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*ACPWorkbench* *AP82X* *DU56X*  
*DU26X*  
*USER MANUAL*

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*Version 2.7.3*

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# 1. INTRODUCTION

The Audio Codec Processor Workbench (ACPWorkbench) is designed for online real-time configuring of MVSilicon's chips such as AP82x, DU56x and DU26x. It has the following features:

- ✓ Direct chip control via UART / USB interface.
- ✓ Simplified GUI for easy operation of hardware registers and configuration of audio effects.
- ✓ Visual filter design tool for parametric EQ.

## 2. GETTING STARTED

ACPWorkbench contains the following files:

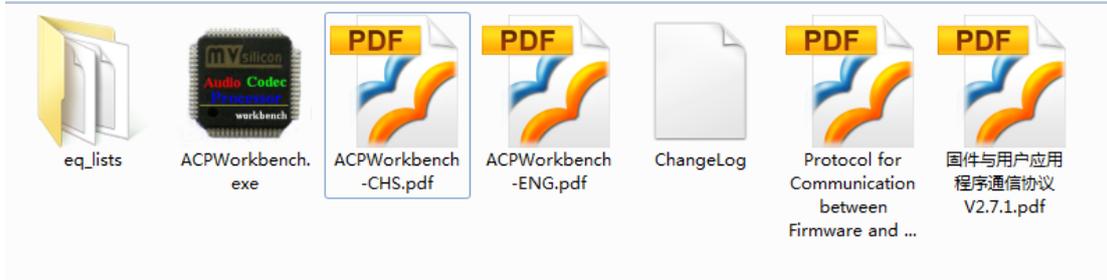


Figure 1. ACPWorkbench files

- ✓ ACPWorkbench.exe is the main executable.
- ✓ eq\_lists folder stores XML files for EQ parameters.
- ✓ ACPWorkbench-ENG.pdf is the help document in English.
- ✓ ACPWorkbench-CHS.pdf is the help document in Chinese.
- ✓ ChangeLog is the change logs file.
- ✓ Protocol document for communication between firmware and User Application.

### 2.1 Launch ACPWorkbench

When ACPWorkbench is launched, it will automatically find and connect the demo board via the connected UART (serial) or USB (HID) port. Once connected, ACPWorkbench will read all the configurations in chip and update its GUI controls accordingly. Please ensure the demo board is connected to PC and powered on. Whenever the demo board is powered on/off, ACPWorkbench will always try to re-connect it.

## 3. MAIN MENUS

### 3.1 Interface

The ACPWorkbench's main window consists of 3 functional pages, i.e. Audio Module, Audio Effect and Log, which are shown below. These pages may look different depending on the type of chips that is connected.

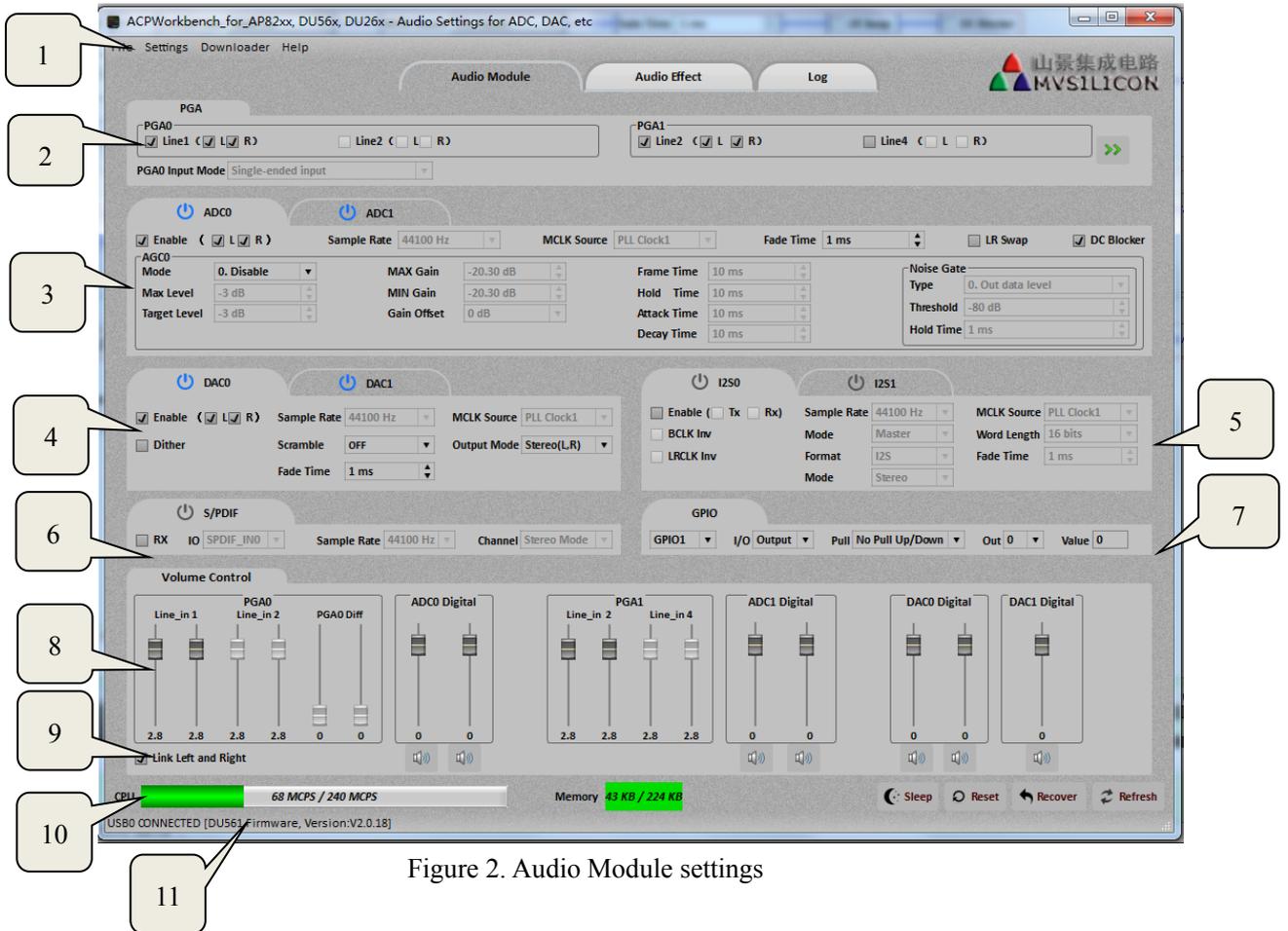


Figure 2. Audio Module settings

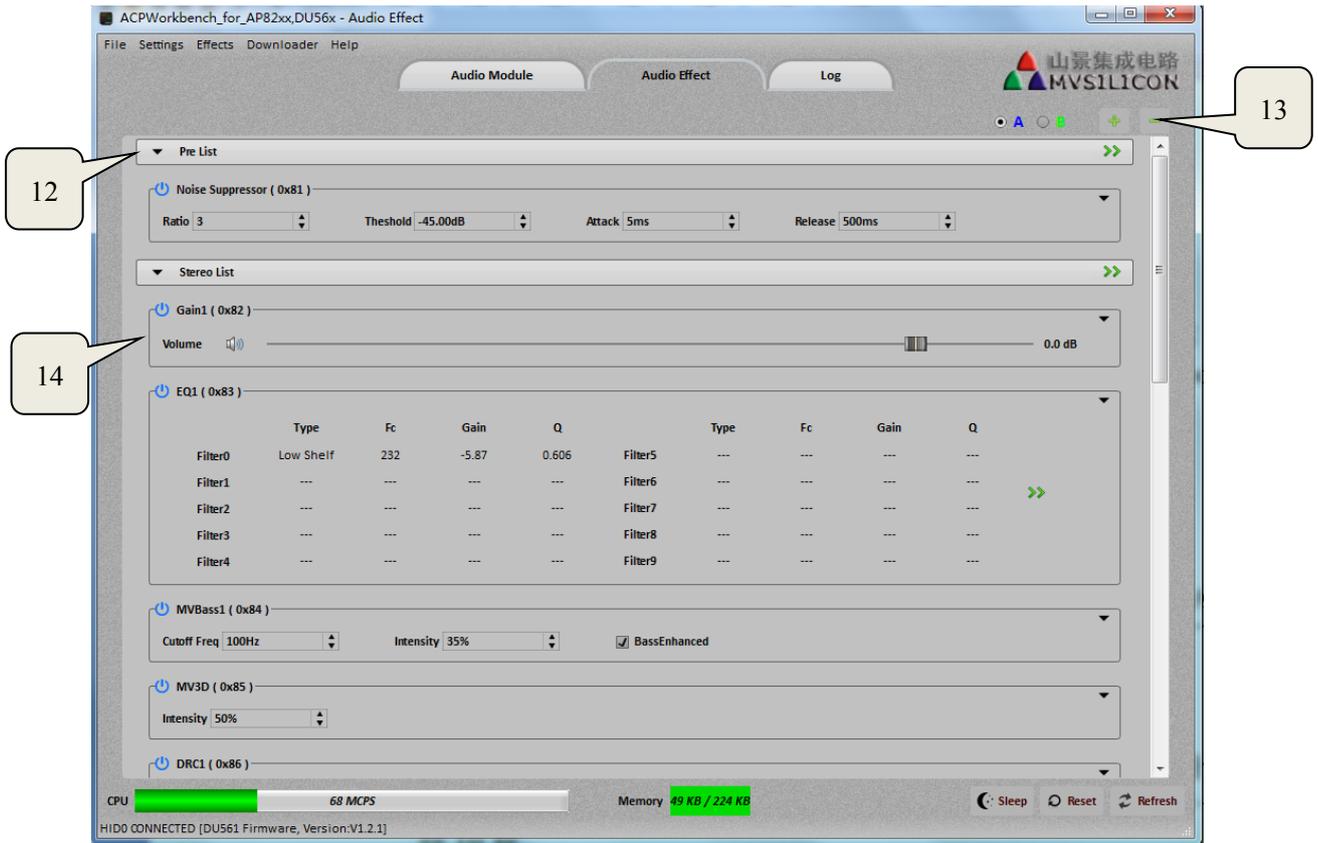


Figure 3. Audio Effect settings

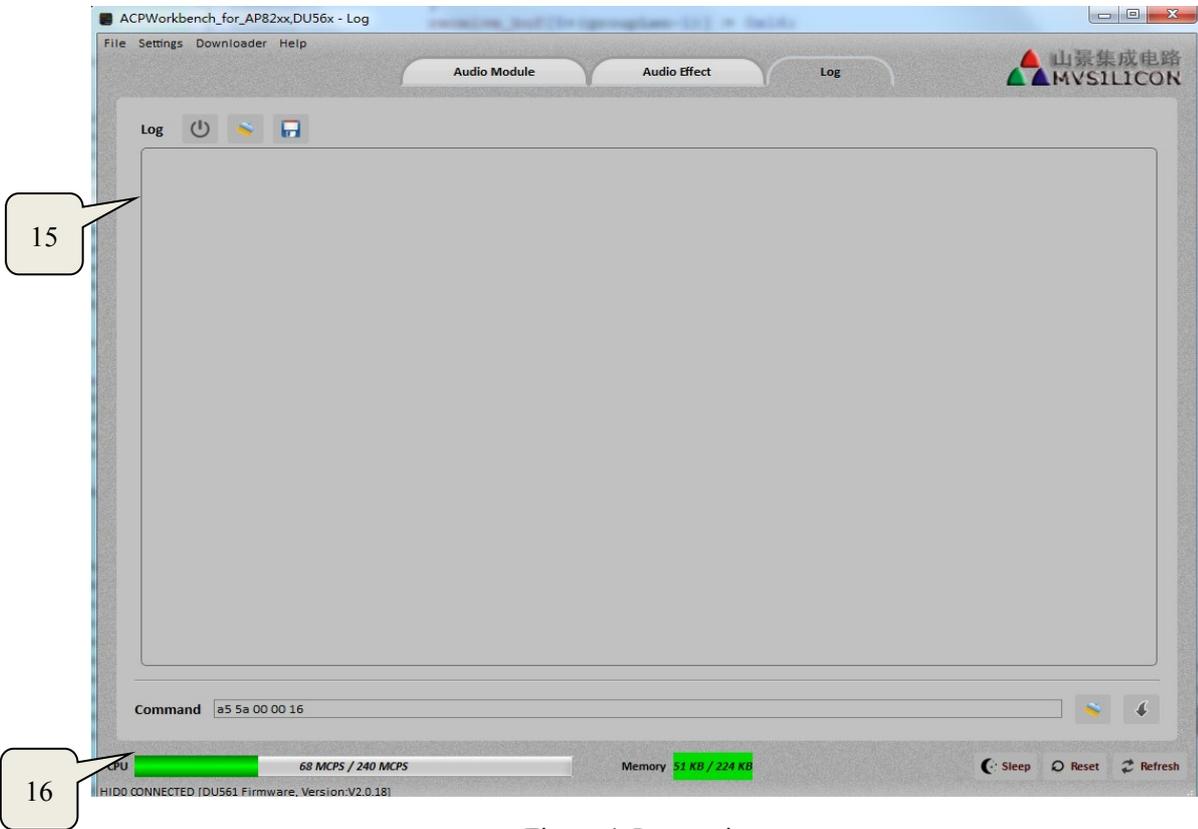


Figure 4. Log settings

1. Main menu: Audio module page includes file, settings, downloader and help menu. Audio effect page has one more menu which is effects menu than audio module page.
2. PGA: audio path selection and mic gain boost selection.
3. ADC: configuration of ADC0 and ADC1 parameters.
4. DAC: configuration of DAC0 and DAC1 parameters.
5. I2S: configuration of I2S0 and I2S1 parameters.
6. S/PDIF: configuration of S/PDIF parameters.
7. GPIO: configuration of GPIO1 and GPIO2 parameters.
8. Volume Control: PGA0, PGA1, ADC0 digital, ADC1 digital, DAC0 digital and DAC1 digital volume control.
9. Link Left and Right Channels: if it is in the selected state, the left and right channels are controlled simultaneously. Otherwise, it will be controlled separately
10. Display of CPU and memory usage in real time.
11. Status bar: status of UART or USB port connection.
12. Audio effect list: name of audio effects list.
13. Audio effect switching: switches between different configurations of audio effects
14. Effects: configuration of audio effects parameters.
15. Audio effect switching: switches between different configurations of audio effects
16. Log interface: information display.
17. Log interface: command control.

## 3.2 Menu

### 3.2.1 File

File menu contains import/export audio settings, export flash binary file and exit as follows.

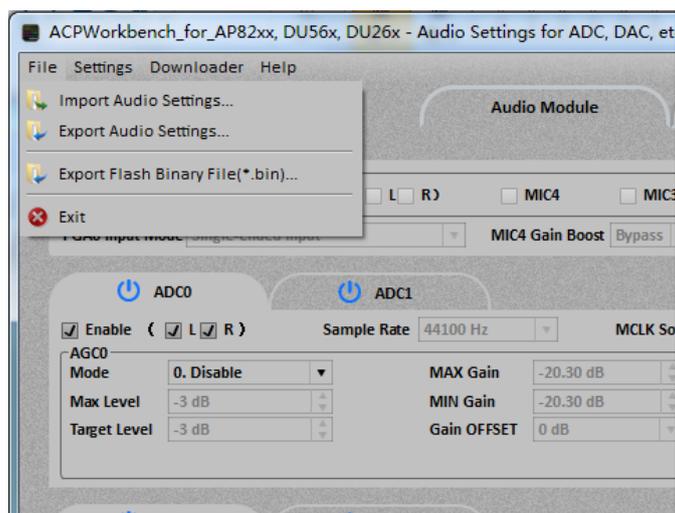


Figure 5. file menu

**Import Audio Setting:** imports audio settings from a stored/exported audio

configuration file.

**Export Audio Setting:** exports audio settings (all the audio configuration) into a file.

**Export Flash Binary File:** exports flash binary file with configurations.

### 3.2.2 Settings

File menu contains communication type, sample rate (global) and MCLK source (global).

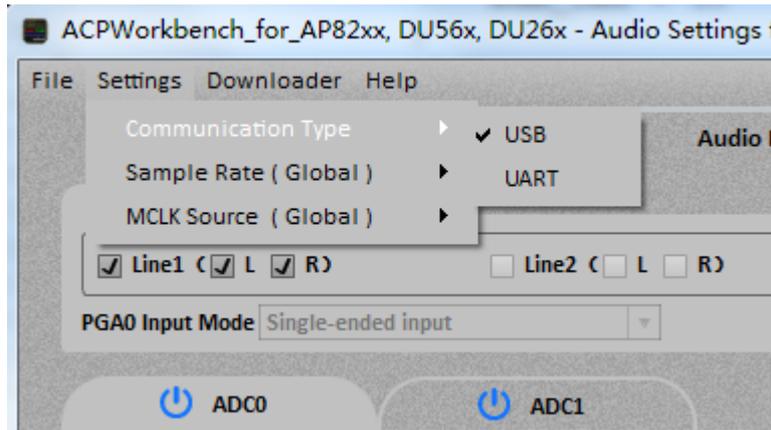


Figure 6. Settings menu

**Communication type:** USB or UART port. ACPWorkbench can automatically detect UART port or USB port and get connected. You can also make a selection if both ports are available.

**Sample rate (global):** inactive mode and 9 sample rates (8000~48000 Hz). Local sample rates of audio module are disabled when one of the 9 sampling rates is selected. Local sample rates of audio module are enabled only when system (global)'s inactive mode is selected.



Figure 7. Global sample rate

**MCLK source (global):** global MCLK source: PLL Clock1, PLL Clock2, OSC IN, MCLK IN0, MCLK IN1 and system MCLK source inactive. Local MCLK sources of audio module are disabled when one of the 5 MCLK sources is selected. Local MCLK sources of audio module are enabled when inactive is selected.

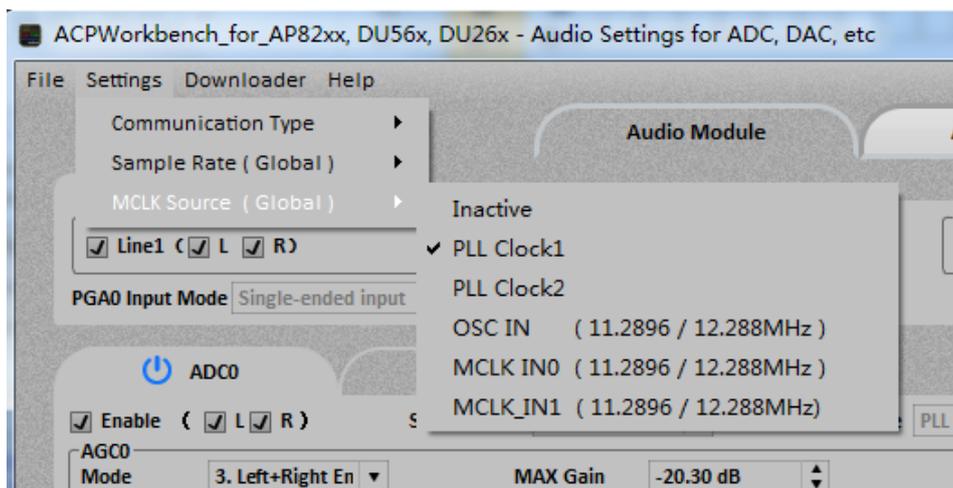


Figure 8. Global MCLK source

### 3.2.3 Downloader

#### 3.2.3.1 Firmware upgrader

The ACPWorkbench supports online upgrading. By clicking “Downloader” menu, “Firmware Upgrader” window will pop up as shown below. After selecting the new firmware bin file, click the upgrade button. The firmware will be downloaded into the flash in chip. The progress of upgrading will be displayed in real-time.

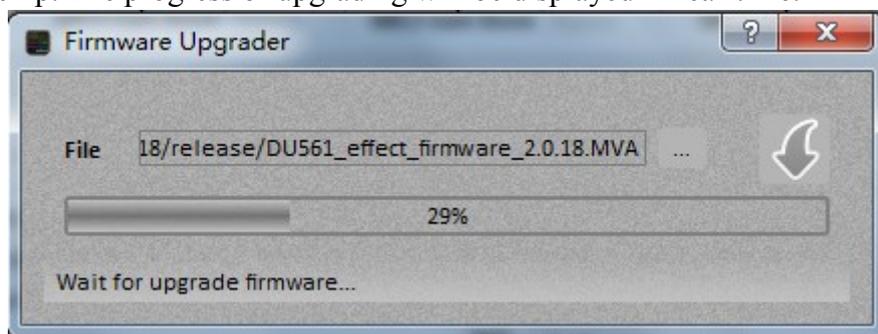


Figure 9. Firmware upgrader window

#### 3.2.3.2 Save configurations to flash

The ACPWorkbench supports saving configurations to flash. The chip will automatically load the saved configurations from flash upon power up.

### 3.2.4 Effects

When ACPWorkbench is switched to Audio Effect page, the main menu will show effect menu items:

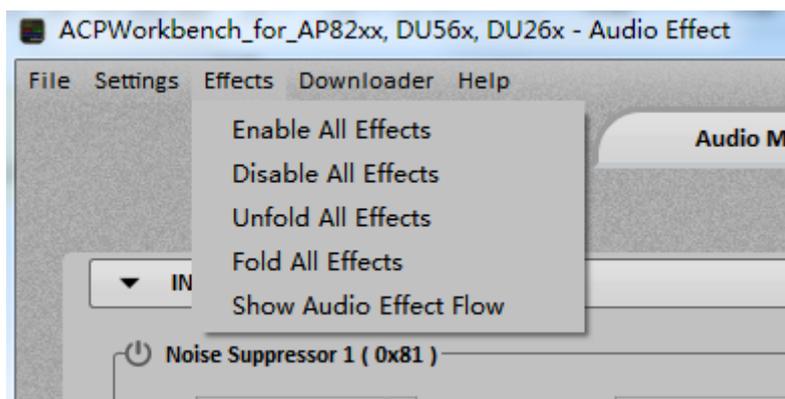


Figure 10. Effects

[Enable all effects](#): enables all effects.

[Disable all effects](#): disables all effects.

[Unfold all effects](#): unfolds all effects interface and show all parameters.

[Fold all effects](#): folds all effects interface and hide all parameters.

[Show audio effect flow](#): show audio effect flow of the current firmware .

### 3.2.5 Help

[Content-Chinese](#): opens the ACPWorkbench Chinese manual.

[Content-English](#): opens the ACPWorkbench English manual.

[Password](#): enters password if current firmware is encrypted.

[About ACPWorkbench](#): information about ACPWorkbench.

## 4. AUDIO MODULE

Audio settings include PGA analog in, ADC, DAC, I2S, S/PDIF, GPIO and volume control.

### 4.1 PGA

#### 4.1.1 Analog input

**Line1:** enables/disables Line 1

**Line2:** enables/disables Line 2

**MIC4:** enables/disables Microphone 4

**MIC3:** enables/disables Microphone 3

**Line4:** enables/disables Line 4

**MIC2:** enables/disables Microphone 2

**MIC1:** enables/disables Microphone 1

**PGA0 Input Mode:** contains Line1/Line2 Single-ended input, Line1 and Line2 left channel differential input, Line1 and Line2 right channel differential input, Line1 and Line2 both channel differential input.

#### 4.1.2 MIC Gain Boost

When MIC is selected as analog input, the corresponding MIC Gain Boost becomes selectable. MIC Gain Boost has 5 options: Bypass, 0dB, 6dB, 12dB and 20dB.

### 4.2 ADC

#### 4.2.1 Enable

It enables/disables ADC's digital part.

#### 4.2.2 Sample Rate ( Local )

ADC supports the following sample rates: 8K, 11.025K, 12K, 16K, 22.05K, 24K, 32K, 44.1K and 48K. Local sample rate of ADC is used only when global sample rate is inactive.

### 4.2.3 MCLK Source ( Local )

ADC local MCLK Source: PLL Clock1, PLL Clock2, OSC IN, MCLK IN0 and MCLK IN1 are supported. Local MCLK source of audio module is used only when global MCLK source is inactive.

### 4.2.4 Fade Time

Fade-in and fade-out time settings.

### 4.2.5 LR Swap

Swap of left and right channel

### 4.2.6 DC Blocker

It removes DC offset in the signals.

### 4.2.7 AGC

**Mode:** AGC function selection. Options include right channel enable, left channel enable, both channels enable.

**Max Level:** AGC maximum level. Range: -34dBFS~-3dBFS.

**Target Level:** AGC target level. Range: -34dBFS~-3dBFS.

**Min Gain:** Minimum gain of PGA. Range: -20.3dB ~ 39.64dB

**Max Gain:** Maximum gain of PGA. Range: -20.3dB ~ 39.64dB

**Gain Offset:** Range:-4~3.5dB

**Frame Time:** AGC frame time. Range: 1ms ~ 4096ms

**Attack Time:** AGC attack time. Range: 1ms ~ 4096ms

**Hold Time:** Hold time. Range: 1ms ~ 4096ms

**Decay Time:** Decay (release) time. Range: 1ms ~ 4096ms

### 4.2.8 Noise Gating

**Threshold:** Noise threshold. Range: -76.5dBFS~-30dBFS

**Type:** Output level or input level.

**Hold Time:** Hold time control. Range: 1ms ~ 4096ms.

## 4.3 DAC

### 4.3.1 Enable

It enables/disables DAC's digital part.

### 4.3.2 Sample Rate ( Local )

Supported DAC sample rates include 8K, 11.025K, 12K, 16K, 22.05K, 24K, 32K, 44.1K and 48K. Local sample rate of audio module is used only when global sample rate is inactive.

### 4.3.3 MCLK Source ( Local )

DAC MCLK Source: PLL Clock1, PLL Clock2, OSC IN, MCLK IN0 and MCLK IN1. Local MCLK source of audio module is used only when global MCLK source is inactive.

### 4.3.4 Fade Time

Fade-in and fade-out time settings.

### 4.3.5 Dither

It enables / disables dithering.

### 4.3.6 Scramble

Scramble mode selection: Off, DWA, Random DWA, Butterfly DWA.

### 4.3.7 Output Mode

Output mode selection:

**Stereo (L, R):** stereo output,  $Lo = Li$ ,  $Ro = Ri$

**Stereo (R, L):** stereo output with left and right channel swapped,  $Lo = Ri$ ,  $Ro = Li$

**Mono1:** mono1 output,  $Lo = Ro = (Li+Ri)/2$

**Mono2:** mono2 output,  $Lo = (Li+Ri)/2$ ,  $Ro = -(Li+Ri)/2$ .

## **4.4 I2S**

### **4.4.1 Enable**

It enables/disables I2S.

### **4.4.2 Sample Rate ( Local )**

Supported I2S sample rates: 8K, 11.025K, 12K, 16K, 22.05K, 24K, 32K, 44.1K, 48K, 88.2K, 96K, 176.4K and 192K. Local sample rate of audio module is used only when global sample rate is inactive.

### **4.4.3 MCLK Source ( Local )**

I2S MCLK Source: PLL Clock1, PLL Clock2, OSC IN, MCLK IN0 and MCLK IN1. Local MCLK source of audio module is used only when global MCLK source is inactive.

### **4.4.4 Fade Time**

Fade-in and fade-out time settings.

### **4.4.5 Master/Slave Mode**

I2S can operate in master or slave mode.

### **4.4.6 Word Length**

Four different audio data word length are supported: 16 bits, 20 bits, 24 bits and 32 bits.

### **4.4.7 Format**

Five different audio data formats are supported: Right ( LSB ), Left ( LSB ), I2S, DSP MODE A, DSP MODE B.

## 4.4.8 BCLK / LRCLK Invert

Both BCLK and LRCLK support clock inversion.

## 4.5 S/PDIF

**TX / RX:** S/PDIF supports TX enable and RX enable.

**IO:** supports GPIO multiplex, contains: SPDIF\_IN0 and SPDIF\_IN1.

**Sample Rate:** sample rate. Range: 8K~192K.

**Channel:** supports stereo mode and mono mode.

## 4.6 GPIO

The control of GPIO1 and GPIO2 includes input and output control (I/O), pull-up control (pull), output high and low level control (out) and input level detection (value).

## 4.7 Volume Control

### 4.7.1 PGA0 volume

**MIC4 / MIC3:** controls microphone volume. Volume range: -18.4dB ~ 41.6dB.

**Line-in 1:** controls Line-in1 volume for both left and right channels. Volume range: -46.1dB ~ 13.6dB

**Line-in 2:** controls Line-in2 volume for both left and right channels. Volume range: -46.1dB ~ 13.6dB

### 4.7.2 ADC0 Digital volume

**ADC0 Digital:** controls ADC0 digital volume for both left and right channels.

Volume range: mute ~ 12dB

**Mute:** controls ADC0 digital mute for both left and right channels.

### 4.7.3 PGA1 volume

**Line-in 2:** controls Line in2 volume for both left and right channels. Volume range: -46.1dB ~ 13.6dB

**Line-in 4:** controls Line in4 volume for both left and right channels. Volume range: -46.1dB ~ 13.6dB

**MIC2:** controls MIC2 volume. Volume range: -18.4dB ~ 41.6dB.

**MIC1:** controls MIC1 volume. Volume range: -18.4dB ~ 41.6dB.

#### **4.7.4 ADC1 Digital volume**

**Digital:** controls ADC1 digital volume for both left and right channels. Volume range: mute ~ 12dB

**Mute:** controls ADC1 digital mute for both left and right channels.

#### **4.7.5 DAC0 volume settings**

**DAC0 Digital:** controls DAC0 digital volume for both left and right channels. Volume range: mute ~12dB.

**Mute:** controls DAC0 digital mute for both left and right channels.

#### **4.7.6 DAC1 volume settings**

**DAC1 Digital:** controls DAC1 digital volume. Volume range: mute ~12dB.

**Mute:** controls DAC0 digital mute for both left and right channels.

#### **4.7.7 Link Left and Right Channels.**

Synchronous volume adjustment for both left and right channels is enabled when "*Link Left and Right Channels*" checkbox is checked.

## 5. AUDIO EFFECTS

### 5.1 Audio Effect Introduction

The content on Audio effect page is updated according to the effect list uploaded by chip.

No.	Effects
0	Auto Tune
1	DC Blocker
2	Dynamic Range Compressor
3	Echo
4	EQ
5	Noise Suppressor
6	Frequency Shifter
7	Howling Suppressor
8	Noise Gate
9	Pitch Shifter
10	Reverb
11	Silence Detector
12	3D
13	Virtual Bass
14	Voice Changer
15	Gain Control
16	Vocal Cut
17	Reverb Plate
18	Reverb Pro
19	Voice Changer Pro
20	Phase Control
21	Vocal Remover

### 5.1.1 Auto Tune

Auto tune refers to an audio effect of adjusting a person's tone to the nearby note. It can be configured by key and snap mode.

**Key:** 0-12 means A-flat, A, B-flat, B, C, D-flat, D, E-flat, E, F, G-flat, G major and chromatic scale.

**Snap mode:** near, upper, lower

### 5.1.2 DC Blocker

### 5.1.3 DRC (Dynamic Range Compression)

The function of the DRC is to adjust the signal gain in conditions where the input amplitude is unknown or varies over a wide range. Visual design tool for DRC by clicking **>>** button at DRC interface.

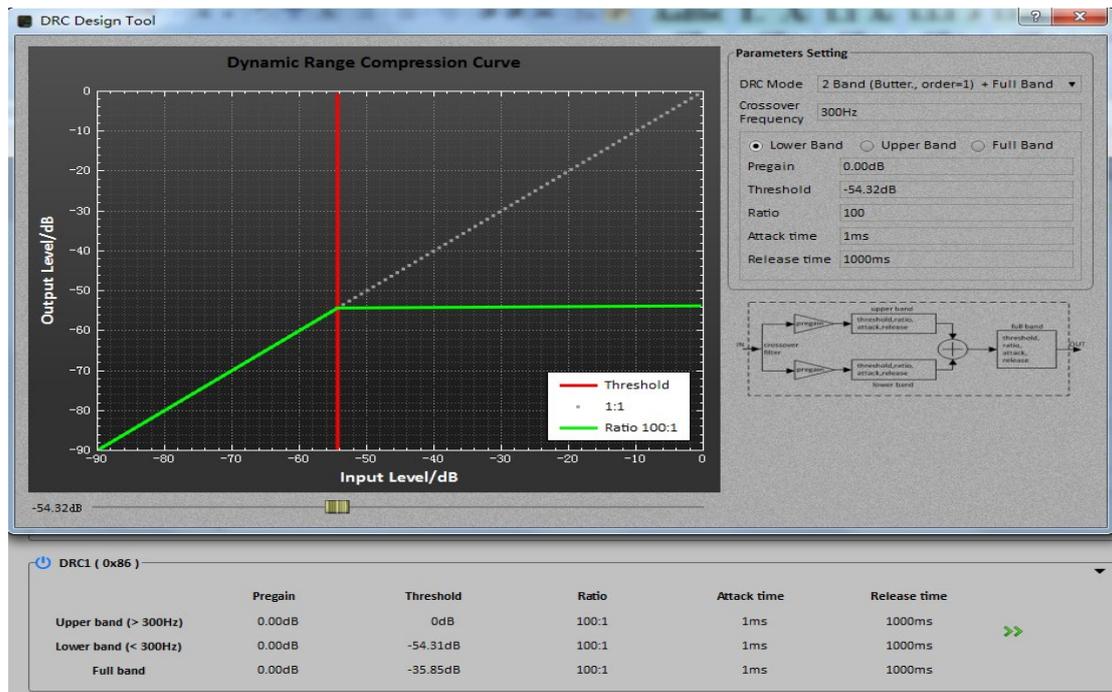


Figure 11. DRC parameters design tool

**Pregain:** pregain (in dB) applied before DRC effects. Q4.12 format to represent value in range [0.000244140625, 8) to cover -72dB ~ 18dB

**Threshold:** a compressor reduces the level of an audio signal if its amplitude exceeds a certain threshold

**Ratio:** the amount of gain reduction is determined by ratio.

**Attack Time:** time to control how quickly it enables the compression.

**Release Time:** time to control how quickly it disables the compression.

**Crossover Frequency:** crossover frequency in Hz to separate the full band into 2 subbands.

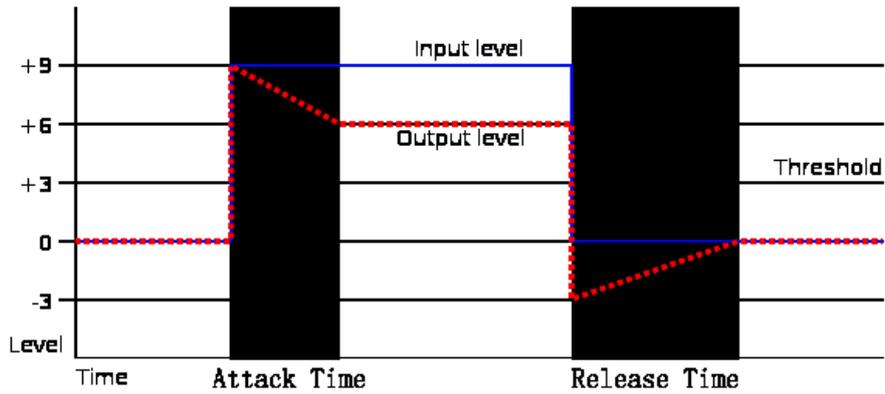


Figure 12. DRC attack time and release time

### 5.1.4 Echo

Echo can be controlled by delay time and attenuation.

**Delay:** delay time. Range: 0 ~1000ms.

**Attenuation:** attenuation applied to the echo. Range from: 0 ~ -90dB.

**Direct Sound:** enables\disables direct sound

**Cutoff Frequency:** cutoff frequency for low pass filter. Range: 0~24000Hz

### 5.1.5 EQ

By clicking  button, users are directed to the equalizer editor tool. For more information about equalizer editor, please refer to section ["Parametric Equalizer Editor"](#).

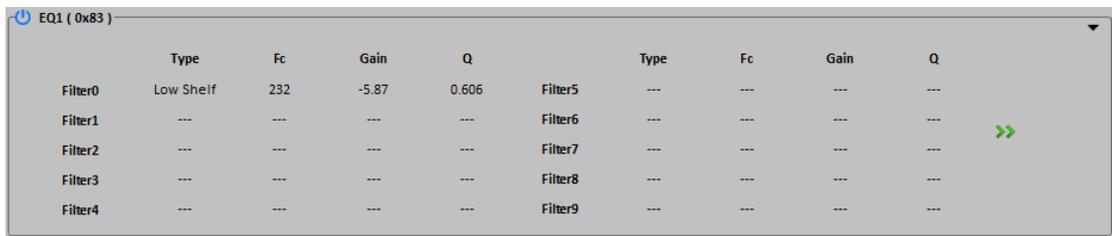


Figure 13. EQ effect interface

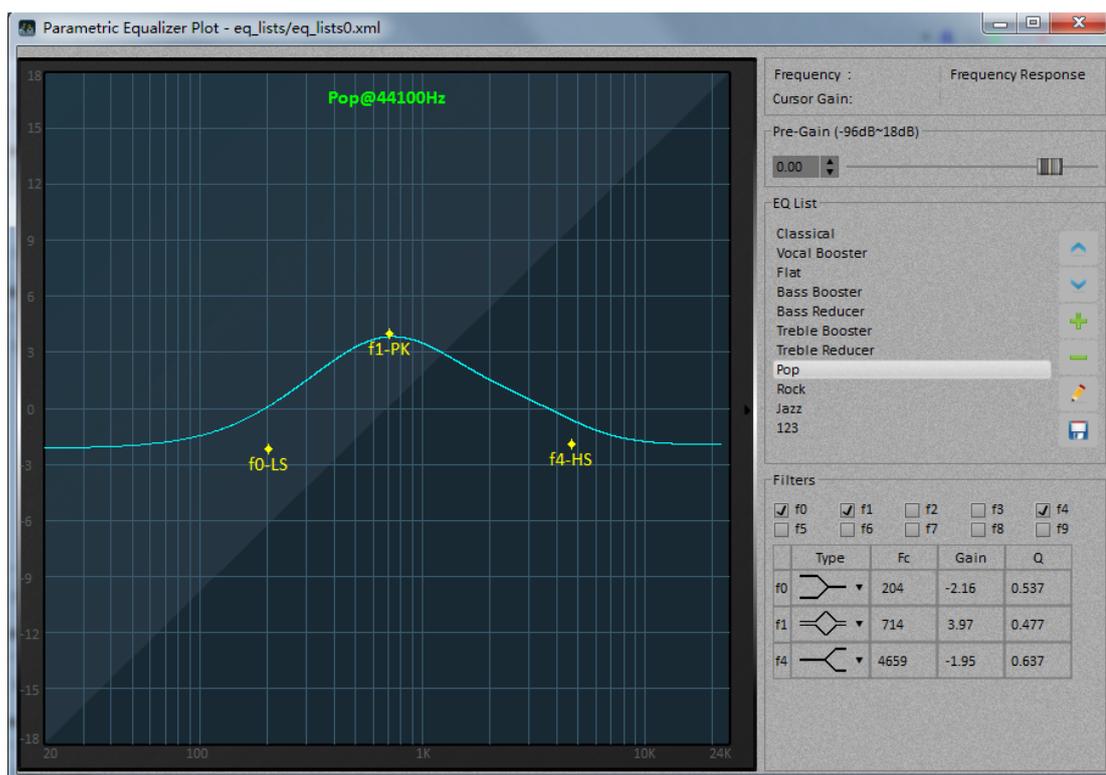


Figure 14. EQ parameters design tool

### 5.1.6 Noise Suppressor

**Threshold:** the level below which gain is applied to the input signal. Range: -90 dB~0dB.

**Ratio:** the output/input ratio for signals that undershoot the operation threshold. Range: 1 ~ 1000.

**Attack Time:** amount of time (in millisecond) it takes applied gain to change. Range: 0~7500ms.

**Release Time:** amount of time (in millisecond) it takes applied gain to restore. Range: 0~7500ms

### 5.1.7 Frequency Shifter

Frequency shifter shifts all the frequencies up or down by a specified amount.

**Frequency Shifter:** frequency shift in Hz. Allowable value: -7, -5, -3, +3, +5, +7. “-” means shift down. “+” means shift up.

### 5.1.8 Howling Control

**Howling suppression:** enables or disables howling suppression.

### 5.1.9 Noise Gate

Noise gate suppresses the input signal when the signals are below threshold and considered as noise. It is configured by the following parameters:

**Lower Threshold:** -90 ~ 0dB

**Upper Threshold:** -90 ~ 0dB

**Attack Time:** 0 ~ 7500ms

**Release Time:** 0 ~ 7500ms

**Hold Time:** 0 ~ 7500ms

### 5.1.10 Pitch Shifter

Pitch shifter raises or lowers the original pitch of the signals.

**Key:** controls pitch shifter semitone step. Range: -12.0 semitone ~ 12.0 semitone.

### 5.1.11 Reverb

The parameters for reverberation configuration are as follows:

**Dry:** percentage of direct sound output. Range: 0 ~ 200%.

**Wet:** percentage of reverberation output. Range: 0 ~ 300%.

**Width:** controls the difference of left and right channel. Wider value gives more obvious stereo feeling. Range: 0 ~ 100%

**Room Size:** reflects the width and depth of the sound field. Larger value will lead to more extensive field and greater the sense of room size. The reverberation time will be indirectly controlled by the parameter. The reverberation time is also called T60. Range: 0 ~ 100%

**Damping:** high frequency signal are highly likely to be attenuated in the reverberation. The greater room size, the more objects in space and the less smooth surface of the object and wall will lead to the more attenuation of the high frequency. The parameter damping controls the speed of the reverberation time. The greater value, the fast attenuation of the high frequency. Range: 0 ~ 100%.

**Mono:** mono processing for sake of speed only.

### 5.1.12 Silence Detector

Signal detector detects the energy level of the current frame.

**PCM amplitude:** Range: 0~32768.

### 5.1.13 MV3D

3D effect. It has one parameter for configuration:

**Intensity:** relates to stereo depth. Range: 0% ~ 100%

### 5.1.14 MV Bass

MV bass is a kind of virtual bass effect to enhance the bass feeling, which is more useful for small speakers.

**Cutoff-Frequency:** speaker's cut-off frequency in Hz. Range: 30 ~300 Hz.

**Intensity:** bass intensity. Range: 0%~100%.

**BassEnhanced:** enhance bass.

### 5.1.15 Voice Changer

Voice changer alters a person's voice characteristics so that it sounds like a target one (e.g. male or female). It is configured by pitch ration and formant ratio.

**Pitch Ratio:** 50 ~ 300%

**Formant Ratio:** 66 ~ 200%

### 5.1.16 Gain Control

Gain control includes the following parameters:

**Mute:** mute or unmute.

**Gain:** range: -72.2 ~ 12dB.

### 5.1.17 Vocal Cut

Vocal Cut removes vocal part from a song leaving only the background music.

### 5.1.18 Reverb Plate

Plate reverb effect.

**High Cutoff Frequency:** cutoff frequency above which the signal are filtered out.  
Range: 0 ~ sample rate/2.

**Predelay:** pre-delay in samples. Range: 0~4410

**Diffusion:** density of reverb tail. Range: 0~100 for 0~100%.

**Decay:** decay of reverb tail. Range: 0~100 for 0~100%.

**Damping:** high frequency damping. Range: 0~10000 for 0.00~100.00%.

**Wet Dry Mix:** the ratio of wet (reverberated) signal to the mixed output (wet + dry). Range: 0~100 for 0~100%.

**Modulation:** enables/disables modulation.

### 5.1.19 Reverb Pro

More advanced reverb effect.

**Dry:** final dry mix [-70 to 10] dB

**Wet:** final wet (late reverb) mix [-70 to 10] dB

**Erwet:** final wet (early reflection) mix [-70 to 10] dB

**Erfactor:** early reflection factor [50 to 250] %

**Erwidth:** early reflection width [-100 to 100] %

**Ertolate:** early reflection amount [0 to 100] %

**Rt60:** reverb time decay [100 to 15000] millisecond

**Delay:** amount of delay [0 to 100] millisecond

**Width:** width of reverb L/R mix [0 to 100] %

**Wander:** LFO(low-frequency oscillator) wander amount [10 to 60] %

**Spin:** LFO(low-frequency oscillator) spin amount [0 to 1000] %

**Inputlpf:** lowpass cutoff for input [200 to 18000] Hz

**Damplpf:** lowpass cutoff for dampening [200 to 18000] Hz

**Basslpf:** lowpass cutoff for bass [50 to 1050] Hz

**Bassb:** bass boost [0 to 50] %

**Outputlpf:** lowpass cutoff for output [200 to 18000] Hz

### 5.1.20 Voice Changer Pro

Voice changer pro's parameters are the same as that of voice changer. It usually has better performance than voice changer.

- ✓ Smaller delay
- ✓ Higher output quality
- ✓ Full pitch coverage

### 5.1.21 Phase Control

**Phase Difference:** 0 or 180 degrees.

### 5.1.22 Vocal Remover

Compared with vocal cut, vocal remover is more frequency dependent and usually has better performance.

**Lower Frequency:** Lower limit frequency of the vocal detect.

**Higher Frequency:** Higher limit frequency of the vocal detect.

### 5.1.23 Pitch Shifter Pro

Compared with pitch shifter effect, the pitch shifter pro effect is more ideal and professional.

**Key:** controls pitch shifter semitone step. Range: -12.0 semitone ~ 12.0 semitone.

### 5.1.24 MVBass Classic

**Cutoff-Frequency:** speaker's cut-off frequency in Hz. Range: 30 ~300 Hz.

**Intensity:** bass intensity. Range: 0%~100%。

### 5.1.25 PCM Delay

This effect is used for data stream delay with a maximum delay of 50 ms.

**Delay:** Time delay of data stream.

### 5.1.26 Harmonic Exciter

Harmonic exciter effect can improve the clarity and penetration of the voice and make it more pleasant.

**Cutoff-Frequency:** cut-off frequency in Hz. Range: 1000 ~10000 Hz.

**Dry:** final dry mix. Range: 0 ~ 100%.

**Wet:** final wet mix. Range: 0 ~ 100%。

## 5.2 Audio Effect List Modification

Karaoke SDK or Normal SDK do not support this feature, but for DU series, it supports the modification of audio effect lists. DU series firmware supports adding, deleting, moving up, moving down audio effects and modifying their name. In addition, audio effect lists interface and each audio effect interface can be folded and unfolded.

- ✓ When expanding or folding the audio effect list, click '▼' or '▶'. The effect is as follows.

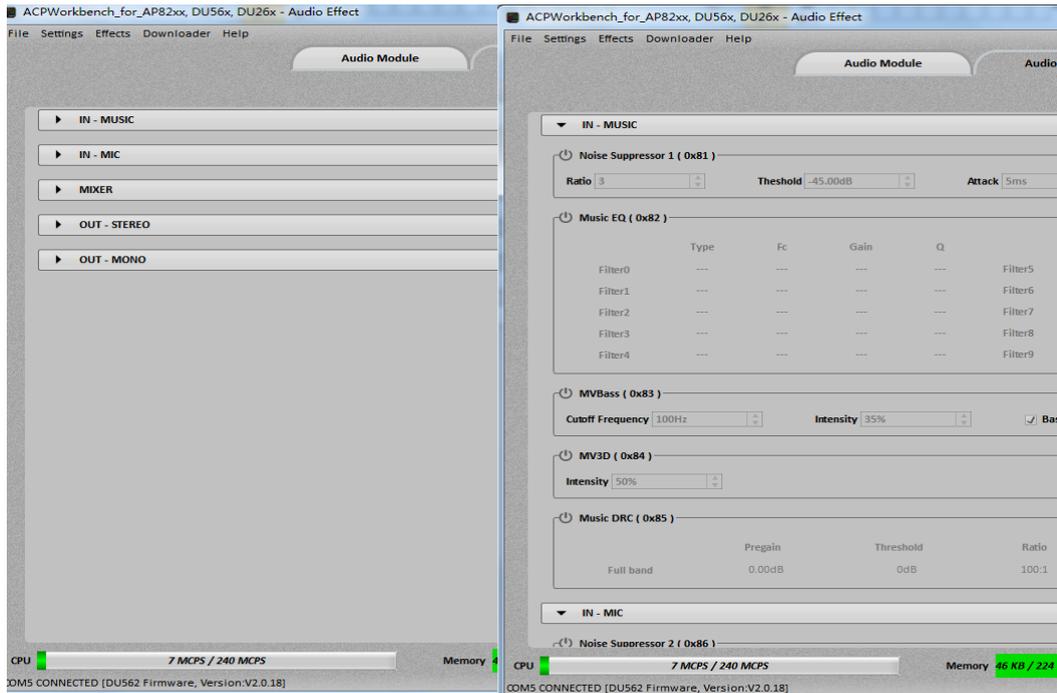


Figure 15. Fold and unfold audio effect lists

- ✓ Clicking on the top right corner '▾' of audio effect interface will fold the current audio effect interface, and clicking on the top right corner '▶' will expand the audio effect interface. If current firmware have many audio effects, right-click mouse can choose one button folding or one button unfolding function.

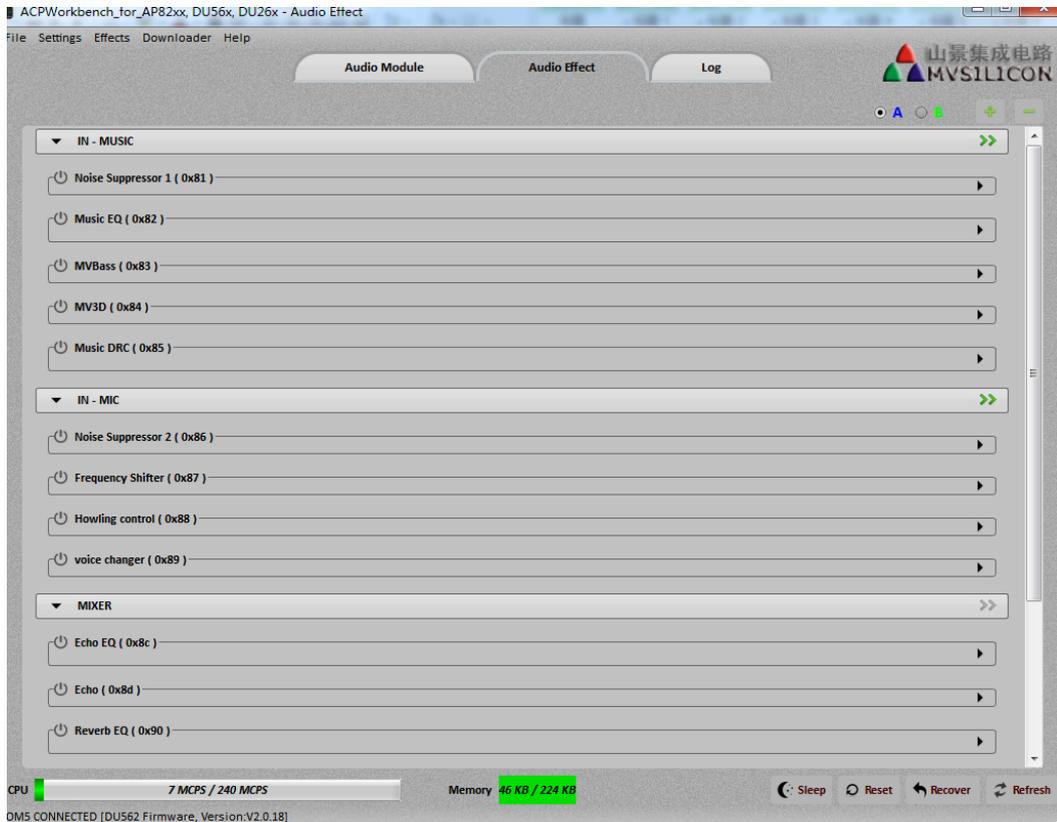


Figure 16. Fold and unfold audio effect interface

- ✓ Add, delete, move up, move down and modify the effect name.
- Click the right button '➡' of the audio effect list to spread the effect of the current effect list. As shown in the figure below. Click the button '⬆' to move up the audio effect position, click the button '⬇' to move down the audio effect position, click the button '⊕' to add the audio effect, click the button '⊖' to delete the audio effect and click the button '✎' to modify the name of the current audio effect. Click the 'OK' button to confirm the operation and click 'Cancel' button to cancel the operation.

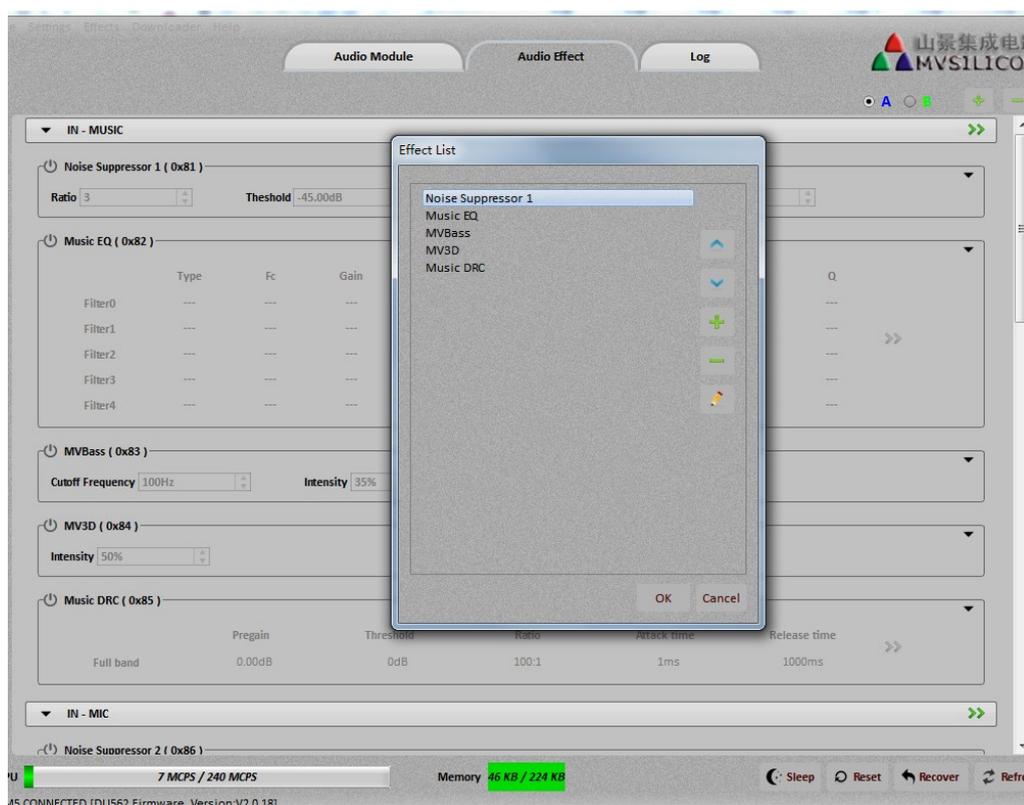


Figure 17. Modify current audio effect list

### 5.3 Fast Switching of Audio Effect

An effect switching function  is implemented in the audio effect interface. The 2 configurations, i.e. A and B are given by default. More or less number of configurations can be added or deleted by clicking '+' or '-', with up to 10 configurations supported. Take the default A and B as an example, the operation steps are as follows:

- ✓ Select A and all the effect parameters will be saved in A.
- ✓ Select B and all the effect parameters will be saved in B.
- ✓ Switch between A and B effect, the difference in output with different configuration can be heard in real-time.
- ✓ Audio effect's parameters can be exported in form of BIN file.

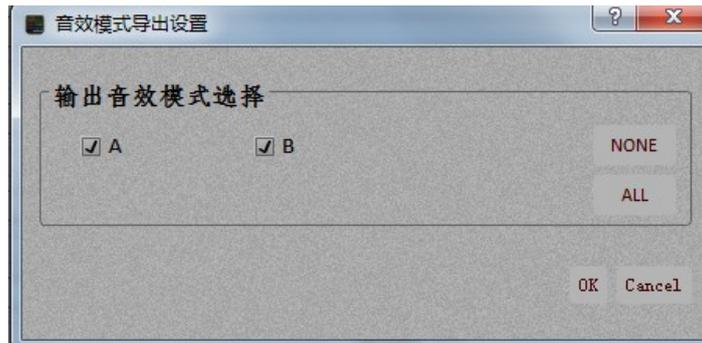


Figure 18. Export bin files

## 6. LOG PAGE

Log page can be used to monitor the communication between ACPWorkbench and the chip. It is turned off by default and can be turned on by clicking on . The log information will appear if the button turns blue. Click  to clean the log and click  to save the log.

## 7. CPU AND MEMORY USAGE

ACPWorkbench can read and display the CPU and memory usage of the chip in real time after it has connected to the demo board successfully.

### 7.1 CPU Usage

ACPWorkbench displays CPU usage in real time on the bottom progress bar. The unit MCPS means mega cycles per second. When CPU usage exceeds that maximum limit, the process bar's color will turn red to give a warning. In this case, the chip is not able to work in real time, and certain delays can be heard in output as a result.

### 7.2 Memory Usage

Memory usage will be displayed in digital form. Range: 0 ~ 224 in KB (Kilobyte)

## 8. PARAMETRIC EQUALIZER EDITOR

Parametric equalizer editor is a graphic tool for designing filters, generating equalizer parameters by setting number of filters, filter parameters (center frequency  $f_0$ , gain, Q).

The equalizer editor has the following features,

- 1). Filter parameters can be set by mouse drag-drop or mouse wheeling.
- 2). Filter parameters can be set directly.
- 3). EQ can be added, deleted, edited and saved.
- 4). EQ filters parameters can be exported to files.
- 5). EQ plot area can be changed in color.

### 8.1 New/Design EQ

Click  to add a new EQ. This EQ will be added into EQ list widget.

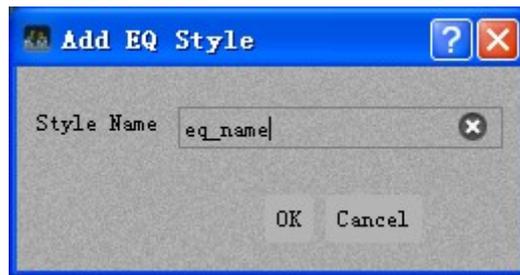


Figure 19. add EQ style pop-up window

To design an EQ, filters can be added or removed by clicking  $f_0$ ~ $f_9$  check box. Filter parameters ( $f_0$ , gain, Q) can be edited directly or changed by drag-dropping mouse or scrolling mouse wheel. If "*Display spectral curve after quantization*" check box is checked, the fixed-point output of the frequency response curve after quantization will be displayed on the plot area. The range for  $f_0$  is 20Hz~24KHz, and the range for gain is -12dB~12dB.

Click  button to rename current EQ.

Click  button to remove current EQ.

After designing EQ filters, click  button to save this EQ into the XML file.

### 8.2 Sample Rate Selection

Frequency response curve will be plotted according to the sample rate set in the main menu.

## 8.3 Export/Import of EQ Parameters

The equalizer editor supports the export of current EQ Parameters into an .ini file. By right-clicking mouse, EQ interface will pop up the import and export parameters menu.

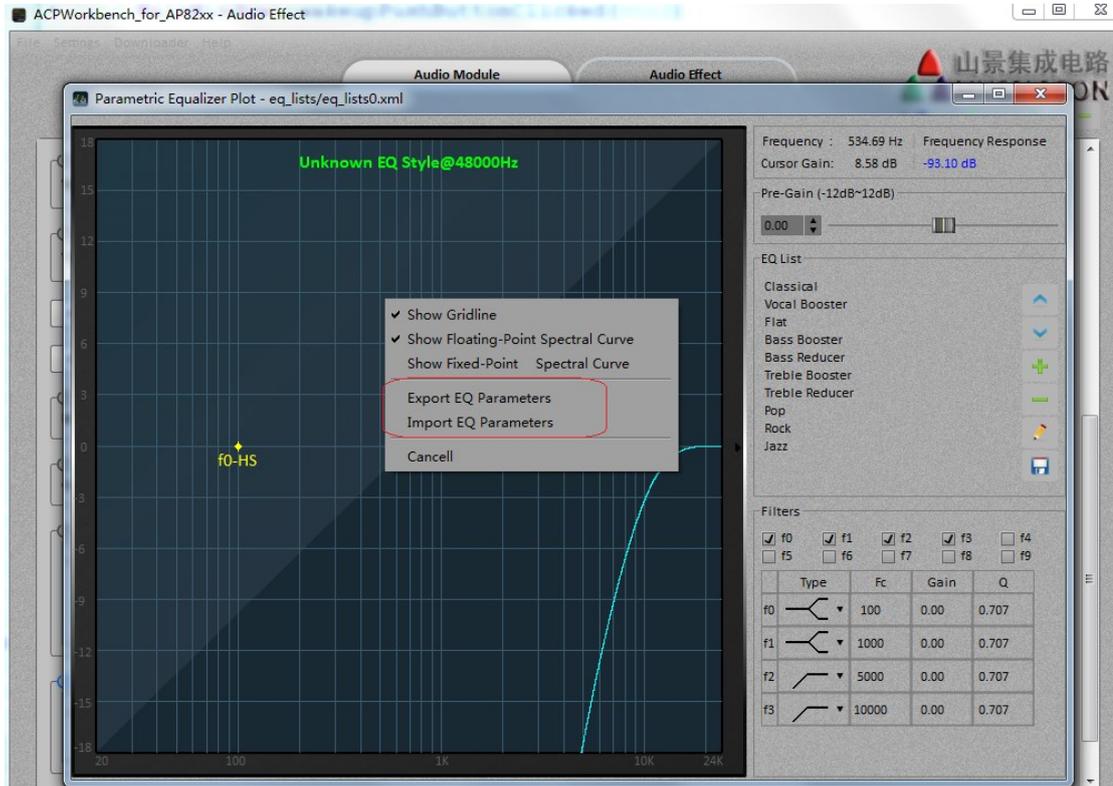


Figure 20. EQ Style Parameters Export/Import