volume zero cross enable: 1 = Change gain on zero cross only 0 = Change gain immediately |
| | 6 | ROUT1MUTE | 0 | Right headphone output mute: 0 = Normal operation 1 = Mute |
| | 5:0 | ROUT1VOL | 111001 | Right headphone output volume: 000000 = -57dB ... 111001 = 0dB ... 111111 = +6dB |
| | 8 | HPVU | Not latched | LOUT1 and ROUT1 volumes do not update until a 1 is written to HPVU (in reg 52 or 53) |

Table 40 OUT1 Volume Control

Headphone Output using DC Blocking Capacitors:



DC Coupled Headphone Output:

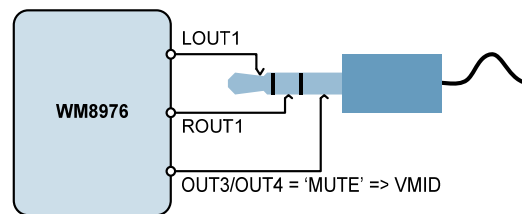


Figure 29 Recommended Headphone Output Configurations

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 C1, C2 = 220µF:

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

In the DC coupled configuration, the headphone “ground” is connected to the VMID pin. The OUT3/4 pins can be configured as a DC output driver by setting the OUT3MUTE and OUT4MUTE register bit. The DC voltage on VMID in this configuration is equal to the DC offset on the LOUT1 and ROUT1 pins therefore no DC blocking capacitors are required. This saves space and material cost in portable applications.

Note that OUT3 and OUT4 have an optional output boost of 1.5x. When these are configured in this output boost mode (OUT3BOOST/OUT4BOOST=1) then the VMID value of these outputs will be equal to $1.5 \times AVDD/2$ and will not match the VMID of the headphone drivers. Do not use the DC coupled output mode in this configuration.

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.

SPEAKER OUTPUTS (LOUT2 AND ROUT2)

The outputs LOUT2 and ROUT2 are designed to drive an 8Ω BTL speaker but can optionally drive two headphone loads of 16Ω/32Ω or a line output (see Headphone Output and Line Output sections, respectively). Each output has an individual volume control PGA, an output boost/level shift bit, a mute and an enable as shown in Figure 30. LOUT2 and ROUT2 output the left and right channel mixer outputs respectively.

The ROUT2 signal path also has an optional invert. The amplifier used for this invert can be used to mix in the AUXR signal with an adjustable gain range of -15dB -> +6dB. This allows a ‘beep’ signal to be applied only to the speaker output without affecting the HP or line outputs.

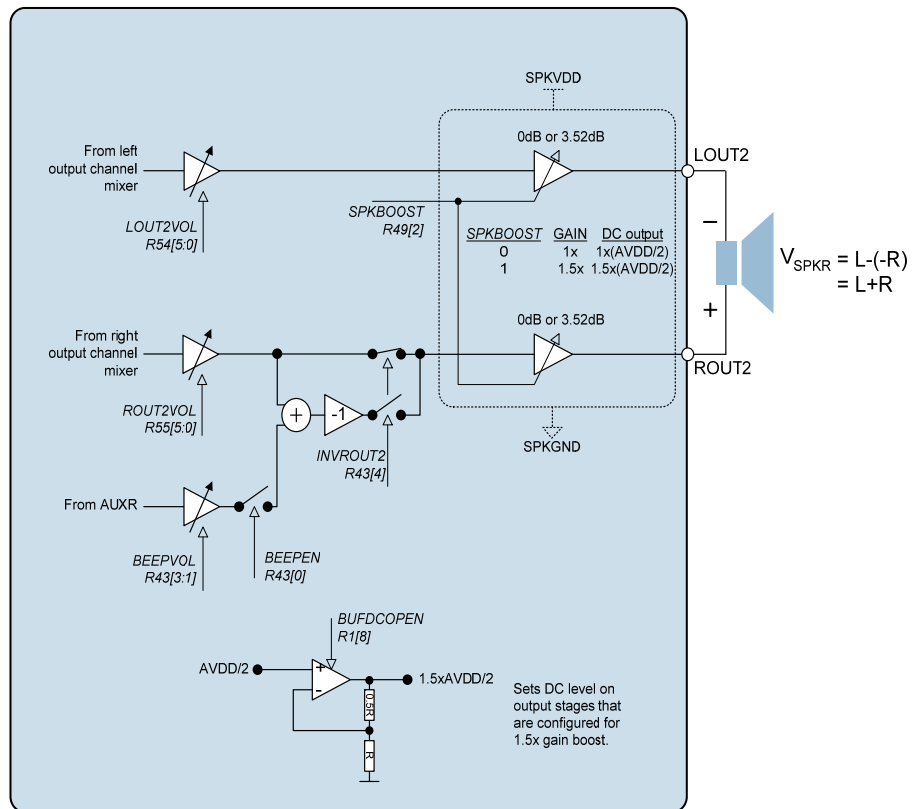


Figure 30 Speaker Outputs LOUT2 and ROUT2

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|--------------------------------------|-----|-----------|-------------|--|
| R54 LOUT2 (SPK) Volume control | 7 | LOUT2ZC | 0 | Speaker volume zero cross enable: 1 = Change gain on zero cross only 0 = Change gain immediately |
| | 6 | LOUT2MUTE | 0 | Left speaker output mute: 0 = Normal operation 1 = Mute |
| | 5:0 | LOUT2VOL | 111001 | Left speaker output volume: 000000 = -57dB ... 111001 = 0dB ... 111111 = +6dB |
| | 8 | SPKVU | Not latched | LOUT2 and ROUT2 volumes do not update until a 1 is written to SPKVU (in reg 54 or 55) |
| R55 ROUT2 (SPK) Volume control | 7 | ROUT2ZC | 0 | Speaker volume zero cross enable: 1 = Change gain on zero cross only 0 = Change gain immediately |
| | 6 | ROUT2MUTE | 0 | Right speaker output mute: 0 = Normal operation 1 = Mute |
| | 5:0 | ROUT2VOL | 111001 | Right speaker output volume: 000000 = -57dB ... 111001 = 0dB ... 111111 = +6dB |
| | 8 | SPKVU | Not latched | LOUT2 and ROUT2 volumes do not update until a 1 is written to SPKVU (in reg 54 or 55) |

Table 41 Speaker Volume Control

The signal output on LOUT2/ROUT2 comes from the Left/Right Mixer circuits and can be any combination of the DAC output, the Bypass path (output of the input boost stage) and the AUX input. The LOUT2/ROUT2 volume is controlled by the LOUT2VOL/ ROUT2VOL register bits. Gains over 0dB may cause clipping if the signal is large. The LOUT2MUTE/ ROUT2MUTE register bits cause the speaker outputs to be muted (the output DC level is driven out). The output pins remain at the same DC level (DCOP), so that no click noise is produced when muting or un-muting.

The speaker output stages also have a selectable gain boost of 1.5x (3.52dB). 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 BUFDCOP, as shown in Figure 30, 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 43 summarises the effect of the SPKBOOST control bits.

Note: When boost mode is selected, it is necessary to set LOUT2MUTE (R54[6]) and ROUT2MUTE (R55[6]) bits for either output to be muted

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|-----------------------------|-----|-----------|---------|---|
| R49 Output control | 2 | SPKBOOST | 0 | 0 = speaker gain = -1; DC = AVDD / 2 1 = speaker gain = +1.5; DC = 1.5 x AVDD / 2 |
| R1 Power management 1 | 8 | BUFDCOPEN | 0 | 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) |

Table 42 Speaker Boost Stage Control

| SPKBOOST | OUTPUT STAGE GAIN | OUTPUT DC LEVEL | OUTPUT STAGE CONFIGURATION |
|----------|-------------------|-----------------|----------------------------|
| 0 | 1x (0dB) | AVDD/2 | Inverting |
| 1 | 1.5x (3.52dB) | 1.5xAVDD/2 | Non-inverting |

Table 43 Output Boost Stage Details

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|---------------------|-----|--------------|---------|---|
| R43 Beep control | 5 | MUTERPGA2INV | 0 | Mute input to INVROUT2 mixer |
| | 4 | INVROUT2 | 0 | Invert ROUT2 output |
| | 3:1 | BEEPVOL | 000 | AUXR input to ROUT2 inverter gain 000 = -15dB ... 111 = +6dB |
| | 0 | BEEPEN | 0 | 0 = mute AUXR beep input 1 = enable AUXR beep input |

Table 44 AUXR – ROUT2 BEEP Mixer Function

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} * input clock period.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|--------------------------|-----|-----------|---------|---|
| R7 Additional Control | 0 | SLOWCLKEN | 0 | Slow clock enable. Used for both the jack insert detect debounce circuit and the zero cross timeout. 0 = slow clock disabled 1 = slow clock enabled |

Table 45 Timeout Clock Enable Control

OUT3/OUT4 MIXERS AND OUTPUT STAGES

The OUT3/OUT4 pins can provide an additional stereo line output, a mono output, or a pseudo ground connection for headphones. There is a dedicated analogue mixer for OUT3 and one for OUT4 as shown in

Figure 31.

The OUT3 and OUT4 output stages are powered from SPKVDD and SPKGND. The individually controllable outputs also incorporate an optional 1.5x boost and level shifting stage.

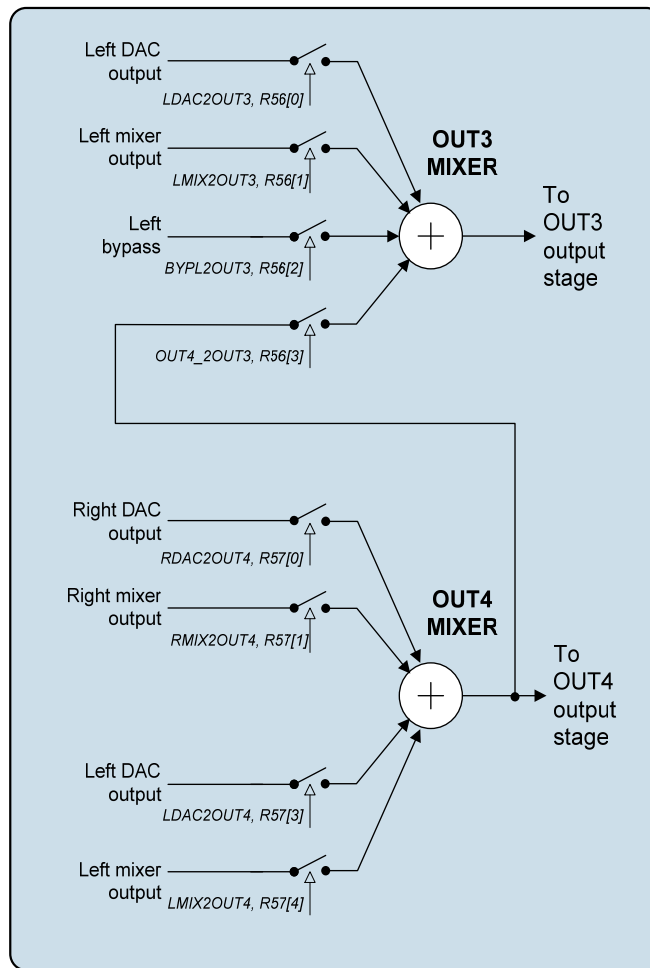


Figure 31 OUT3 and OUT4 Mixers

OUT3 can provide a buffered midrail headphone pseudo-ground, or a left line output.

OUT4 can provide a buffered midrail headphone pseudo-ground, a right line output, or a mono mix output.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|---------------------------|-----|------------|---------|---|
| R56 OUT3 mixer control | 6 | OUT3MUTE | 0 | 0 = Output stage outputs OUT3 mixer 1 = Output stage muted – drives out VMID. Can be used as VMID buffer in this mode. |
| | 3 | OUT4_2OUT3 | 0 | OUT4 mixer output to OUT3 0 = disabled 1 = enabled |
| | 2 | BYPL2OUT3 | 0 | ADC input to OUT3 0 = disabled 1 = enabled |
| | 1 | LMIX2OUT3 | 0 | Left DAC mixer to OUT3 0 = disabled 1 = enabled |
| | 0 | LDAC2OUT3 | 1 | Left DAC output to OUT3 0 = disabled 1 = enabled |
| R57 OUT4 mixer control | 6 | OUT4MUTE | 0 | 0 = Output stage outputs OUT4 mixer 1 = Output stage muted – drives out VMID. Can be used as VMID buffer in this mode. |
| | 5 | HALFSIG | 0 | 0=OUT4 normal output 1=OUT4 attenuated by 6dB |
| | 4 | LMIX2OUT4 | 0 | Left DAC mixer to OUT4 0 = disabled 1 = enabled |
| | 3 | LDAC2OUT4 | 0 | Left DAC to OUT4 0 = disabled 1 = enabled |
| | 1 | RMIX2OUT4 | 0 | Right DAC mixer to OUT4 0 = disabled 1 = enabled |
| | 0 | RDAC2OUT4 | 1 | Right DAC output to OUT4 0 = disabled 1 = enabled |

Table 46 OUT3/OUT4 Mixer Registers

The OUT3 and OUT4 output stages each have a selectable gain boost of 1.5x (3.52dB). 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 BUFDCOP, as shown in Figure 32, 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 43 summarises the effect of the OUT3BOOST and OUT4BOOST control bits.

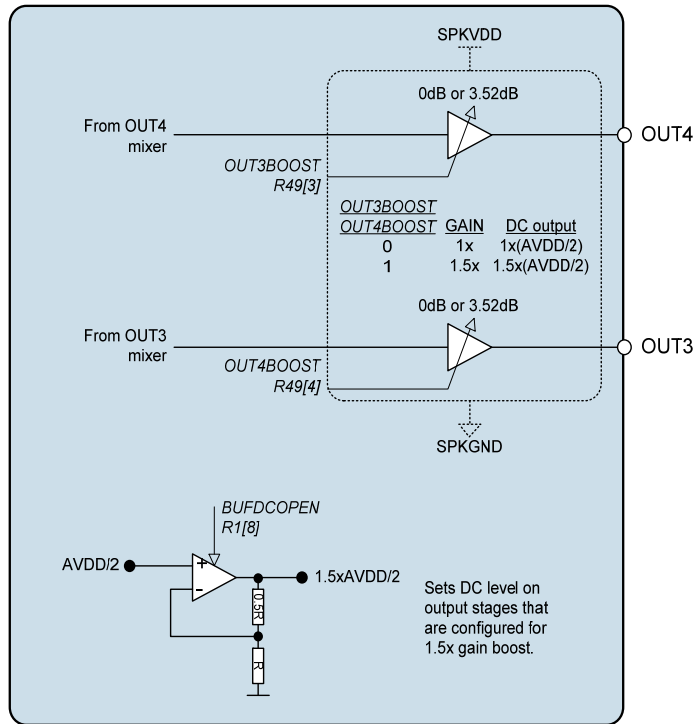


Figure 32 Outputs OUT3 and OUT4

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|-----------------------------|-----|-----------|---------|---|
| R49 Output control | 3 | OUT3BOOST | 0 | 0 = OUT3 output gain = -1; DC = AVDD / 2 1 = OUT3 output gain = +1.5 DC = 1.5 x AVDD / 2 |
| | 4 | OUT4BOOST | 0 | 0 = OUT4 output gain = -1; DC = AVDD / 2 1 = OUT4 output gain = +1.5 DC = 1.5 x AVDD / 2 |
| R1 Power management 1 | 8 | BUFDCOPEN | 0 | 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) |

Table 47 OUT3 and OUT4 Boost Stages Control

| OUT3BOOST/ OUT4BOOST | OUTPUT STAGE GAIN | OUTPUT DC LEVEL | OUTPUT STAGE CONFIGURATION |
|-------------------------|----------------------|--------------------|-------------------------------|
| 0 | 1x | AVDD/2 | Inverting |
| 1 | 1.5x | 1.5xAVDD/2 | Non-inverting |

Table 48 OUT3/OUT4 Output Boost Stage Details

OUTPUT PHASING

The relative phases of the analogue outputs will depend upon the following factors:

1. DACPOLL and DACPOLR invert bits: Setting these bits to 1 will invert the DAC output.
2. Mixer configuration: The polarity of the signal will depend upon the route through the mixer path. For example, DACL can be directly input to the OUT3 mixer, giving a 180° phase shift at the OUT3 mixer output. However, if DACL is input to the OUT3 mixer via the left mixer, an additional phase shift will be introduced, giving 0° phase shift at the OUT3 mixer output.
3. Output boost set-up: When 1.5x boost is enabled on an output, no phase shift occurs. When 1.5x boost is not enabled, a 180° phase shift occurs.

Figure 27 shows where these phase inversions can occur in the output signal path.

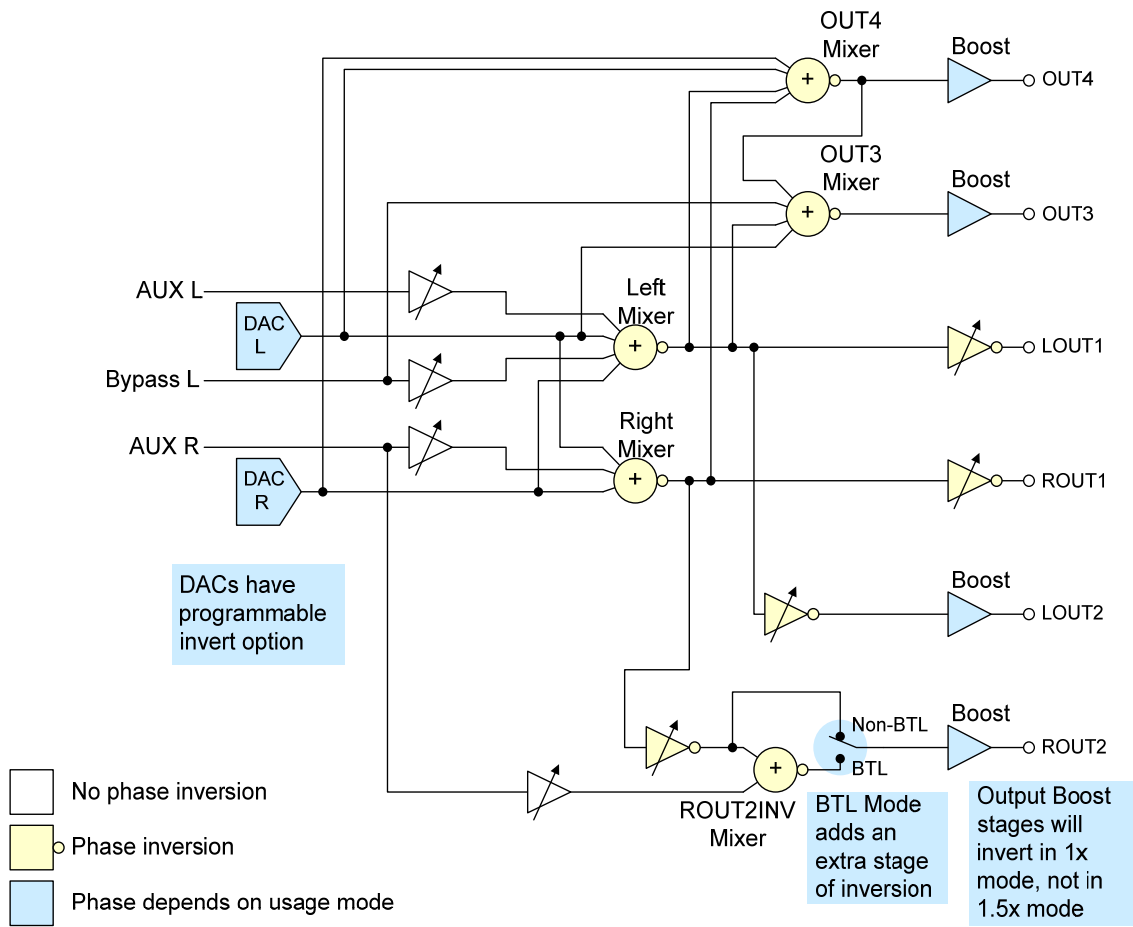


Figure 33 Output Signal Path Phasing

Table 49 shows the polarities of the outputs in various configurations.

Unless otherwise stated, polarity is shown with respect to left DAC output in non-inverting mode.

Note that only registers relating to the mixer paths are shown here (Mixer enables, volume settings, output enables etc are not shown).

| CONFIGURATION | DACPOLL | DACPOLR | INROUT2 | SPKBOOST | OUT3BOOST | OUT4BOOST | MIXER PATH REGISTERS DIFFERENT FROM DEFAULT | OUT4 PHASE / MAG | OUT3 PHASE / MAG | LOUT1 PHASE / MAG | ROUT1 PHASE / MAG | LOUT2 PHASE / MAG | ROUT2 PHASE / MAG |
|---|---------|---------|---------|----------|-----------|-----------|--|------------------------|------------------------|-------------------------|-------------------------|-------------------------|-------------------------|
| Default: Stereo DAC playback to LOUT1/ROUT1, LOUT2/ROUT2 and OUT4/OUT3 | 0 | 0 | 0 | 0 | 0 | 0 | | 0° 1 | 0° 1 | 0° 1 | 0° 1 | 180° 1 | 180° 1 |
| DACs inverted | 1 | 1 | 0 | 0 | 0 | 0 | | 180° 1 | 180° 1 | 180° 1 | 180° 1 | 0° 1 | 0° 1 |
| Stereo DAC playback to LOUT1/ROUT1 and LOUT2/ROUT2 and OUT4/OUT3 (Speaker boost enabled) | 0 | 0 | 0 | 1 | 0 | 0 | | 0° 1 | 0° 1 | 0° 1 | 0° 1 | 0° 1.5 | 0° 1.5 |
| Stereo DAC playback to LOUT1/ROUT1 and LOUT2/ROUT2 and OUT4/OUT3 (OUT3 and OUT4 boost enabled) | 0 | 0 | 0 | 0 | 1 | 1 | | 180° 1.5 | 180° 1.5 | 0° 1 | 0° 1 | 180° 1 | 180° 1 |
| Stereo playback to OUT3/OUT4 (DACs input to OUT3/OUT4 mixers via left/right mixers) | 0 | 0 | 0 | 0 | 0 | 0 | LDAC2OUT3=0 RDAC2OUT4=0 LMIX2OUT3=1 RMIX2OUT4=1 | 180° 1 | 180° 1 | 0° 1 | 0° 1 | 180° 1 | 180° 1 |
| Differential output of mono mix of DACs via LOUT2/ROUT2 (e.g. BTL speaker drive) | 0 | 0 | 1 | 0 | 0 | 0 | | 0° 1 | 0° 1 | 0° 1 | 0° 1 | 180° 1 | 0° 1 |
| High power speaker drive | 0 | 0 | 1 | 1 | 0 | 0 | | 0° 1 | 0° 1 | 0° 1 | 0° 1 | 0° 1.5 | 180° 1.5 |

Table 49 Relative Output Phases

Note that differential output should not be set up by combining outputs in boost mode with outputs which are not in boost mode as this would cause a DC offset current on the outputs.

ENABLING THE OUTPUTS

Each analogue output of the WM8976 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 WM8976 should remain disabled.

Outputs can be enabled at any time, but it is not recommended to do so when BUFIO is disabled (BUFIOEN=0) or when BUFDCOP is disabled (BUFDCOPEN=0) when configured in output boost mode, as this may cause pop noise (see "Power Management" and "Applications Information" sections).

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|--------------------------|-----|-----------|---------|--|
| R1 Power Management 1 | 2 | BUFIOEN | 0 | Unused input/output tie off buffer enable |
| | 6 | OUT3MIXEN | 0 | OUT3 mixer enable |
| | 7 | OUT4MIXEN | 0 | OUT4 mixer enable |
| | 8 | BUFDCOPEN | 0 | Output stage 1.5xAVDD/2 driver enable |
| R2 Power Management 2 | 8 | ROUT1EN | 0 | ROUT1 output enable |
| | 7 | LOUT1EN | 0 | LOUT1 output enable |
| | 6 | SLEEP | 0 | 0 = normal device operation 1 = residual current reduced in device standby mode if clocks still running |
| R3 Power Management 3 | 2 | LMIXEN | 0 | Left mixer enable |
| | 3 | RMIXEN | 0 | Right mixer enable |
| | 5 | ROUT2EN | 0 | ROUT2 output enable |
| | 6 | LOUT2EN | 0 | LOUT2 output enable |
| | 7 | OUT3EN | 0 | OUT3 enable |
| | 8 | OUT4EN | 0 | OUT4 enable |

Note: All "Enable" bits are 1 = ON, 0 = OFF

Table 50 Output Stages Power Management Control

THERMAL SHUTDOWN

The speaker outputs can drive very large currents. To protect the WM8976 from overheating a thermal shutdown circuit is included. If the device temperature reaches approximately 125°C and the thermal shutdown circuit is enabled (TSDEN=1) then the speaker amplifiers will be disabled if TSDEN is set. The thermal shutdown may also be configured to generate an interrupt. See the GPIO and Interrupt Controller section for details.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|-----------------------|-----|-------|---------|--|
| R49 Output control | 1 | TSDEN | 1 | Thermal Shutdown Enable 0 : thermal shutdown disabled 1 : thermal shutdown enabled |

Table 51 Thermal Shutdown

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.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|------------------|-----|-------|---------|---|
| R49 | 0 | VROI | 0 | VREF (AVDD/2 or 1.5xAVDD/2) to analogue output resistance 0: approx 1kΩ 1: approx 30 kΩ |

Table 52 Disabled Outputs to VREF Resistance

A dedicated buffer is available for tying off unused analogue I/O pins as shown in Figure 34. This buffer can be enabled using the BUFIOEN register bit.

If the SPKBOOST, OUT3BOOST or OUT4BOOST bits are set then the relevant outputs will be tied to the output of the DC level shift buffer at $1.5 \times AVDD/2$ when disabled.

Figure 34 summarises the tie-off options for the speaker and mono output pins.

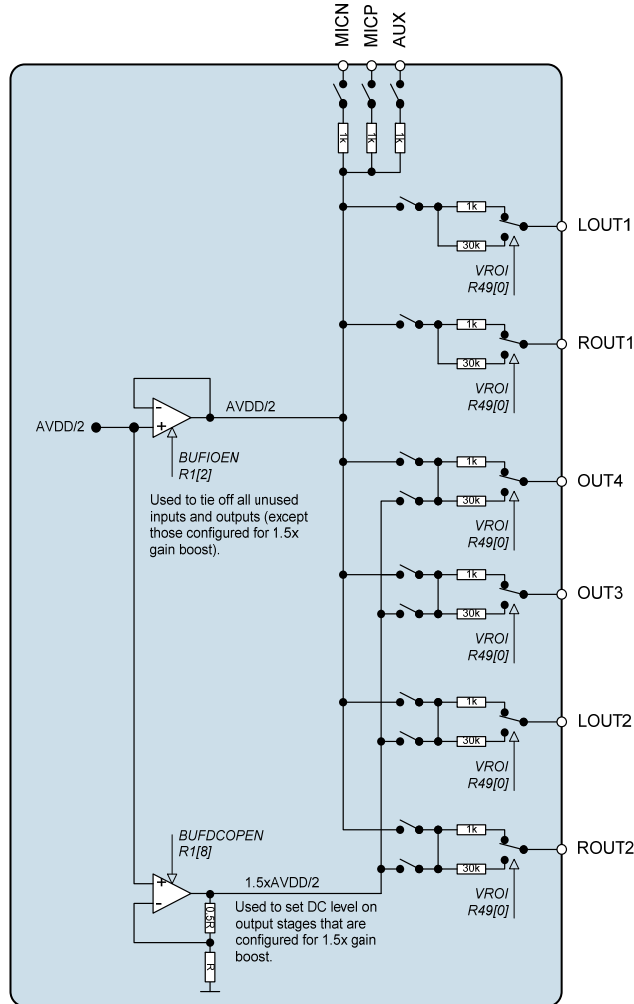


Figure 34 Unused Input/Output Pin Tie-off Buffers

| L/ROUT2EN/ OUT3/4EN | OUT3BOOST/ OUT4BOOST/ SPKBOOST | VROI | OUTPUT CONFIGURATION |
|------------------------|--------------------------------------|------|---|
| 0 | 0 | 0 | 1kΩ tie-off to AVDD/2 |
| 0 | 0 | 1 | 30kΩ tie-off to AVDD/2 |
| 0 | 1 | 0 | 1kΩ tie-off to $1.5 \times AVDD/2$ |
| 0 | 1 | 1 | 30kΩ tie-off to $1.5 \times AVDD/2$ |
| 1 | 0 | X | Output enabled (DC level=AVDD/2) |
| 1 | 1 | X | Output enabled (DC level= $1.5 \times AVDD/2$) |

Table 53 Unused Output Pin Tie-off Options

DIGITAL AUDIO INTERFACES

The audio interface has four pins:

- ADCDAT: ADC data output
- DACDAT: DAC data input
- LRC: Data Left/Right alignment clock
- BCLK: Bit clock, for synchronisation

The clock signals BCLK, and LRC can be outputs when the WM8976 operates as a master, or inputs when it is a slave (see Master and Slave Mode Operation, below).

Five different audio data formats are supported:

- Left justified
- Right justified
- I²S
- DSP mode A
- DSP mode B

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 WM8976 audio interface may be configured as either master or slave. As a master interface device the WM8976 generates BCLK and LRC 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 WM8976 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 LRC 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 LRC transition.

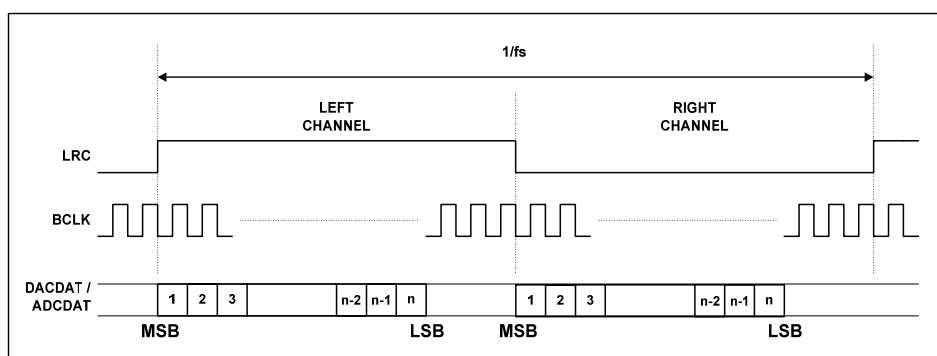


Figure 35 Left Justified Audio Interface (assuming n-bit word length)

In Right Justified mode, the LSB is available on the last rising edge of BCLK before a LRC 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 LRC transition.

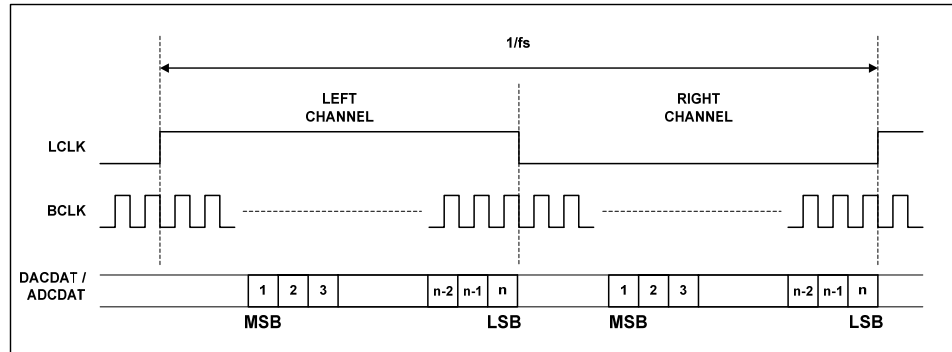


Figure 36 Right Justified Audio Interface (assuming n-bit word length)

In I²S mode, the MSB is available on the second rising edge of BCLK following a LRC 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.

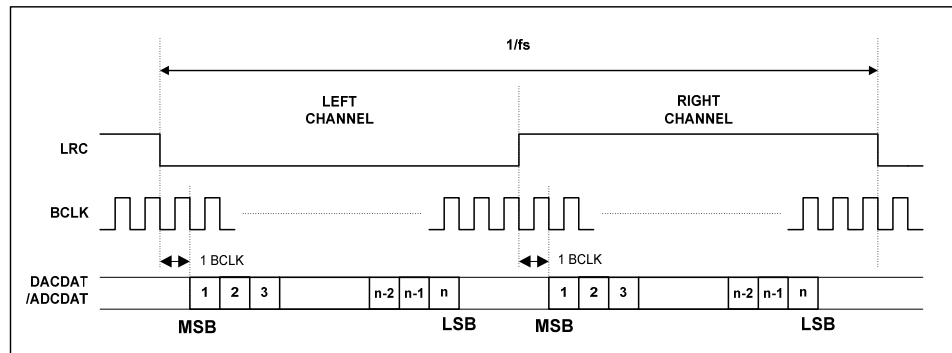


Figure 37 I²S Audio Interface (assuming n-bit word length)

In DSP/PCM mode, the left channel MSB is available on either the 1st (mode B) or 2nd (mode A) rising edge of BCLK (selectable by LRP) following a rising edge of LRC. 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.

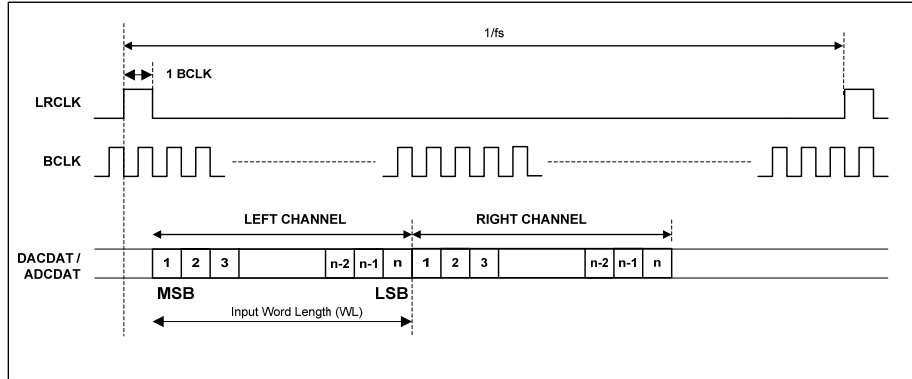


Figure 38 DSP/PCM Mode Audio Interface (mode A, LRP=0)

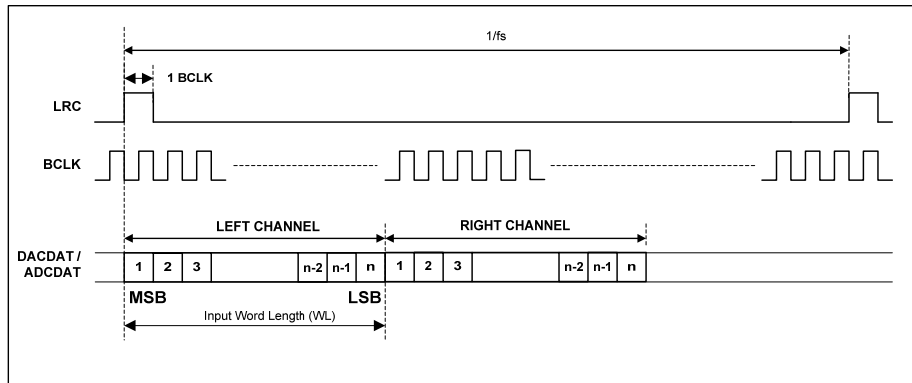


Figure 39 DSP/PCM Mode Audio Interface (mode B, LRP=1)

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|-------------------------------------|-----|-----------|---------|---|
| R4 Audio Interface Control | 0 | DACMONO | 0 | Selects between stereo and mono DAC operation: 0=Stereo device operation 1=Mono device operation. DAC data appears in 'left' phase of LRC |
| | 1 | ADCLRSWAP | 0 | Controls whether ADC data appears in 'right' or 'left' phases of LRC clock: 0=ADC data appear in 'left' phase of LRC 1=ADC data appears in 'right' phase of LRC |
| | 2 | DACLRSWAP | 0 | Controls whether DAC data appears in 'right' or 'left' phases of LRC clock: 0=DAC data appear in 'left' phase of LRC 1=DAC data appears in 'right' phase of LRC |
| | 4:3 | FMT | 10 | Audio interface Data Format Select: 00=Right Justified 01=Left Justified 10=i ² S format 11= DSP/PCM mode |
| | 6:5 | WL | 10 | Word length 00=16 bits 01=20 bits 10=24 bits 11=32 bits (see note) |
| | 7 | LRP | | right, left and i ² s modes – LRCLK polarity 1 = invert LRCLK polarity 0 = normal LRCLK polarity DSP Mode – mode A/B select 1 = MSB is available on 1st BCLK rising edge after LRC rising edge (mode B) 0 = MSB is available on 2nd BCLK rising edge after LRC rising edge (mode A) |
| | 8 | BCP | | BCLK polarity 0=normal 1=inverted |

Table 54 Audio Interface Control

ADCLRSWAP bit controls whether the ADC data appears in the right or left phase of the LRC clock as defined for each audio format. Similarly, DACLRSWAP can be used to swap the left DAC data from the left phase to the right phase of the LRC clock and the right DAC data from the right phase to the left phase of the LRC clock.

Note: Right Justified Mode will only operate with a maximum of 24 bits. If 32-bit mode is selected, the device will operate in 24-bit mode.

AUDIO INTERFACE CONTROL

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

Register bit MS selects audio interface operation in master or slave mode. In Master mode BCLK, and LRC are outputs. The frequency of BCLK in master mode are controlled with BCLKDIV. These are divided down versions of master clock.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|--------------------------------------|-----|---------|---------|--|
| R6 Clock Generation Control | 0 | MS | 0 | Sets the chip to be master over LRC and BCLK 0=BCLK and LRC clock are inputs 1=BCLK and LRC clock are outputs generated by the WM8976 (MASTER) |
| | 4:2 | BCLKDIV | 000 | Configures the BCLK output frequency, for use when the chip is master over BCLK. 000=divide by 1 (BCLK=SYSCLK) 001=divide by 2 (BCLK=SYSCLK/2) 010=divide by 4 011=divide by 8 100=divide by 16 101=divide by 32 110=reserved 111=reserved |
| | 7:5 | MCLKDIV | 010 | 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 |
| | 8 | CLKSEL | 1 | Controls the source of the clock for all internal operation: 0=MCLK 1=PLL output |

Table 55 Clock Control

The CLKSEL bit selects the internal source of the Master clock from the PLL (CLKSEL=1) or from MCLK (CLKSEL=0). When the internal clock is switched from one source to another using the CLKSEL bit, the clock originally selected must generate at least one falling edge after CLKSEL has changed for the switching of clocks to be successful.

EXAMPLE:

If the PLL is the current source of the internal clock (CLKSEL=1) and it is required to switch to the MCLK, change CLKSEL to select MCLK (CLKSEL=0) and then disable PLL (PLLEN=0).

AUDIO SAMPLE RATES

The WM8976 sample rates for the ADC and the DACs are set using the SR register bits. The cutoffs 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.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|--------------------------|-----|-------|---------|--|
| R7 Additional Control | 3:1 | SR | 000 | 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 |

Table 56 Sample Rate Control

MASTER CLOCK AND PHASE LOCKED LOOP (PLL)

The WM8976 has an on-chip phase-locked loop (PLL) circuit that can be used to:

Generate master clocks for the WM8976 audio functions from another external clock, e.g. in telecoms applications.

Generate and output (on pin CSB/GPIO1 and/or GPIO4) a clock for another part of the system that is derived from an existing audio master clock.

Figure 40 shows the PLL and internal clocking arrangement on the WM8976.

The PLL can be enabled or disabled by the PLEN 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.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|--------------------------|-----|-------|---------|-------------------------------------|
| R1 Power management 1 | 5 | PLEN | 0 | PLL enable 0=PLL off 1=PLL on |

Table 57 PLEN Control Bit

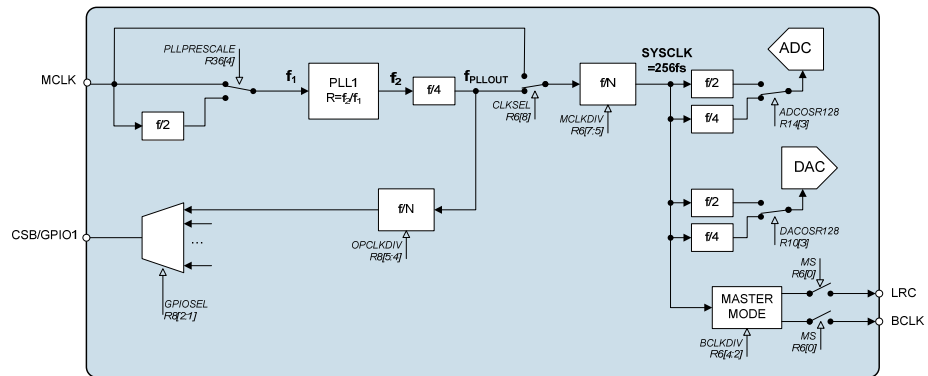


Figure 40 PLL and Clock Select Circuit

The PLL frequency ratio $R = f_2/f_1$ (see Figure 40) can be set using the register bits PLLK and PLLN:

$$PLL\ N = \text{int } R$$

$$PLL\ K = \text{int } (2^{24} (R - PLLN))$$

Note: The PLL is designed to operate with best performance (shortest lock time and optimum stability) when f_2 is between 90 and 100MHz and PLL_N is 8. However, acceptable PLL_N values lie in the range $5 \leq PLL_N \leq 13$. Do not use values outwith this range and it is recommended that the chosen value of PLL_N is as close to 8 as possible for optimum performance.

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 $f_2 = 4 \times 2 \times 12.288\text{MHz} = 98.304\text{MHz}$.

$$R = 98.304 / 12 = 8.192$$

$$PLL\ N = \text{int } R = 8$$

$$k = \text{int } (2^{24} \times (8.192 - 8)) = 3221225 = 3126\text{E}9\text{h}$$

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|----------------------|-----|--------------|---------|---|
| R36 PLL N value | 4 | PLLPRESCALE | 0 | 0 = MCLK input not divided (default) 1 = Divide MCLK by 2 before input to PLL |
| | 3:0 | PLLN | 1000 | Integer (N) part of PLL input/output frequency ratio. Use values greater than 5 and less than 13. |
| R37 PLL K value 1 | 5:0 | PLLK [23:18] | 0Ch | Fractional (K) part of PLL1 input/output frequency ratio (treat as one 24-digit binary number). |
| R38 PLL K Value 2 | 8:0 | PLLK [17:9] | 093h | |
| R39 PLL K Value 3 | 8:0 | PLLK [8:0] | 0E9h | |

Table 58 PLL Frequency Ratio Control

The PLL performs best when f_2 is around 90MHz. Its stability peaks at N=8. Some example settings are shown in Table 59.

| MCLK (MHz) (F1) | DESIRED OUTPUT (MHz) | F2 (MHz) | PRESCALE DIVIDE | POSTSCALE DIVIDE | R | N (Hex) | K (Hex) |
|-----------------------|----------------------------|-------------|--------------------|---------------------|----------|------------|------------|
| 12 | 11.29 | 90.3168 | 1 | 2 | 7.5264 | 7 | 86C226 |
| 12 | 12.288 | 98.304 | 1 | 2 | 8.192 | 8 | 3126E8 |
| 13 | 11.29 | 90.3168 | 1 | 2 | 6.947446 | 6 | F28BD4 |
| 13 | 12.288 | 98.304 | 1 | 2 | 7.561846 | 7 | 8FD525 |
| 14.4 | 11.29 | 90.3168 | 1 | 2 | 6.272 | 6 | 45A1CA |
| 14.4 | 12.288 | 98.304 | 1 | 2 | 6.826667 | 6 | D3A06E |
| 19.2 | 11.29 | 90.3168 | 2 | 2 | 9.408 | 9 | 6872AF |
| 19.2 | 12.288 | 98.304 | 2 | 2 | 10.24 | A | 3D70A3 |
| 19.68 | 11.29 | 90.3168 | 2 | 2 | 9.178537 | 9 | 2DB492 |
| 19.68 | 12.288 | 98.304 | 2 | 2 | 9.990243 | 9 | FD809F |
| 19.8 | 11.29 | 90.3168 | 2 | 2 | 9.122909 | 9 | 1F76F7 |
| 19.8 | 12.288 | 98.304 | 2 | 2 | 9.929697 | 9 | EE009E |
| 24 | 11.29 | 90.3168 | 2 | 2 | 7.5264 | 7 | 86C226 |
| 24 | 12.288 | 98.304 | 2 | 2 | 8.192 | 8 | 3126E8 |
| 26 | 11.29 | 90.3168 | 2 | 2 | 6.947446 | 6 | F28BD4 |
| 26 | 12.288 | 98.304 | 2 | 2 | 7.561846 | 7 | 8FD525 |
| 27 | 11.29 | 90.3168 | 2 | 2 | 6.690133 | 6 | BOAC93 |
| 27 | 12.288 | 98.304 | 2 | 2 | 7.281778 | 7 | 482296 |

Table 59 PLL Frequency Examples

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 WM8976 supports A-law and μ -law and companding and linear mode 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.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|--------------------------|-----|----------|---------|---|
| R5 Companding Control | 0 | LOOPBACK | 0 | Digital loopback function 0=No loopback 1=Loopback enabled, ADC data output is fed directly into left DAC data input. |
| | 2:1 | ADC_COMP | 0 | ADC companding 00=off (linear mode) 01=reserved 10= μ -law 11=A-law |
| | 4:3 | DAC_COMP | 0 | DAC companding 00=off (linear mode) 01=reserved 10= μ -law 11=A-law |
| | 5 | WL8 | 0 | 0=off 1=device operates in 8-bit mode |

Table 60 Companding Control

Companding involves using a piecewise linear approximation of the following equations (as set out by ITU-T G.711 standard) for data compression:

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

$$F(x) = \ln(1 + \mu|x|) / \ln(1 + \mu) \quad -1 \leq x \leq 1$$

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

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

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

The companded data is also inverted as recommended by the G.711 standard (all 8 bits are inverted for μ -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 (μ -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).

Setting the WL8 register bit allows the device to operate with 8-bit data. In this mode it is possible to use 8 BCLK's per LRC frame. When using DSP mode B, this allows 8-bit data words to be output consecutively every 8 BCLK's and can be used with 8-bit data words using the A-law and μ -law companding functions.

| BIT7 | BIT[6:4] | BIT[3:0] |
|------|----------|----------|
| SIGN | EXPONENT | MANTISSA |

Table 61 8-bit Companded Word Composition

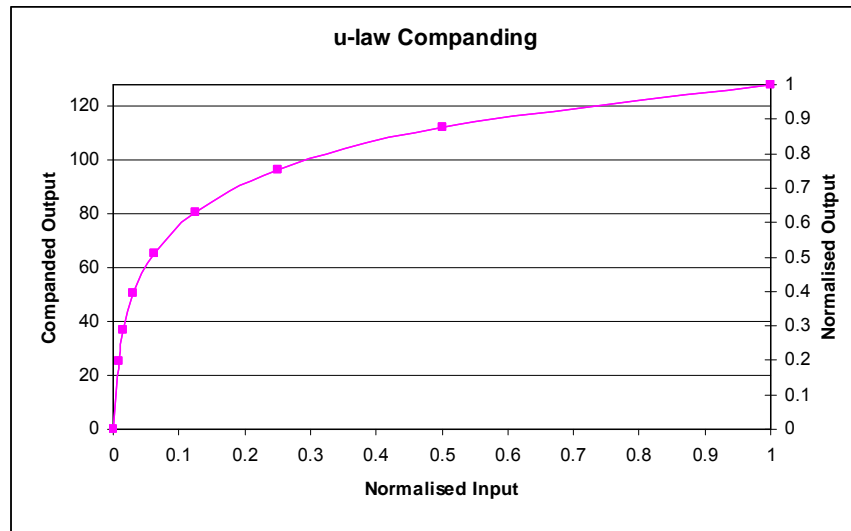


Figure 41 u-Law Comanding

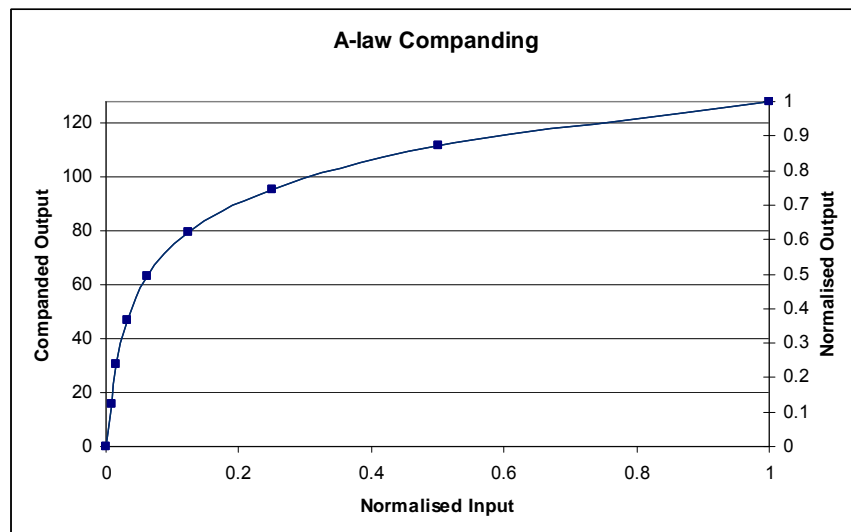


Figure 42 A-Law Comanding

GENERAL PURPOSE INPUT/OUTPUT

The WM8976 has two dual purpose input/output pins.

- CSB/GPIO1: CSB / GPIO pin
- L2/GPIO2: Line input / headphone detection input

The GPIO2 function is provided for use as a jack detection input.

The GPIO1 function is provided for use as a jack detection input or a general purpose output.

The default configuration for the CSB/GPIO1 pin is to be an input.

When setup as an input, the CSB/GPIO1 pin can either be used as CSB or for jack detection, depending on how the MODE pin is set.

Table 49 illustrates the functionality of the GPIO1 pin when used as a general purpose output.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|-----------------------|-----|----------|---------|--|
| R8 GPIO Control | 2:0 | GPIO1SEL | 000 | CSB/GPIO1 pin function select: 000= input (CSB/jack detection: depending on MODE setting) 001= reserved 010=Temp ok 011=Amute active 100=PLL clk o/p 101=PLL lock 110=logic 0 111=logic 1 |
| | 3 | GPIO1POL | 0 | GPIO1 Polarity invert 0=Non inverted 1=Inverted |
| | 5:4 | OPCLKDIV | 00 | PLL Output clock division ratio 00=divide by 1 01=divide by 2 10=divide by 3 11=divide by 4 |

Table 62 CSB/GPIO Control

Note: If MODE is set to 3 wire mode, CSB/GPIO1 shall be used as CSB input irrespective of the GPIO1SEL[2:] bits.

Note that SLOWCLKEN must be enabled when using the Jack Detect function.

For further details of the Jack detect operation see the OUTPUT SWITCHING section.

OUTPUT SWITCHING (JACK DETECT)

When the device is operated using a 2-wire interface the CSB/GPIO1 pin can be used as a switch control input to automatically disable one set of outputs and enable another the most common use for this functionality is as jack detect circuitry. The L2/GPIO2 pins can also be used for this purpose.

The GPIO pins have 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 $2^{21} \times \text{MCLK}$ and is enabled by the SLOWCLKEN bit.

Notes:

1. The SLOWCLKEN bit must be enabled for the jack detect circuitry to operate.
2. The GPIOPOL bit is not relevant for jack detection, it is the signal detected at the pin which is used

Switching on/off of the outputs is fully configurable by the user. Each output, OUT1, OUT2, OUT3 and OUT4 has 2 associated enables. OUT1_EN_0, OUT2_EN_0, OUT3_EN_0 and OUT4_EN_0 are the output enable signals which are used if the selected jack detection pin is at logic 0 (after de-bounce). OUT1_EN_1, OUT2_EN_1, OUT3_EN_1 and OUT4_EN_1 are the output enable signals which are used if the selected jack detection pin is at logic 1 (after de-bounce).

The jack detection enables operate as follows:

All OUT_EN signals have an AND function performed with their normal enable signals (in Table 50). When an output is normally enabled as per Table 50 the selected jack detection enable (controlled by selected jack detection pin polarity) is set 0; it will turn the output off. If the normal enable signal is already OFF (0), the jack detection signal will have no effect due to the AND function.

During jack detection if the user desires an output to be un-changed whether the jack is in or not, both the JD_EN settings i.e. JD_EN0 and JD_EN1, should be set to 0000.

The VMID_EN signal has an OR function performed with the normal VMID driver enable. If the VMID_EN signal is to have no effect to normal functionality when jack detection is enabled, it should be set to 0 for all JD_EN0 or JD_EN1 settings.

If jack detection is not enabled (JD_EN=0), the output enables default to all 1's, allowing the outputs to be controlled as normal via the normal output enables found in Table 51.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|--------------------|-----|---------|---------|--|
| R9 GPIO control | 4 | JD_SEL | 0 | Pin selected as jack detection input 0 = GPIO1 1 = GPIO2 |
| | 5 | | 0 | Reserved |
| | 6 | JD_EN | 0 | Jack Detection Enable 0 = disabled 1 = enabled |
| | 8:7 | JD_VMID | 00 | [7] VMID_EN_0 [8] VMID_EN_1 |
| R13 | 3:0 | JD_EN0 | 0000 | Output enables when selected jack detection input is logic 0. [0]= OUT1_EN_0 [1]= OUT2_EN_0 [2]= OUT3_EN_0 [3]= OUT4_EN_0 |
| | 7:4 | JD_EN1 | 0000 | Output enables when selected jack detection input is logic 1 0000-0011 = Reserved [4]= OUT1_EN_1 [5]= OUT2_EN_1 [6]= OUT3_EN_1 [7]= OUT4_EN_1 |

Table 63 Jack Detect Register Control Bits

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 64.

The WM8976 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.

| MODE | INTERFACE FORMAT |
|------|------------------|
| Low | 2 wire |
| High | 3 wire |

Table 64 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/GPIO1 pin latches in a complete control word consisting of the last 16 bits.

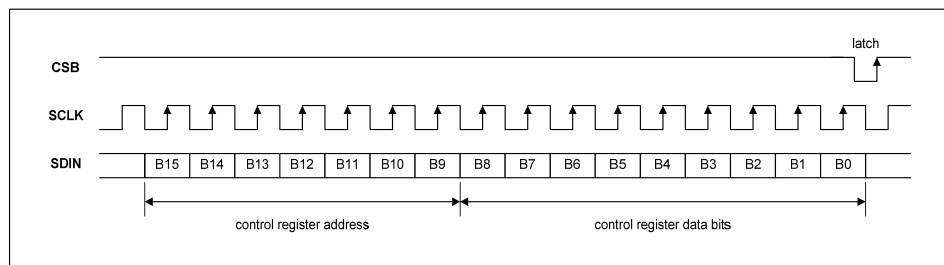


Figure 43 3-Wire Serial Control Interface

2-WIRE SERIAL CONTROL MODE

The WM8976 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 WM8976).

The WM8976 operates as a slave 2-wire 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 WM8976, then the WM8976 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 WM8976 returns to the idle condition and wait for a new start condition and valid address.

During a write, once the WM8976 has acknowledged a correct address, the controller sends the first byte of control data (B15 to B8, i.e. the WM8976 register address plus the first bit of register data). The WM8976 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 WM8976 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 WM8976 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.

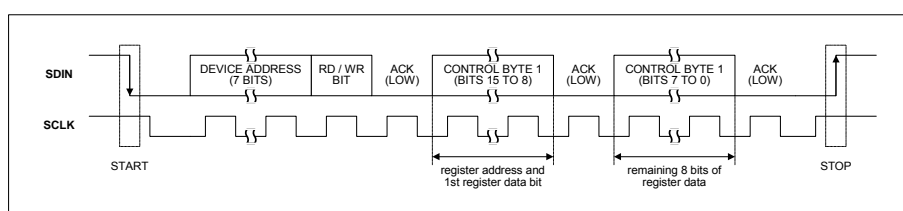


Figure 44 2-Wire Serial Control Interface

In 2-wire mode the WM8976 has a fixed device address, 0011010.

RESETTING THE CHIP

The WM8976 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 WM8976 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 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, 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.

DCVDD should be greater than or equal to 1.9V when using the PLL.

RECOMMENDED POWER UP/DOWN SEQUENCE

In order to minimise output pop and click noise, it is recommended that the WM8976 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. Mute all analogue outputs.
3. Set L/RMIXEN = 1 and DACENL/R = 1 in register R3.
4. Set BUFIOEN = 1 and VMIDSEL[1:0] to required value in register R1. Wait for the VMID supply to settle. *Refer notes 1 and 2.
5. Set BIASEN = 1 in register R1.
6. Set L/ROUT1EN = 1 in register R2.
7. Enable other mixers as required.
8. Enable other outputs as required.
9. Set remaining registers.

Power-up when using the output 1.5x boost stage:

1. Turn on external power supplies. Wait for supply voltage to settle.
2. Mute all analogue outputs.
3. Enable unused output chosen from L/ROUT2, OUT3 or OUT4. If unused output not available, chose one of these outputs not required at power up.
4. Set BUFDCOPEN = 1 and BUFIOEN = 1 in register R1.
5. Set SPKBOOST = 1 in register R49.
6. Set VMIDSEL[1:0] to required value in register R1. Wait for the VMID supply to settle. *Refer notes 1 and 2.
7. Set L/RMIXEN = 1 and DACENL/R = 1 in register R3.
8. Set BIASEN = 1 in register R1.
9. Set L/ROUT2EN = 1 in register R3. *Note 3.
10. Enable other mixers as required.
11. Enable other outputs as required.
12. Set remaining registers.

Power Down (all cases):

1. Mute all analogue outputs.
2. Disable Power Management Register 1. R1 = 0x00.
3. Disable Power Management Register 2. R2 = 0x00.
4. Disable Power Management Register 3. R3 = 0x00.
5. Remove 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 startup 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 startup time (VMIDSEL=10 for slowest startup, VMIDSEL=11 for fastest startup). Startup time is defined by the value of the VMIDSEL bits (the reference impedance) and the external decoupling capacitor on VMID.
3. Setting DACEN to off while operating in x1.5 boost mode will cause the VMID voltage to drop to AVDD/2 midrail level and cause an output pop.

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.

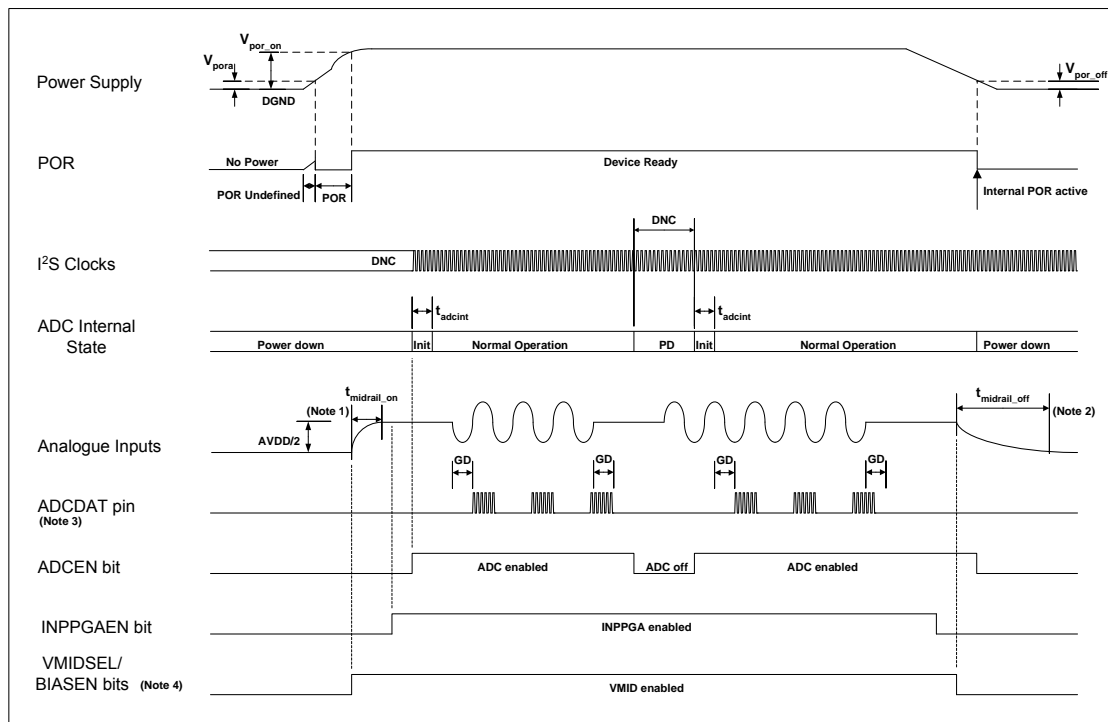


Figure 45 ADC Power Up and Down Sequence (not to scale)

| SYMBOL | MIN | TYPICAL | MAX | UNIT |
|--------------------|-----|---------|-----|------|
| $t_{midrail_on}$ | | 500 | | ms |
| $t_{midrail_off}$ | | >10 | | s |
| t_{adcint} | | 2/fs | | n/fs |
| ADC Group Delay | | 29/fs | | n/fs |

Table 65 Typical POR Operation (typical values, not tested)

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, $2/f_s$.

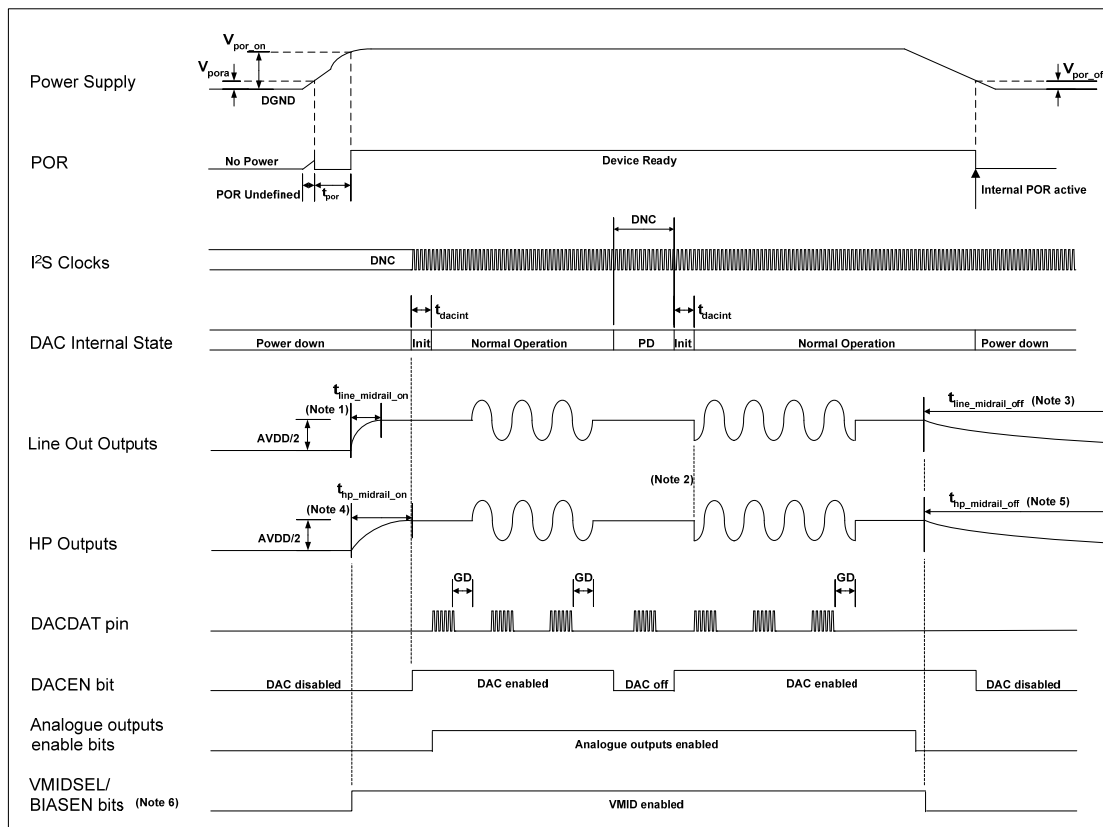


Figure 46 DAC Power Up and Down Sequence (not to scale)

| SYMBOL | MIN | TYPICAL | MAX | UNIT |
|--------------------------|-----|---------|-----|------|
| $t_{line_midrail_on}$ | | 500 | | ms |
| $t_{line_midrail_off}$ | | 1 | | s |
| $t_{hp_midrail_on}$ | | 500 | | ms |
| $t_{hp_midrail_off}$ | | 6 | | s |
| t_{dacint} | | 2/fs | | n/fs |
| DAC Group Delay | | 29/fs | | n/fs |

Table 66 Typical POR Operation (typical values, not tested)

Notes:

1. The lineout charge time, $t_{line_midrail_on}$, 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 μ F 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, $t_{line_midrail_off}$, 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 μ F output capacitor.
4. The headphone charge time, $t_{hp_midrail_on}$, 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 μ F VMID decoupling capacitor.
5. The headphone discharge time, $t_{hp_midrail_off}$, 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 μ F 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.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|--------------------|-----|-----------|---------|---|
| R10 DAC control | 3 | DACOSR128 | 0 | DAC oversample rate select 0 = 64x (lowest power) 1 = 128x (best SNR) |
| R14 ADC control | 3 | ADCOSR128 | 0 | ADC oversample rate select 0 = 64x (lowest power) 1 = 128x (best SNR) |

Table 67 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 startup time of the VMID circuit.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|--------------------------|-----|---------|---------|--|
| R1 Power management 1 | 1:0 | VMIDSEL | 00 | Reference string impedance to VMID pin determines startup time): 00=off (open circuit) 01=75k Ω 10=300k Ω 11=5k Ω (for fastest startup) |

Table 68 VMID Impedance Control

BIASEN

The analogue amplifiers will not operate unless BIASEN is enabled.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION |
|--------------------------|-----|--------|---------|--|
| R1 Power management 1 | 3 | BIASEN | 0 | Analogue amplifier bias control 0=disabled 1=enabled |

Table 69 Analogue Bias Control

REGISTER MAP

| ADDR B[15:9] | | REGISTER NAME | B8 | B7 | B6 | B5 | B4 | B3 | B2 | B1 | B0 | DEF'T VAL (HEX) | |
|--------------|-----|----------------------|----------------|------------|------------|---------------|--------------|------------|---------------|-------------|-------------|-----------------|-----|
| DEC | HEX | | | | | | | | | | | | |
| 0 | 00 | Software Reset | Software reset | | | | | | | | | | |
| 1 | 01 | Power manage't 1 | BUFDCOP EN | OUT4MIX EN | OUT3MIX EN | PLLEN | MICBEN | BIASEN | BUFIOEN | VMIDSEL | | 000 | |
| 2 | 02 | Power manage't 2 | ROUT1EN | LOUT1EN | SLEEP | 0 | BOOST ENL | 0 | INPPGA ENL | 0 | ADCENL | 000 | |
| 3 | 03 | Power manage't 3 | OUT4EN | OUT3EN | LOUT2EN | ROUT2EN | 0 | RMIXEN | LMIXEN | DACENR | DACENL | 000 | |
| 4 | 04 | Audio Interface | BCP | LRP | WL | | FMT | | DAC LRSWAP | ADC LRSWAP | DAC MONO | 050 | |
| 5 | 05 | Companing ctrl | 0 | 0 | 0 | WL8 | DAC_COMP | | ADC_COMP | | LOOPBACK | 000 | |
| 6 | 06 | Clock Gen ctrl | CLKSEL | MCLKDIV | | | BCLKDIV | | | 0 | MS | 140 | |
| 7 | 07 | Additional ctrl | 0 | 0 | 0 | 0 | 0 | SR | | | SLOWCLKEN | 000 | |
| 8 | 08 | GPIO Stuff | 0 | 0 | 0 | OPCLKDIV | | GPIO1POL | GPIO1SEL[2:0] | | | 000 | |
| 9 | 09 | Jack detect control | JD_VMID | | JD_EN | 0 | JD_SEL | 0 | 0 | 0 | 0 | 000 | |
| 10 | 0A | DAC Control | 0 | 0 | SOFT MUTE | 0 | 0 | DACOSR 128 | AMUTE | DACPOLR | DACPOLL | 000 | |
| 11 | 0B | Left DAC digital Vol | DACVU | DACVOLL | | | | | | | | OFF | |
| 12 | 0C | Right DAC dig'l Vol | DACVU | DACVOLR | | | | | | | | OFF | |
| 13 | 0D | Jack Detect Control | JD_EN1 | | | | JD_EN0 | | | | | | 000 |
| 14 | 0E | ADC Control | HPFEN | HPFAPP | HPFCUT | | | ADCOSR 128 | 0 | 0 | ADCLPOL | 100 | |
| 15 | 0F | ADC Digital Vol | ADCVU | ADCVOLL | | | | | | | | OFF | |
| 18 | 12 | EQ1 – low shelf | EQ3DMODE | 0 | EQ1C | | | EQ1G | | | | 12C | |
| 19 | 13 | EQ2 – peak 1 | EQ2BW | 0 | EQ2C | | | EQ2G | | | | 02C | |
| 20 | 14 | EQ3 – peak 2 | EQ3BW | 0 | EQ3C | | | EQ3G | | | | 02C | |
| 21 | 15 | EQ4 – peak 3 | EQ4BW | 0 | EQ4C | | | EQ4G | | | | 02C | |
| 22 | 16 | EQ5 – high shelf | 0 | 0 | EQ5C | | | EQ5G | | | | 02C | |
| 24 | 18 | DAC Limiter 1 | LIMEN | LIMDCY | | | | LIMATK | | | | 032 | |
| 25 | 19 | DAC Limiter 2 | 0 | 0 | LIMLVL | | | | LIMBOOST | | | | 000 |
| 27 | 1B | Notch Filter 1 | NFU | NFEN | NFA0[13:7] | | | | | | | 000 | |
| 28 | 1C | Notch Filter 2 | NFU | 0 | NFA0[6:0] | | | | | | | 000 | |
| 29 | 1D | Notch Filter 3 | NFU | 0 | NFA1[13:7] | | | | | | | 000 | |
| 30 | 1E | Notch Filter 4 | NFU | 0 | NFA1[6:0] | | | | | | | 000 | |
| 32 | 20 | ALC control 1 | ALCSEL | 0 | 0 | ALCMAXGAIN | | | ALCMINGAIN | | | 038 | |
| 33 | 21 | ALC control 2 | 0 | ALCHLD | | | | ALCLVL | | | | 00B | |
| 34 | 22 | ALC control 3 | ALCMODE | ALCDCY | | | | ALCATK | | | | 032 | |
| 35 | 23 | Noise Gate | 0 | 0 | 0 | 0 | 0 | NGEN | NGTH | | | 000 | |
| 36 | 24 | PLL N | 0 | 0 | 0 | 0 | PLLPRE SCALE | PLLN[3:0] | | | 008 | | |
| 37 | 25 | PLL K 1 | 0 | 0 | 0 | PLLK[23:18] | | | | | | 00C | |
| 38 | 26 | PLL K 2 | PLLK[17:9] | | | | | | | | | 093 | |
| 39 | 27 | PLL K 3 | PLLK[8:0] | | | | | | | | | 0E9 | |
| 41 | 29 | 3D control | 0 | 0 | 0 | 0 | 0 | DEPTH3D | | | | 000 | |
| 43 | 2B | Beep control | 0 | 0 | 0 | MUTER PGA2INV | INVROUT2 | BEEPVOL | | | BEEPEN | 000 | |
| 44 | 2C | Input ctrl | MBVSEL | 0 | 0 | 0 | 0 | 0 | L2_2 INPPGA | LIN2 INPPGA | LIP2 INPPGA | 033 | |
| 45 | 2D | INP PGA gain ctrl | INPPGA | INPPGAZC | INPPGA | INPPGAVOLL | | | | | | 010 | |

| ADDR B[15:9] | | REGISTER NAME | B8 | B7 | B6 | B5 | B4 | B3 | B2 | B1 | B0 | DEF'T VAL (HEX) |
|-----------------|-----|----------------------------|------------|---------|---------------|---------------|---------------|----------------|---------------|---------------|---------------|-----------------------|
| DEC | HEX | | | | | | | | | | | |
| | | | UPDATE | L | MUTEL | | | | | | | |
| 47 | 2F | ADC Boost ctrl | PGABOOSTL | 0 | L2_2BOOSTVOL | | | 0 | AUXL2BOOSTVOL | | | 100 |
| 49 | 31 | Output ctrl | 0 | 0 | DACL2 RMIX | DACR2 LMIX | OUT4 BOOST | OUT3 BOOST | SPK BOOST | TSDEN | VROI | 002 |
| 50 | 32 | Left mixer ctrl | AUXLMIXVOL | | | AUXL2LMIX | BYPLMIXVOL | | | BYPL2LMIX | DACL2LMIX | 001 |
| 51 | 33 | Right mixer ctrl | AUXRMIXVOL | | | AUXR2RMI X | 0 | | | 0 | DACR2RMIX | 001 |
| 52 | 34 | LOUT1 (HP) volume ctrl | HPVU | LOUT1ZC | LOUT1 MUTE | LOUT1VOL | | | | | | 039 |
| 53 | 35 | ROUT1 (HP) volume ctrl | HPVU | ROUT1ZC | ROUT1 MUTE | ROUT1VOL | | | | | | 039 |
| 54 | 36 | LOUT2 (SPK) volume ctrl | SPKVU | LOUT2ZC | LOUT2 MUTE | LOUT2VOL | | | | | | 039 |
| 55 | 37 | ROUT2 (SPK) volume ctrl | SPKVU | ROUT2ZC | ROUT2 MUTE | ROUT2VOL | | | | | | 039 |
| 56 | 38 | OUT3 mixer ctrl | 0 | 0 | OUT3 MUTE | 0 | 0 | OUT4_ 2OUT3 | BYPL2 OUT3 | LMIX2 OUT3 | LDAC2 OUT3 | 001 |
| 57 | 39 | OUT4 (MONO) mixer ctrl | 0 | 0 | OUT4 MUTE | HALFSIG | LMIX2 OUT4 | LDAC2 OUT4 | 0 | RMIX2 OUT4 | RDAC2 OUT4 | 001 |

Table 70 WM8976 Register Map

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.

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION | REFER TO |
|------------------|-------|-----------|---------|---|--|
| 0 (00h) | [8:0] | RESET | N/A | Software reset | Resetting the Chip |
| 1 (01h) | 8 | BUFDCOPEN | 0 | 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) | Analogue Outputs |
| | 7 | OUT4MIXEN | 0 | OUT4 mixer enable 0=disabled 1=enabled | Power Management |
| | 6 | OUT3MIXEN | 0 | OUT3 mixer enable 0=disabled 1=enabled | Power Management |
| | 5 | PLEN | 0 | PLL enable 0=PLL off 1=PLL on | Master Clock and Phase Locked Loop (PLL) |
| | 4 | MICBEN | 0 | Microphone Bias Enable 0 = OFF (high impedance output) 1 = ON | Input Signal Path |
| | 3 | BIASEN | 0 | Analogue amplifier bias control 0=disabled 1=enabled | Power Management |
| | 2 | BUFIOEN | 0 | Unused input/output tie off buffer enable 0=disabled 1=enabled | Power Management |
| | 1:0 | VMIDSEL | 00 | Reference string impedance to VMID pin 00=off (open circuit) 01=75kΩ 10=300kΩ 11=5kΩ | Power Management |
| 2 (02h) | 8 | ROUT1EN | 0 | ROUT1 output enable 0=disabled 1=enabled | Power Management |
| | 7 | LOUT1EN | 0 | LOUT1 output enable 0=disabled 1=enabled | Power Management |
| | 6 | SLEEP | 0 | 0 = normal device operation 1 = residual current reduced in device standby mode | Power Management |
| | 5 | | 0 | Reserved | |
| | 4 | BOOSTENL | 0 | Input BOOST enable 0 = Boost stage OFF 1 = Boost stage ON | Power Management |
| | 3 | | 0 | Reserved | |
| | 2 | INPPGAENL | 0 | Input PGA enable 0 = disabled 1 = enabled | Power Management |

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION | REFER TO |
|------------------|-----|---------|--|--|-------------------------------------|
| | 1 | | 0 | Reserved | |
| | 0 | ADCENL | 0 | Enable ADC: 0 = ADC disabled 1 = ADC enabled | Analogue to Digital Converter (ADC) |
| 3 (03h) | 8 | OUT4EN | 0 | OUT4 enable 0 = disabled 1 = enabled | Power Management |
| | 7 | OUT3EN | 0 | OUT3 enable 0 = disabled 1 = enabled | Power Management |
| | 6 | LOUT2EN | 0 | LOUT2 enable 0 = disabled 1 = enabled | Power Management |
| | 5 | ROUT2EN | 0 | ROUT2 enable 0 = disabled 1 = enabled | Power Management |
| | 3 | RMIXEN | 0 | Right output channel mixer enable: 0 = disabled 1 = enabled | Analogue Outputs |
| | 2 | LMIXEN | 0 | Left output channel mixer enable: 0 = disabled 1 = enabled | Analogue Outputs |
| | 1 | DACENR | 0 | Right channel DAC enable 0 = DAC disabled 1 = DAC enabled | Analogue Outputs |
| | 0 | DACENL | 0 | Left channel DAC enable 0 = DAC disabled 1 = DAC enabled | Analogue Outputs |
| 4 (04h) | 8 | BCP | 0 | BCLK polarity 0=normal 1=inverted | Digital Audio Interfaces |
| | 7 | LRP | 0 | right, left and i2s modes – LRCLK polarity 1 = invert LRCLK polarity 0 = normal LRCLK polarity | Digital Audio Interfaces |
| | | | | DSP Mode – mode A/B select 1 = MSB is available on 1 st BCLK rising edge after LRC rising edge (mode B) 0 = MSB is available on 2 nd BCLK rising edge after LRC rising edge (mode A) | |
| | 6:5 | WL | 10 | Word length 00=16 bits 01=20 bits 10=24 bits 11=32 bits | Digital Audio Interfaces |
| 4:3 | FMT | 10 | Audio interface Data Format Select: 00=Right Justified 01=Left Justified 10=i ² S format 11= DSP/PCM mode | Digital Audio Interfaces | |

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION | REFER TO |
|------------------|-----|-----------|---------|---|--------------------------|
| | 2 | DACLRSWAP | 0 | Controls whether DAC data appears in 'right' or 'left' phases of LRC clock: 0=DAC data appear in 'left' phase of LRC 1=DAC data appears in 'right' phase of LRC | Digital Audio Interfaces |
| | 1 | ADCLRSWAP | 0 | Controls whether ADC data appears in 'right' or 'left' phases of LRC clock: 0=ADC data appear in 'left' phase of LRC 1=ADC data appears in 'right' phase of LRC | Digital Audio Interfaces |
| | 0 | DACMONO | 0 | Selects between stereo and mono DAC operation: 0=Stereo device operation 1=Mono device operation. DAC data appears in 'left' phase of LRC | Digital Audio Interfaces |
| 5 (05h) | 8:6 | | 000 | Reserved | |
| | 5 | WL8 | 0 | Companding Control 8-bit mode 0=off 1=device operates in 8-bit mode | Digital Audio Interfaces |
| | 4:3 | DAC_COMP | 00 | DAC companding 00=off (linear mode) 01=reserved 10= μ -law 11=A-law | Digital Audio Interfaces |
| | 2:1 | ADC_COMP | 00 | ADC companding 00=off (linear mode) 01=reserved 10= μ -law 11=A-law | Digital Audio Interfaces |
| | 0 | LOOPBACK | 0 | Digital loopback function 0=No loopback 1=Loopback enabled, ADC data output is fed directly into DAC data input. | Digital Audio Interfaces |
| 6 (06h) | 8 | CLKSEL | 1 | Controls the source of the clock for all internal operation: 0=MCLK 1=PLL output | Digital Audio Interfaces |
| | 7:5 | MCLKDIV | 010 | 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 | Digital Audio Interfaces |

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION | REFER TO |
|------------------|-----|----------------|---------|--|-------------------------------------|
| | 4:2 | BCLKDIV | 000 | Configures the BCLK output frequency, for use when the chip is master over BCLK. 000=divide by 1 (BCLK=SYSCLK) 001=divide by 2 (BCLK=SYSCLK/2) 010=divide by 4 011=divide by 8 100=divide by 16 101=divide by 32 110=reserved 111=reserved | Digital Audio Interfaces |
| | 1 | | 0 | Reserved | |
| | 0 | MS | 0 | Sets the chip to be master over LRC and BCLK 0=BCLK and LRC clock are inputs 1=BCLK and LRC clock are outputs generated by the WM8976 (MASTER) | Digital Audio Interfaces |
| 7 (07h) | 8:4 | | 00000 | Reserved | |
| | 3:1 | SR | 000 | 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 | Audio Sample Rates |
| | 0 | SLOWCLKEN | 0 | Slow clock enable. Used for both the jack insert detect debounce circuit and the zero cross timeout. 0 = slow clock disabled 1 = slow clock enabled | Analogue Outputs |
| 8 (08h) | 8:6 | | 000 | Reserved | |
| | 5:4 | OPCLKDIV | 00 | PLL Output clock division ratio 00=divide by 1 01=divide by 2 10=divide by 3 11=divide by 4 | General Purpose Input/Output (GPIO) |
| | 3 | GPIO1POL | 0 | GPIO1 Polarity invert 0=Non inverted 1=Inverted | General Purpose Input/Output (GPIO) |
| | 2:0 | GPIO1SEL [2:0] | 000 | CSB/GPIO1 pin function select: 000= input (CSB/jack detection: depending on MODE setting) 001= reserved 010=Temp ok 011=Amute active 100=PLL clk o/p 101=PLL lock 110=logic 1 111=logic 0 | General Purpose Input/Output (GPIO) |
| 9 (09h) | 8:7 | JD_VMID | 00 | [7] VMID_EN_0 [8] VMID_EN_1 | Output Switching (Jack Detect) |

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION | REFER TO |
|------------------|-----|-----------|----------|--|-------------------------------------|
| | 6 | JD_EN | 0 | Jack Detection Enable 0=disabled 1=enabled | Output Switching (Jack Detect) |
| | 5 | | 0 | Reserved | |
| | 4 | JD_SEL | 0 | Pin selected as jack detection input 0 = GPIO1 1 = GPIO2 | Output Switching (Jack Detect) |
| | 3:0 | | 0 | Reserved | |
| 10 (0Ah) | 8:7 | | 00 | Reserved | |
| | 6 | SOFTMUTE | 0 | Softmute enable: 0=Disabled 1=Enabled | Output Signal Path |
| | 5:4 | | 00 | Reserved | |
| | 3 | DACOSR128 | 0 | DAC oversample rate select 0 = 64x (lowest power) 1 = 128x (best SNR) | Power Management |
| | 2 | AMUTE | 0 | Automute enable 0 = Amute disabled 1 = Amute enabled | Output Signal Path |
| | 1 | DACPOLR | 0 | Right DAC output polarity: 0 = non-inverted 1 = inverted (180 degrees phase shift) | Output Signal Path |
| | 0 | DACPOLL | 0 | Left DAC output polarity: 0 = non-inverted 1 = inverted (180 degrees phase shift) | Output Signal Path |
| 11 (0Bh) | 8 | DACVU | N/A | DAC left and DAC right volume do not update until a 1 is written to DACVU (in reg 11 or 12) | Digital to Analogue Converter (DAC) |
| | 7:0 | DACVOLL | 11111111 | Left DAC Digital Volume Control 0000 0000 = Digital Mute 0000 0001 = -127dB 0000 0010 = -126.5dB ... 0.5dB steps up to 1111 1111 = 0dB | Digital to Analogue Converter (DAC) |
| 12 (0Ch) | 8 | DACVU | N/A | DAC left and DAC right volume do not update until a 1 is written to DACVU (in reg 11 or 12) | Output Signal Path |
| | 7:0 | DACVOLR | 11111111 | Right DAC Digital Volume Control 0000 0000 = Digital Mute 0000 0001 = -127dB 0000 0010 = -126.5dB ... 0.5dB steps up to 1111 1111 = 0dB | Output Signal Path |
| 13 (0Dh) | 8 | | 0 | Reserved | |

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION | REFER TO |
|------------------|-----|---------------|----------|---|-------------------------------------|
| | 7:4 | JD_EN1 | 0000 | Output enabled when selected jack detection input is logic 1 [4]= OUT1_EN_1 [5]= OUT2_EN_1 [6]= OUT3_EN_1 [7]= OUT4_EN_1 | Output Switching (Jack Detect) |
| | 3:0 | JD_EN0 | 0000 | Output enabled when selected jack detection input is logic 0. [0]= OUT1_EN_0 [1]= OUT2_EN_0 [2]= OUT3_EN_0 [3]= OUT4_EN_0 | Output Switching (Jack Detect) |
| 14 (0Eh) | 8 | HPFEN | 1 | High Pass Filter Enable 0=disabled 1=enabled | Analogue to Digital Converter (ADC) |
| | 7 | HPFAPP | 0 | Select audio mode or application mode 0=Audio mode (1 st order, fc = ~3.7Hz) 1=Application mode (2 nd order, fc = HPFCUT) | Analogue to Digital Converter (ADC) |
| | 6:4 | HPFCUT | 000 | Application mode cut-off frequency See Table 15 for details. | Analogue to Digital Converter (ADC) |
| | 3 | ADCOSR 128 | 0 | ADC oversample rate select 0 = 64x (lowest power) 1 = 128x (best SNR) | Power Management |
| | 2:1 | | 00 | Reserved | |
| | 0 | ADCLPOL | 0 | ADC polarity adjust: 0=normal 1=inverted | Analogue to Digital Converter (ADC) |
| 15 (0Fh) | 8 | ADCVU | N/A | ADC volume does not update until a 1 is written to ADCVU | Analogue to Digital Converter (ADC) |
| | 7:0 | ADCVOLL | 11111111 | ADC Digital Volume Control 0000 0000 = Digital Mute 0000 0001 = -127dB 0000 0010 = -126.5dB ... 0.5dB steps up to 1111 1111 = 0dB | Analogue to Digital Converter (ADC) |
| 16 (10h) | 8:0 | | 11111111 | Reserved | |
| 18 (12h) | 8 | EQ3DMODE | 1 | 0 = Equaliser and 3D Enhancement applied to ADC path 1 = Equaliser and 3D Enhancement applied to DAC path | Output Signal Path |
| | 7 | | 0 | Reserved | |
| | 6:5 | EQ1C | | EQ Band 1 Cut-off Frequency: 00=80Hz 01=105Hz 10=135Hz 11=175Hz | Output Signal Path |
| | 4:0 | EQ1G | 01100 | EQ Band 1 Gain Control. See Table 37 for details. | Output Signal Path |

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION | REFER TO |
|------------------|-----|-------|---------|---|--------------------|
| 19 (13h) | 8 | EQ2BW | 0 | EQ Band 2 Bandwidth Control 0=narrow bandwidth 1=wide bandwidth | Output Signal Path |
| | 7 | | 0 | Reserved | |
| | 6:5 | EQ2C | 01 | EQ Band 2 Centre Frequency: 00=230Hz 01=300Hz 10=385Hz 11=500Hz | Output Signal Path |
| | 4:0 | EQ2G | 01100 | EQ Band 2 Gain Control. See Table 37 for details. | Output Signal Path |
| 20 (14h) | 8 | EQ3BW | 0 | EQ Band 3 Bandwidth Control 0=narrow bandwidth 1=wide bandwidth | Output Signal Path |
| | 7 | | 0 | Reserved | |
| | 6:5 | EQ3C | 01 | EQ Band 3 Centre Frequency: 00=650Hz 01=850Hz 10=1.1kHz 11=1.4kHz | Output Signal Path |
| | 4:0 | EQ3G | 01100 | EQ Band 3 Gain Control. See Table 37 for details. | Output Signal Path |
| 21 (15h) | 8 | EQ4BW | 0 | EQ Band 4 Bandwidth Control 0=narrow bandwidth 1=wide bandwidth | Output Signal Path |
| | 7 | | 0 | Reserved | |
| | 6:5 | EQ4C | 01 | EQ Band 4 Centre Frequency: 00=1.8kHz 01=2.4kHz 10=3.2kHz 11=4.1kHz | Output Signal Path |
| | 4:0 | EQ4G | 01100 | EQ Band 4 Gain Control. See Table 37 for details. | Output Signal Path |
| 22 (16h) | 8:7 | | 00 | Reserved | |
| | 6:5 | EQ5C | 01 | EQ Band 5 Cut-off Frequency: 00=5.3kHz 01=6.9kHz 10=9kHz 11=11.7kHz | Output Signal Path |
| | 4:0 | EQ5G | 01100 | EQ Band 5 Gain Control. See Table 37 for details. | Output Signal Path |
| 24 (18h) | 8 | LIMEN | 0 | Enable the DAC digital limiter: 0=disabled 1=enabled | Output Signal Path |

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION | REFER TO |
|------------------|-----|----------|---------|--|-------------------------------------|
| | 7:4 | LIMDCY | 0011 | 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 | Output Signal Path |
| | 3:0 | LIMATK | 0010 | 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 | Output Signal Path |
| 25 (19h) | 8:7 | | 00 | Reserved | |
| | 6:4 | LIMLVL | 000 | Programmable signal threshold level (determines level at which the DAC limiter starts to operate) 000=-1dB 001=-2dB 010=-3dB 011=-4dB 100=-5dB 101 to 111=-6dB | Output Signal Path |
| | 3:0 | LIMBOOST | 0000 | 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 | Output Signal Path |
| 27 (1Bh) | 8 | NFU | 0 | Notch filter update. The notch filter values used internally only update when one of the NFU bits is set high. | Analogue to Digital Converter (ADC) |

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION | REFER TO |
|------------------|-----|------------|---------|--|--|
| | 7 | NFEN | 0 | Notch filter enable: 0=Disabled 1=Enabled | Analogue to Digital Converter (ADC) |
| | 6:0 | NFA0[13:7] | 0000000 | Notch Filter a0 coefficient, bits [13:7] | Analogue to Digital Converter (ADC) |
| 28 (1Ch) | 8 | NFU | 0 | Notch filter update. The notch filter values used internally only update when one of the NFU bits is set high. | Analogue to Digital Converter (ADC) |
| | 7 | | 0 | Reserved | |
| | 6:0 | NFA0[6:0] | 0000000 | Notch Filter a0 coefficient, bits [6:0] | Analogue to Digital Converter (ADC) |
| 29 (1Dh) | 8 | NFU | 0 | Notch filter update. The notch filter values used internally only update when one of the NFU bits is set high. | Analogue to Digital Converter (ADC) |
| | 7 | | 0 | Reserved | |
| | 6:0 | NFA1[13:7] | 0000000 | Notch Filter a1 coefficient, bits [13:7] | Analogue to Digital Converter (ADC) |
| 30 (1Eh) | 8 | NFU | 0 | Notch filter update. The notch filter values used internally only update when one of the NFU bits is set high. | Analogue to Digital Converter (ADC) |
| | 7 | | 0 | Reserved | |
| | 6:0 | NFA1[6:0] | 0000000 | Notch Filter a1 coefficient, bits [6:0] | Analogue to Digital Converter (ADC) |
| 32 (20h) | 8 | ALCSEL | 0 | ALC function select: 0=ALC off 1=ALC on | Input Limiter/ Automatic Level Control (ALC) |
| | 7:6 | | 00 | Reserved | |
| | 5:3 | ALCMAXGAIN | 111 | Set Maximum Gain of PGA 111=+35.25dB 110=+29.25dB 101=+23.25dB 100=+17.25dB 011=+11.25dB 010=+5.25dB 001=-0.75dB 000=-6.75dB | Input Limiter/ Automatic Level Control (ALC) |

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION | REFER TO | | | |
|------------------|------------------------------------|-----------------|--|---|---|----------|--------------|--------------|
| | 2:0 | ALCMINGAIN | 000 | Set minimum gain of PGA 000=-12dB 001=-6dB 010=0dB 011=+6dB 100=+12dB 101=+18dB 110=+24dB 111=+30dB | Input Limiter/ Automatic Level Control (ALC) | | | |
| 33 (21h) | 8 | | 0 | Reserved | | | | |
| | 7:4 | ALCHLD | 0000 | ALC hold time before gain is increased. 0000 = 0ms 0001 = 2.67ms 0010 = 5.33ms ... (time doubles with every step) 1010 or higher = 1.36s | Input Limiter/ Automatic Level Control (ALC) | | | |
| | 3:0 | ALCLVL | 1011 | ALC target – sets signal level at ADC input 1111 : -1.5dBFS 1110 : -1.5dBFS 1101 : -3dBFS 1100 : -4.5management..... (-1.5dB steps) 0001 : -21dBFS 0000 : -22.5dBFS | Input Limiter/ Automatic Level Control (ALC) | | | |
| 34 (22h) | 8 | ALCMODE | 0 | Determines the ALC mode of operation: 0=ALC mode 1=Limiter mode | Input Limiter/ Automatic Level Control (ALC) | | | |
| | 7:4 | ALCDCY [3:0] | 0011 | Decay (gain ramp-up) time (ALCMODE ==0) | Input Limiter/ Automatic Level Control (ALC) | | | |
| | | | | | | Per step | Per 6dB | 90% of range |
| | | | | 0000 | | 410us | 3.28ms | 23.6ms |
| | | | | 0001 | | 820us | 6.6ms | 47.2ms |
| | | | | 0010 | | 1.64ms | 13.1ms | 94.5ms |
| | | | | ... (time doubles with every step) | | | | |
| | | | 1010 or higher | 420ms | | 3.36s | 24.2s | |
| | | | 0011 | Decay (gain ramp-up) time (ALCMODE ==1) | | | | |
| | | | | | | Per step | Per 6dB | 90% of range |
| 0000 | | | | 90.8us | | 726us | 5.23ms | |
| 0001 | 182us | 1.45ms | | 10.5ms | | | | |
| 0010 | 363us | 2.91ms | | 20.9ms | | | | |
| | ... (time doubles with every step) | | | | | | | |
| 1010 | 93ms | 744ms | 5.36s | | | | | |
| 3:0 | ALCATK | 0010 | ALC attack (gain ramp-down) time (ALCMODE == 0) | Input Limiter/ Automatic Level Control (ALC) | | | | |
| | | | | | Per step | Per 6dB | 90% of range | |
| | | | 0000 | | 104us | 832us | 6ms | |
| | | | 0001 | | 208us | 1.66ms | 12ms | |
| | | | 0010 | | 416us | 3.33ms | 24.1ms | |

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION | REFER TO | | | |
|------------------|-----|-----------------|---------------|--|----------|--------------|-------|---|
| | | | | ... (time doubles with every step) | | | | |
| | | | | 1010 or higher | 106ms | 852ms | 6.13s | |
| | | | 0010 | ALC attack (gain ramp-down) time (ALCMODE == 1) | | | | |
| | | | | Per step | Per 6dB | 90% of range | | |
| | | | 0000 | 22.7us | 182us | 1.31ms | | |
| | | | 0001 | 45.4us | 363us | 2.62ms | | |
| | | | 0010 | 90.8us | 726us | 5.23ms | | |
| | | | | ... (time doubles with every step) | | | | |
| | | | 1010 | 23.2ms | 186ms | 1.34s | | |
| 35 (23h) | 8:4 | | 00000 | Reserved | | | | |
| | 3 | NGEN | 0 | ALC Noise gate function enable 1 = enable 0 = disable | | | | Input Limiter/ Automatic Level Control (ALC) |
| | 2:0 | NGTH | 000 | ALC Noise gate threshold: 000=-39dB 001=-45dB 010=-51db ... (6dB steps) 111=-81dB | | | | Input Limiter/ Automatic Level Control (ALC) |
| 36 (24h) | 8:5 | | 0000 | Reserved | | | | |
| | 4 | PLL PRESCALE | 0 | 0 = MCLK input not divided (default) 1 = Divide MCLK by 2 before input to PLL | | | | Master Clock and Phase Locked Loop (PLL) |
| | 3:0 | PLLN[3:0] | 1000 | Integer (N) part of PLL input/output frequency ratio. Use values greater than 5 and less than 13. | | | | Master Clock and Phase Locked Loop (PLL) |
| 37 (25h) | 8:6 | | 000 | Reserved | | | | |
| | 5:0 | PLLK[23:18] | 01100 | Fractional (K) part of PLL1 input/output frequency ratio (treat as one 24-digit binary number). | | | | Master Clock and Phase Locked Loop (PLL) |
| 38 (26h) | 8:0 | PLLK[17:9] | 01001001 1 | Fractional (K) part of PLL1 input/output frequency ratio (treat as one 24-digit binary number). | | | | Master Clock and Phase Locked Loop (PLL) |
| 39 (27h) | 8:0 | PLLK[8:0] | 01110100 1 | Fractional (K) part of PLL1 input/output frequency ratio (treat as one 24-digit binary number). | | | | Master Clock and Phase Locked Loop (PLL) |
| 40 (28h) | 8:0 | | 00000000 0 | Reserved | | | | |
| 41 (29h) | 8:4 | | 00000 | Reserved | | | | |
| | 3:0 | DEPTH3D | 0000 | Stereo depth 0000: 0% (minimum 3D effect) 0001: 6.67% 1110: 93.3% 1111: 100% (maximum 3D effect) | | | | 3D Stereo Enhancement |
| 43 (2Bh) | 8:6 | | 000 | Reserved | | | | |

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION | REFER TO |
|------------------|-----|------------------|---------------|--|----------------------|
| | 5 | MUTERPGA 2INV | 0 | Mute input to INVROUT2 mixer | Analogue Outputs |
| | 4 | INVROUT2 | 0 | Mute input to INVROUT2 mixer | Analogue Outputs |
| | 3:1 | BEEPVOL | 000 | AUXR input to ROUT2 inverter gain 000 = -15dB ... 111 = +6dB | Analogue Outputs |
| | 0 | BEEPEN | 0 | 0 = mute AUXR beep input 1 = enable AUXR beep input | Analogue Outputs |
| 44 (2Ch) | 8 | MBVSEL | 0 | Microphone Bias Voltage Control 0 = 0.9 * AVDD 1 = 0.6 * AVDD | Input Signal Path |
| | 7:3 | | 00000 | Reserved | |
| | 2 | L2_2INP PGA | 0 | Connect L2 pin to input PGA positive terminal. 0=L2 not connected to input PGA 1=L2 connected to input PGA amplifier positive terminal (constant input impedance). | Input Signal Path |
| | 1 | LIN2INP PGA | 1 | Connect LIN pin to input PGA negative terminal. 0=LIN not connected to input PGA 1=LIN connected to input PGA amplifier negative terminal. | Input Signal Path |
| | 0 | LIP2INP PGA | 1 | Connect LIP pin to input PGA amplifier positive terminal. 0 = LIP not connected to input PGA 1 = input PGA amplifier positive terminal connected to LIP (constant input impedance) | Input Signal Path |
| 45 (2Dh) | 8 | INPPGA UPDATE | N/A | INPPGAVOLL and INPPGAVOLR volume do not update until a 1 is written to INPPGAUPDATE (in reg 45 or 46) | Input Signal Path |
| | 7 | INPPGAZCL | 0 | Input PGA zero cross enable: 0=Update gain when gain register changes 1=Update gain on 1 st zero cross after gain register write. | Input Signal Path |
| | 6 | INPPGA MUTEL | 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). | Input Signal Path |
| | 5:0 | INPPGA VOLL | 010000 | Input PGA volume 000000 = -12dB 000001 = -11.25db . 010000 = 0dB . 111111 = 35.25dB | Input Signal Path |
| 46 (2Eh) | 8:0 | | 00001000 0 | Reserved | |
| 47 (2Fh) | 8 | PGA BOOSTL | 1 | Boost enable for input PGA: 0 = PGA output has +0dB gain through input BOOST stage. 1 = PGA output has +20dB gain through input BOOST stage. | Input Signal Path |
| | 7 | | 0 | Reserved | |

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION | REFER TO |
|------------------|-----|-------------------|---------------|---|----------------------|
| | 6:4 | L2_2 BOOSTVOL | 000 | Controls the L2 pin 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 | Input Signal Path |
| | 3 | | 0 | Reserved | |
| | 2:0 | AUXL2 BOOSTVOL | 000 | Control recommendliary 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 | Input Signal Path |
| 48 (30h) | 8:0 | | 10000000 0 | Reserved | |
| 49 (31h) | 8:7 | | 00 | Reserved | |
| | 6 | DACL2RMIX | 0 | Left DAC output to right output mixer 0 = not selected 1 = selected | Analogue Outputs |
| | 5 | DACR2LMIX | 0 | Right DAC output to left output mixer 0 = not selected 1 = selected | Analogue Outputs |
| | 4 | OUT4 BOOST | 0 | 0 = OUT4 output gain = -1; DC = AVDD / 2 1 = OUT4 output gain = +1.5 DC = 1.5 x AVDD / 2 | Analogue Outputs |
| | 3 | OUT3 BOOST | 0 | 0 = OUT3 output gain = -1; DC = AVDD / 2 1 = OUT3 output gain = +1.5 DC = 1.5 x AVDD / 2 | Analogue Outputs |
| | 2 | SPKBOOST | 0 | 0 = speaker gain = -1; DC = AVDD / 2 1 = speaker gain = +1.5; DC = 1.5 x AVDD / 2 | Analogue Outputs |
| | 1 | TSDEN | 1 | Thermal Shutdown Enable 0 : thermal shutdown disabled 1 : thermal shutdown enabled | Analogue Outputs |
| | 0 | VROI | 0 | VREF (AVDD/2 or 1.5xAVDD/2) to analogue output resistance 0: approx 1kΩ 1: approx 30 kΩ | Analogue Outputs |
| 50 (32h) | 8:6 | AUXLMIX VOL | 000 | Aux left channel input to left mixer volume control: 000 = -15dB 001 = -12dB ... 101 = 0dB 110 = +3dB 111 = +6dB | Analogue Outputs |
| | 5 | AUXL2L MIX | 0 | Left Auxiliary input to left channel output mixer: 0 = not selected 1 = selected | Analogue Outputs |

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION | REFER TO |
|------------------|-----|----------------|---------|---|---------------------|
| | 4:2 | BYPLMIX VOL | 000 | Bypass volume control to left output channel mixer: 000 = -15dB 001 = -12dB ... 101 = 0dB 110 = +3dB 111 = +6dB | Analogue Outputs |
| | 1 | BYPL2L MIX | 0 | Bypass path (from the input boost output) to left output mixer 0 = not selected 1 = selected | Analogue Outputs |
| | 0 | DACL2L MIX | 1 | Left DAC output to left output mixer 0 = not selected 1 = selected | Analogue Outputs |
| 51 (33h) | 8:6 | AUXRMIX VOL | 000 | Aux right channel input to right mixer volume control: 000 = -15dB 001 = -12dB ... 101 = 0dB 110 = +3dB 111 = +6dB | Analogue Outputs |
| | 5 | AUXR2R MIX | 0 | Right Auxiliary input to right channel output mixer: 0 = not selected 1 = selected | Analogue Outputs |
| | 4:1 | | 0000 | Reserved | |
| | 0 | DACR2R MIX | 1 | Right DAC output to right output mixer 0 = not selected 1 = selected | Analogue Outputs |
| 52 (34h) | 8 | HPVU | N/A | LOUT1 and ROUT1 volumes do not update until a 1 is written to HPVU (in reg 52 or 53) | Analogue Outputs |
| | 7 | LOUT1ZC | 0 | Headphone volume zero cross enable: 1 = Change gain on zero cross only 0 = Change gain immediately | Analogue Outputs |
| | 6 | LOUT1 MUTE | 0 | Left headphone output mute: 0 = Normal operation 1 = Mute | Analogue Outputs |
| | 5:0 | LOUT1VOL | 111001 | Left headphone output volume: 000000 = -57dB ... 111001 = 0dB ... 111111 = +6dB | Analogue Outputs |
| 53 (35h) | 8 | HPVU | N/A | LOUT1 and ROUT1 volumes do not update until a 1 is written to HPVU (in reg 52 or 53) | Analogue Outputs |
| | 7 | ROUT1ZC | 0 | Headphone volume zero cross enable: 1 = Change gain on zero cross only 0 = Change gain immediately | Analogue Outputs |
| | 6 | ROUT1 MUTE | 0 | Right headphone output mute: 0 = Normal operation 1 = Mute | Analogue Outputs |

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION | REFER TO |
|------------------|-----|------------|---------|---|------------------|
| | 5:0 | ROUT1VOL | 111001 | Right headphone output volume: 000000 = -57dB ... 111001 = 0dB ... 111111 = +6dB | Analogue Outputs |
| 54 (36h) | 8 | SPKVU | N/A | LOUT2 and ROUT2 volumes do not update until a 1 is written to SPKVU (in reg 54 or 55) | Analogue Outputs |
| | 7 | LOUT2ZC | 0 | Speaker volume zero cross enable: 1 = Change gain on zero cross only 0 = Change gain immediately | Analogue Outputs |
| | 6 | LOUT2 MUTE | 0 | Left speaker output mute: 0 = Normal operation 1 = Mute | Analogue Outputs |
| | 5:0 | LOUT2VOL | 111001 | Left speaker output volume: 000000 = -57dB ... 111001 = 0dB ... 111111 = +6dB | Analogue Outputs |
| 55 (37h) | 8 | SPKVU | N/A | LOUT2 and ROUT2 volumes do not update until a 1 is written to SPKVU (in reg 54 or 55) | Analogue Outputs |
| | 7 | ROUT2ZC | 0 | Speaker volume zero cross enable: 1 = Change gain on zero cross only 0 = Change gain immediately | Analogue Outputs |
| | 6 | ROUT2 MUTE | 0 | Right speaker output mute: 0 = Normal operation 1 = Mute | Analogue Outputs |
| | 5:0 | ROUT2VOL | 111001 | Right speaker output volume: 000000 = -57dB ... 111001 = 0dB ... 111111 = +6dB | Analogue Outputs |
| 56 (38h) | 8:7 | | 00 | Reserved | |
| | 6 | OUT3MUTE | 0 | 0 = Output stage outputs OUT3 mixer 1 = Output stage muted – drives out VMID. Can be used as VMID buffer in this mode. | Analogue Outputs |
| | 5:4 | | 00 | Reserved | |
| | 3 | OUT4_2OUT3 | 0 | OUT4 mixer output to OUT3 0 = disabled 1 = enabled | Analogue Outputs |
| | 2 | BYPL2OUT3 | 0 | ADC input to OUT3 0 = disabled 1 = enabled | Analogue Outputs |
| | 1 | LMIX2OUT3 | 0 | Left DAC mixer to OUT3 0 = disabled 1 = enabled | Analogue Outputs |
| | 0 | LDAC2OUT3 | 1 | Left DAC output to OUT3 0 = disabled 1 = enabled | Analogue Outputs |
| 57 (39h) | 8:7 | | 00 | Reserved | |

| REGISTER ADDRESS | BIT | LABEL | DEFAULT | DESCRIPTION | REFER TO |
|------------------|-----|-----------|---------|---|------------------|
| | 6 | OUT4MUTE | 0 | 0 = Output stage outputs OUT4 mixer 1 = Output stage muted – drives out VMID. Can be used as VMID buffer in this mode. | Analogue Outputs |
| | 5 | HALFSIG | 0 | 0=OUT4 normal output 1=OUT4 attenuated by 6dB | Analogue Outputs |
| | 4 | LMIX2OUT4 | 0 | Left DAC mixer to OUT4 0 = disabled 1= enabled | Analogue Outputs |
| | 3 | LDAC2OUT4 | 0 | Left DAC to OUT4 0 = disabled 1= enabled | Analogue Outputs |
| | 2 | | 0 | Reserved | |
| | 1 | RMIX2OUT4 | 0 | Right DAC mixer to OUT4 0 = disabled 1= enabled | Analogue Outputs |
| | 0 | RDAC2OUT4 | 1 | Right DAC output to OUT4 0 = disabled 1= enabled | Analogue Outputs |

DIGITAL FILTER CHARACTERISTICS

| PARAMETER | TEST CONDITIONS | MIN | TYP | MAX | UNIT |
|-----------------------------------|-----------------|---------|-------|-----------|------|
| ADC Filter | | | | | |
| Passband | +/- 0.025dB | 0 | | 0.454fs | |
| | -6dB | | 0.5fs | | |
| Passband Ripple | | | | +/- 0.025 | dB |
| Stopband | | 0.546fs | | | |
| Stopband Attenuation | f > 0.546fs | -60 | | | dB |
| Group Delay | | | 21/fs | | |
| 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 | | | | +/-0.035 | dB |
| Stopband | | 0.546fs | | | |
| Stopband Attenuation | f > 0.546fs | -55 | | | dB |
| Group Delay | | | 29/fs | | |

Table 71 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

DAC FILTER RESPONSES

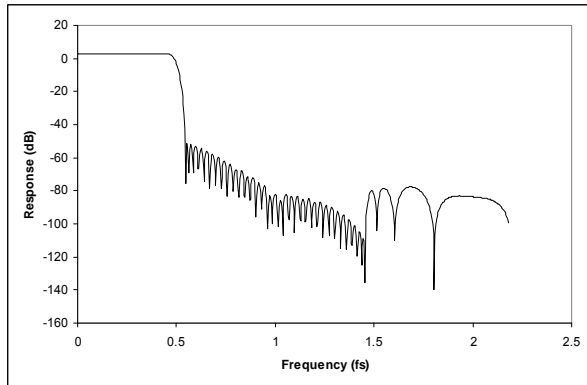


Figure 47 DAC Digital Filter Frequency Response (128xOSR)

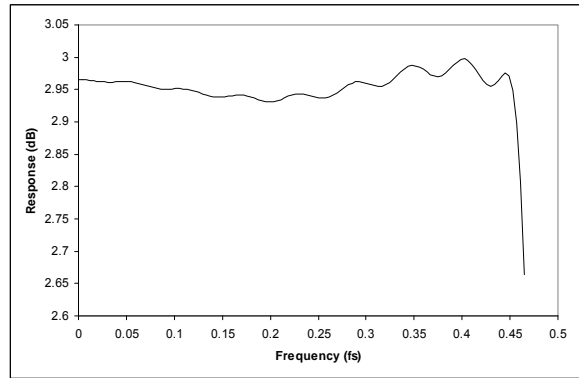


Figure 48 DAC Digital Filter Ripple (128xOSR)

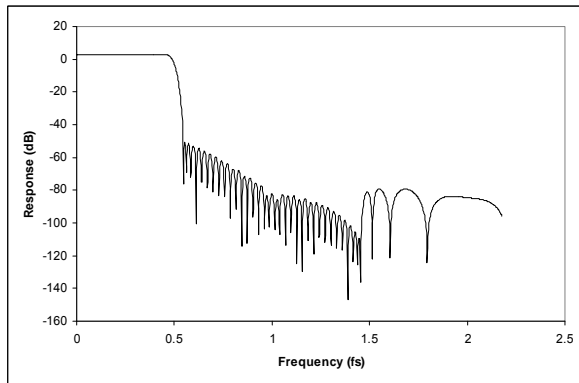


Figure 49 DAC Digital Filter Frequency Response (64xOSR)

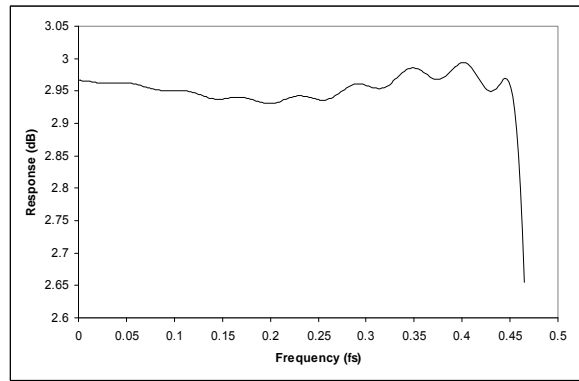


Figure 50 DAC Digital Filter Ripple (64xOSR)

ADC FILTER RESPONSES

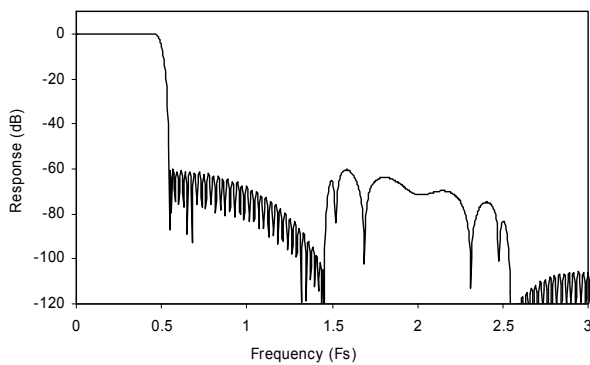


Figure 51 ADC Digital Filter Frequency Response

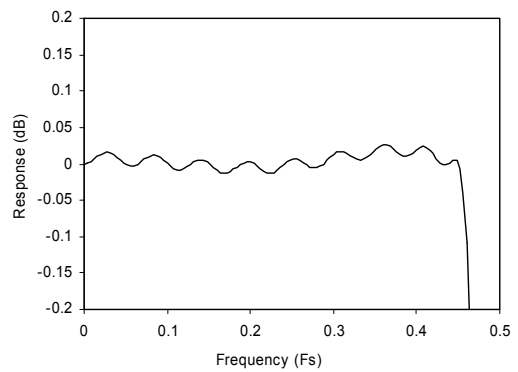


Figure 52 ADC Digital Filter Ripple

HIGHPASS FILTER

The WM8976 has a selectable digital highpass 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.

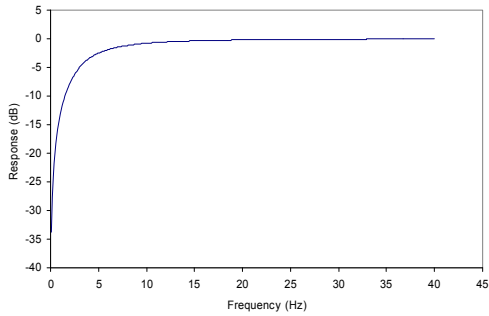


Figure 53 ADC Highpass Filter Response, HPFAPP=0

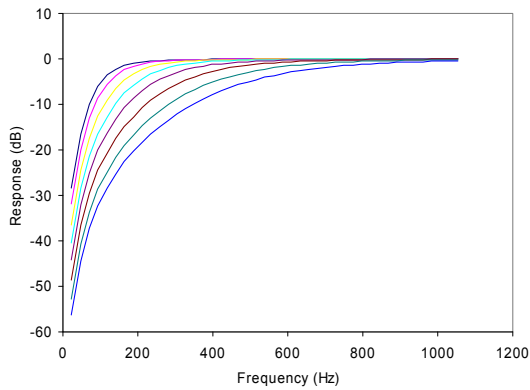


Figure 54 ADC Highpass Filter Responses (48kHz), HPFAPP=1, all cut-off settings shown.

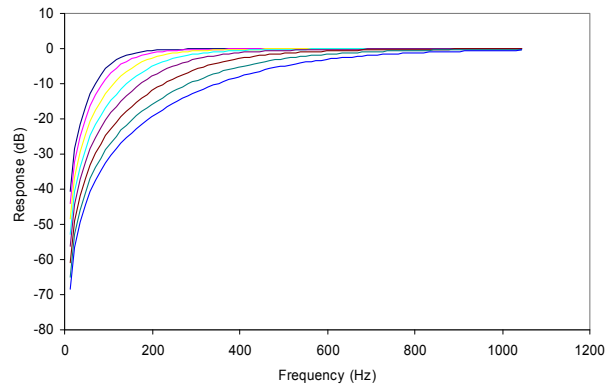


Figure 55 ADC Highpass Filter Responses (24kHz), HPFAPP=1, all cut-off settings shown.

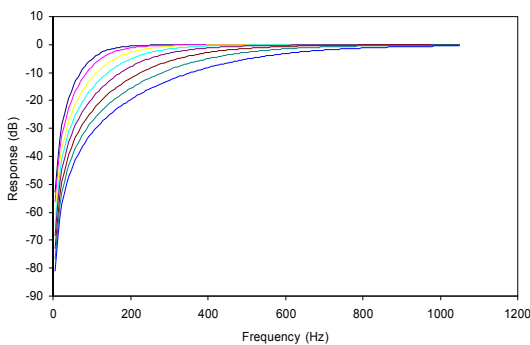


Figure 56 ADC Highpass Filter Responses (12kHz), HPFAPP=1, all cut-off settings shown.

5-BAND EQUALISER

The WM8976 has a 5-band equaliser which can be applied to either the ADC path or the DAC path. The plots from Figure 57 to Figure 70 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 $\pm 12\text{dB}$, and secondly a sweep of the gain from -12dB to $+12\text{dB}$ for the lowest cut-off/centre frequency of each filter.

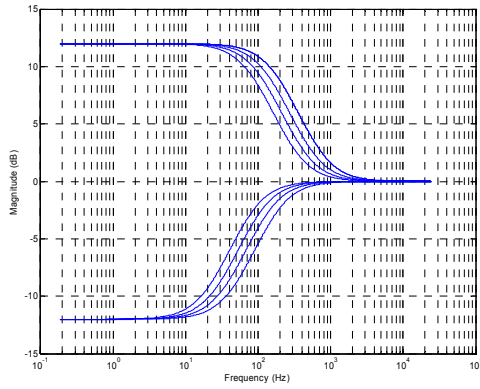


Figure 57 EQ Band 1 Low Frequency Shelf Filter Cut-offs

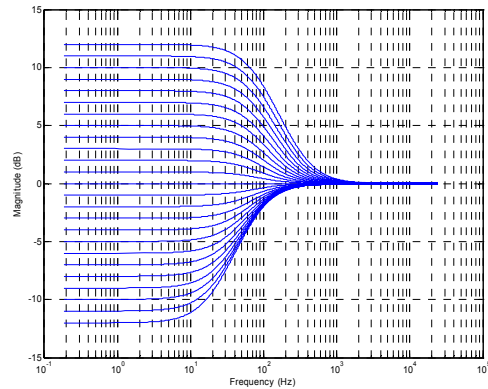


Figure 58 EQ Band 1 Gains for Lowest Cut-off Frequency

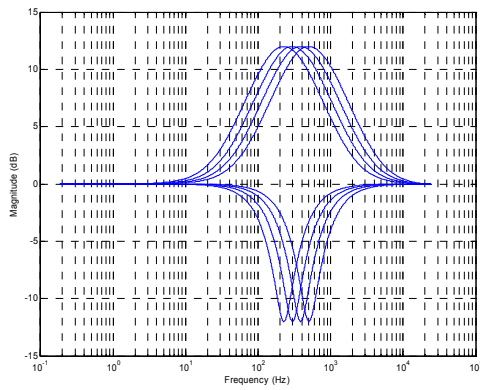


Figure 59 EQ Band 2 – Peak Filter Centre Frequencies, EQ2BW=0

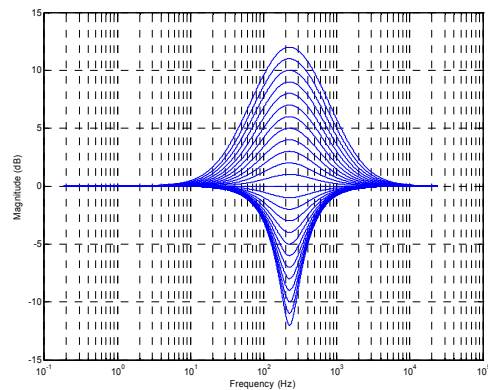


Figure 60 EQ Band 2 – Peak Filter Gains for Lowest Cut-off Frequency, EQ2BW=0

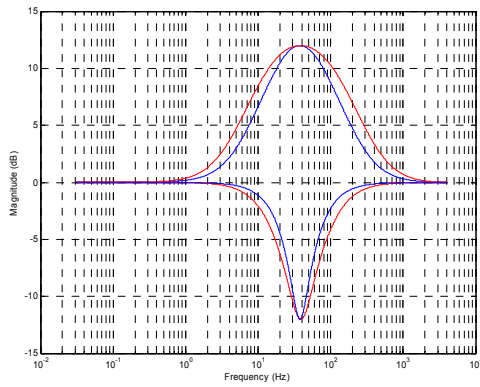


Figure 61 EQ Band 2 – EQ2BW=0, EQ2BW=1

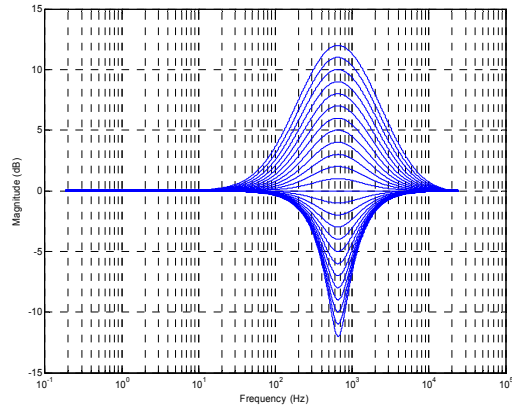
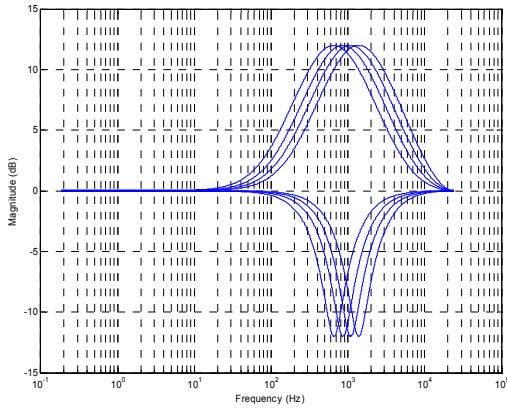


Figure 62 EQ Band 3 – Peak Filter Centre Frequencies, EQ3E Figure 63 EQ Band 3 – Peak Filter Gains for Lowest Cut-off Frequency, EQ3BW=0

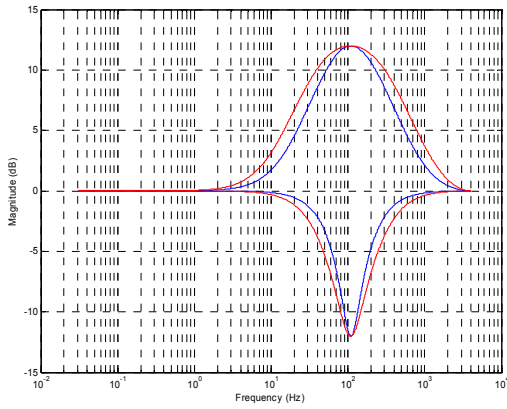


Figure 64 EQ Band 3 – EQ3BW=0, EQ3BW=1

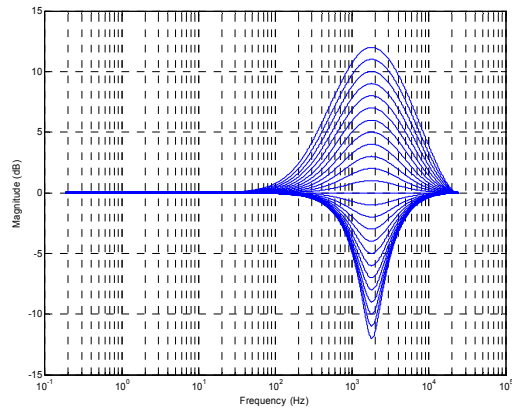
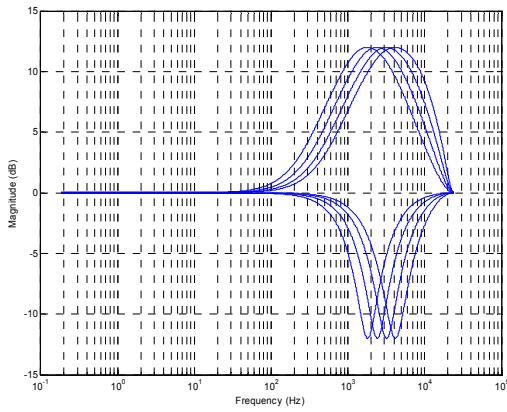


Figure 65 EQ Band 4 – Peak Filter Centre Frequencies, EQ3E Figure 66 EQ Band 4 – Peak Filter Gains for Lowest Cut-off Frequency, EQ4BW=0

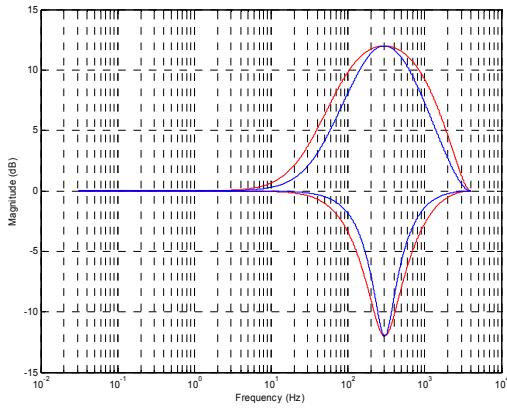


Figure 67 EQ Band 4 – EQ3BW=0, EQ3BW=1

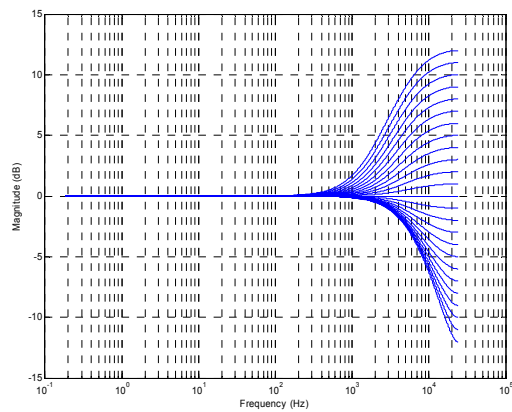
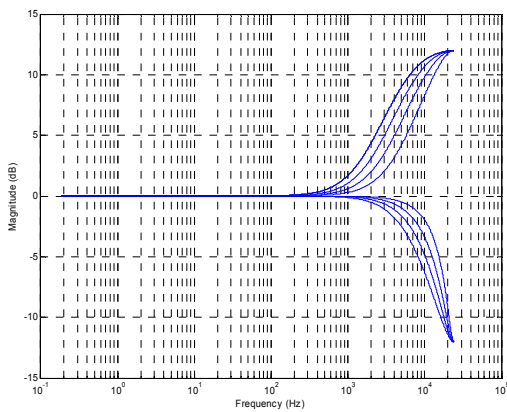


Figure 68 EQ Band 5 High Frequency Shelf Filter Cut-offs Figure 69 EQ Band 5 Gains for Lowest Cut-off Frequency

Figure 70 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 $\pm 12\text{dB}$ gain. The red traces show the cumulative effect of all bands with $+12\text{dB}$ gain and all bands -12dB gain, with $\text{EqxBW}=0$ for the peak filters.

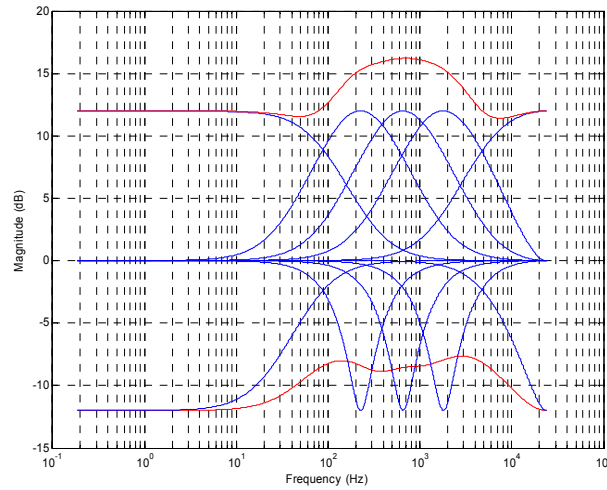


Figure 70 Cumulative Frequency Boost/Cut

APPLICATION INFORMATION

RECOMMENDED EXTERNAL COMPONENTS

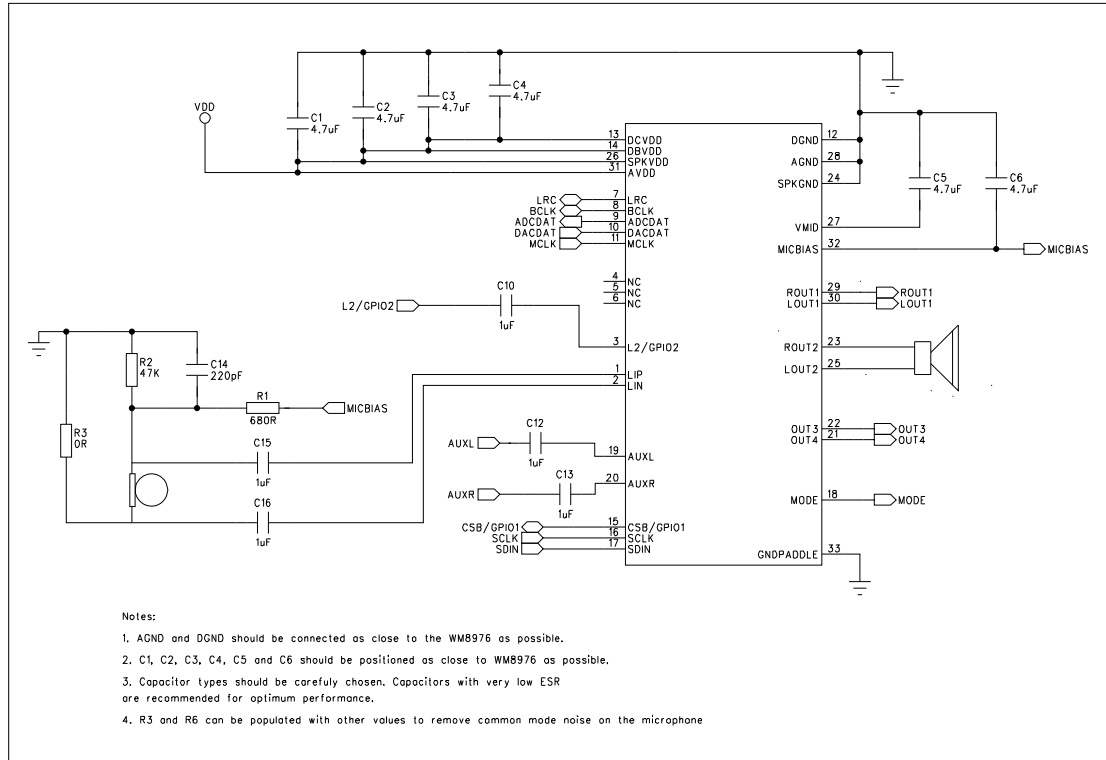
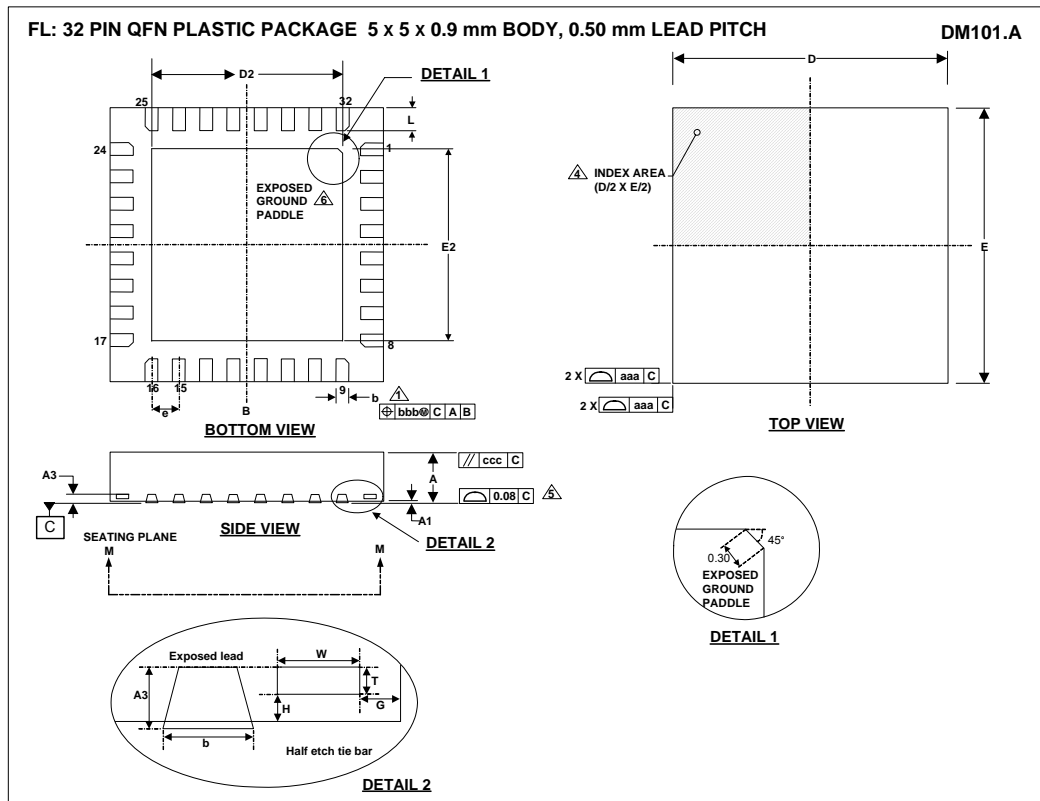


Figure 71 Recommended External Component Diagram

PACKAGE DIAGRAM



| Symbols | Dimensions (mm) | | | |
|--|----------------------------------|-----------|------|------|
| | MIN | NOM | MAX | NOTE |
| A | 0.80 | 0.90 | 1.00 | |
| A1 | 0 | 0.02 | 0.05 | |
| A3 | | 0.203 REF | | |
| b | 0.18 | 0.25 | 0.30 | 1 |
| D | | 5.00 BSC | | |
| D2 | 3.30 | 3.45 | 3.60 | 2 |
| E | | 5.00 BSC | | |
| E2 | 3.30 | 3.45 | 3.60 | 2 |
| e | | 0.50 BSC | | |
| G | | 0.20 | | |
| H | | 0.1 | | |
| L | 0.30 | 0.40 | 0.50 | |
| T | | 0.103 | | |
| W | | 0.15 | | |
| Tolerances of Form and Position | | | | |
| aaa | | 0.15 | | |
| bbb | | 0.10 | | |
| ccc | | 0.10 | | |
| REF: | JEDEC, MO-220, VARIATION VHHD-5. | | | |

- NOTES:
- DIMENSION b APPLIES TO METALLIZED TERMINAL AND IS MEASURED BETWEEN 0.15 mm AND 0.30 mm FROM TERMINAL TIP.
 - FALLS WITHIN JEDEC, MO-220, VARIATION VHHD-5.
 - ALL DIMENSIONS ARE IN MILLIMETRES.
 - THE TERMINAL #1 IDENTIFIER AND TERMINAL NUMBERING CONVENTION SHALL CONFORM TO JEDEC 95-1 SPP-002.
 - COPLANARITY APPLIES TO THE EXPOSED HEAT SINK SLUG AS WELL AS THE TERMINALS.
 - REFER TO APPLICATION NOTE WAN_0118 FOR FURTHER INFORMATION REGARDING PCB FOOTPRINTS AND QFN PACKAGE SOLDERING.
 - THIS DRAWING IS SUBJECT TO CHANGE WITHOUT NOTICE.

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REVISION HISTORY

| DATE | REV | ORIGINATOR | CHANGES |
|----------|-----|------------|---|
| 29/09/11 | 4.5 | JMacD | Order codes changed from WM8976GEFL/V and WM8976GEFL/RV to WM8976CGEFL/V and WM8976CGEFL/RV to reflect change to copper wire bonding. |
| 29/09/11 | 4.5 | JMacD | Package Diagram changed to DM101.A |

Mouser Electronics

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[Cirrus Logic:](#)

[WM8976CGEFL/V](#) [WM8976CGEFL/RV](#)

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