HANDBOOK of GEAZAN Recording-Type 12-In/ 6-out Audio Processor (MX1206)

CONTENTS

1 Introduction

1.1 What does this handbook do?

The handbook will help users how to install and use the recording-type 12-in/ 6-out audio processor. The readers will be engineers who participate in site operation and test.

1.2 Overview of the audio processor

The recording-type 12-in/ 6-out audio processor MX1206 is a professional digital audio processor, designed for both local education and conference by GEAZAN, with 8-channel microphone inputs with adaptive noise reduction and smart mixing, 4-channel line inputs, and 6-channel line outputs. MX1206 has an extremely high SNR reproducing a clear and natural sound. Hi-fidelity sound quality can be achieved with a simple connection. At the same time, users can also freely design the system through PC software to fully meet the high-quality sound mixing needs of education and conference.

2 Functionality and Interface

2.1 Functionality

- 8 balanced Mic inputs, 4 balanced Line inputs, Phoenix connector
- 6 balanced Line outputs, Phoenix connector
- Smart mixing and microphone selection technology
- Dynamic adaptive noise reduction up to 18dB
- Support 48V phantom power
- Sampling frequency 48kHz, A/D-D/A in 24 bit
- Ethernet port for software setting and control
- Serial interface for third-party RS-232 remote control
- RMS and peak value voltage meters; real-time signal magnitude monitoring

- Abundant signal processing module:
  - Filtering: high pass and low pass
  - Equalization: 10 band parametric equalization
  - Volume control: volume balance and volume control
  - Audio processing: adaptive noise suppression
  - Mixing: smart mixing and matrix mixing
  - Delay module & meters module

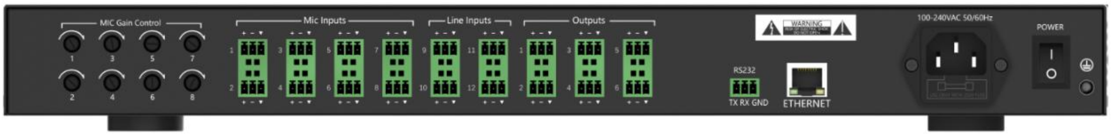
2.2 Appearance and Interfaces

Front

Instructions for Front Panel

Indicator, Interface and Knobs	Function Description
POWER	Power indicator, power on indicator on
RUN	Operation indicator, slow flashing indicates normal operation
NETWORK	Network connection indicator, always on when network and PC control terminal connection is established
LINE OUT(1-2)	Line output 1-2 gain adjustment
LINE OUT(3-4)	Line output 3-4 gain adjustment
LINE OUT(5-6)	Line output 5-6 gain adjustment
LINE IN(9-10)	Line input 9-10 gain adjustment
LINE IN(11-12)	Line input 11-12 gain adjustment
PHANTOM POWER	Phantom power dial switch

Rear

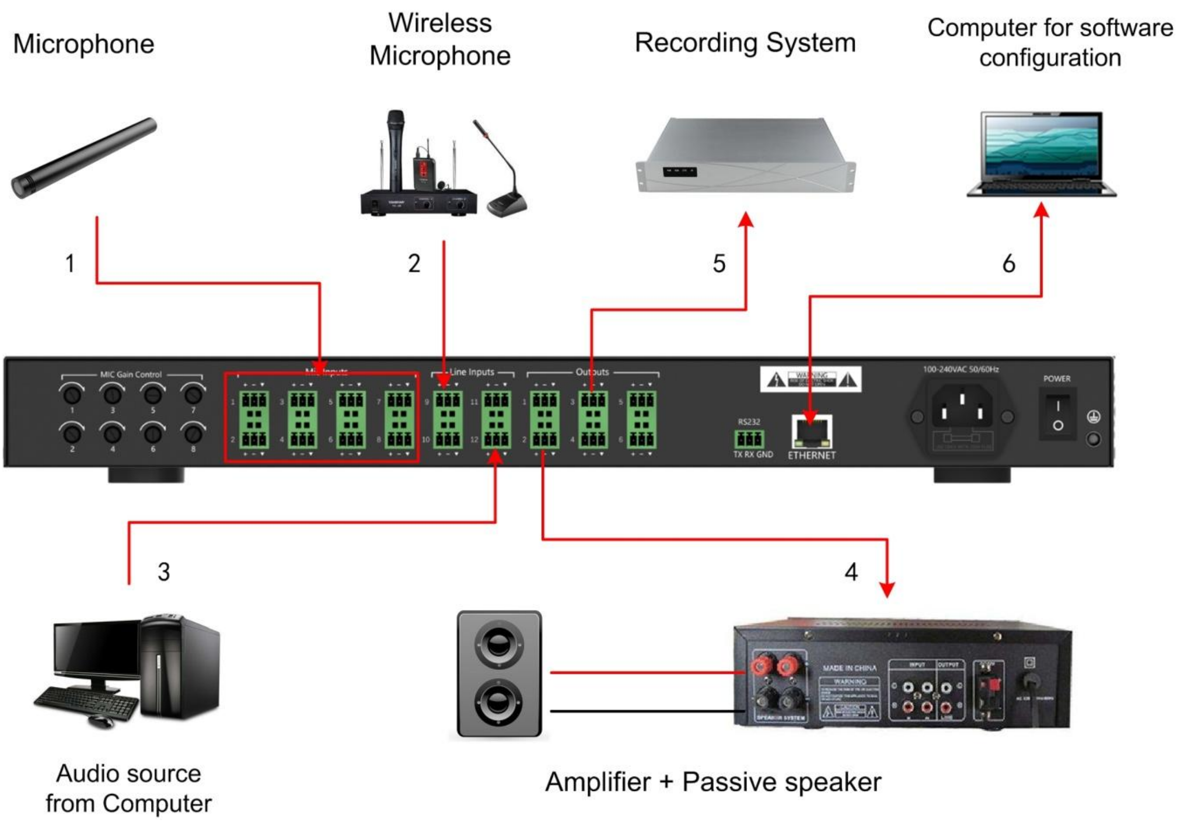


Instructions for Rear Panel

Indicator, Interface and Knobs	Function Description
Mic Gain Control	8-channel Microphone input gain adjustment knob
Mic Inputs(1-8)	8-channel microphone inputs, Phoenix connector
Line Inputs(9-10)	2 (left and right channel) Line inputs, usually for the local audio source, like DVD, laptop, etc.
Line Inputs(11-12)	2 (left and right channel) Line inputs, usually for the local audio source, like DVD, laptop, etc.
Line Outputs(1-2)	2 (left and right channel) Line outputs, local audio line output for local sound reinforcement
Line Outputs(3-4)	2 (left and right channel) line outputs, mix output of microphone inputs and local audio inputs, used for recording service
Line Outputs(5-6)	2 (left and right channel) line outputs, mix output of microphone inputs and local audio inputs, used for recording service
RS232	The serial control interface, may connect with external control terminal.
Ethernet	RJ45 port, connect to PC for software configuration
100-240VAC 50/60Hz	Power interface
POWER	Power switch

3 Connection and Usage

3.1 Connection



Devices will be connected through the interfaces in the rear panel, details are below.

1. Inputs 1-8 are Microphone inputs. When phantom power is powered on, these inputs can connect suspended microphone or gooseneck microphone. The input gains can be adjusted through the knobs on the rear panel. The phantom power is powered on by default. (See section 3.3).

2. Wireless microphone can be connected to Line Input 9 of Audio Processor.

3. Line inputs 10, 11 or 12 can be connected to the audio source from DVD or computer.

4. Outputs 1 & 2 are line output, connecting to the Amplifier or active speaker for playing local music and audio signals from local wireless microphone.

5. Output 3 & 4 are line output, connecting to recording host or other recording devices. The sound signal processed by the microphone and the input audio source such as a computer or a DVD are connected to the recording host for recording.

6. In actual debugging and testing, it may be necessary to reconfigure the parameters of the processor. In this case, the PC and the processor need to be directly connected through the network cable, see Section 4.1.

3.2 Phantom Power Setting

On the right side of the front panel of the audio processor, there are 8 DIP switches to set the opening and closing of the phantom power supply of 8 Mic Inputs channels, as shown below.

The DIP switch downward means that the phantom power supply "on" of the corresponding microphone, and upward means the phantom power supply "off". Normally the phantom power supply remains "on" status.

3.3 Microphone Input Gain Adjustment

The sensitivity of different microphones is different when they are used. Therefore the audio processor provides the Input Gain Adjustment knobs on the left of rear panel for different microphones. There are 8 Input Gain Adjustment knobs, see the picture below.

When you adjust the Gain, you can use a “one” screwdriver to turn up in clockwise direction, and turn down in counterclockwise direction.

3.4 Line Input and Output Gain Adjustment

On the front panel of the audio processor, there are 5 gain adjustment knobs that adjust the gain of the line input and output. The clockwise direction is the gain increase and the counterclockwise direction is the small gain.

4 Software

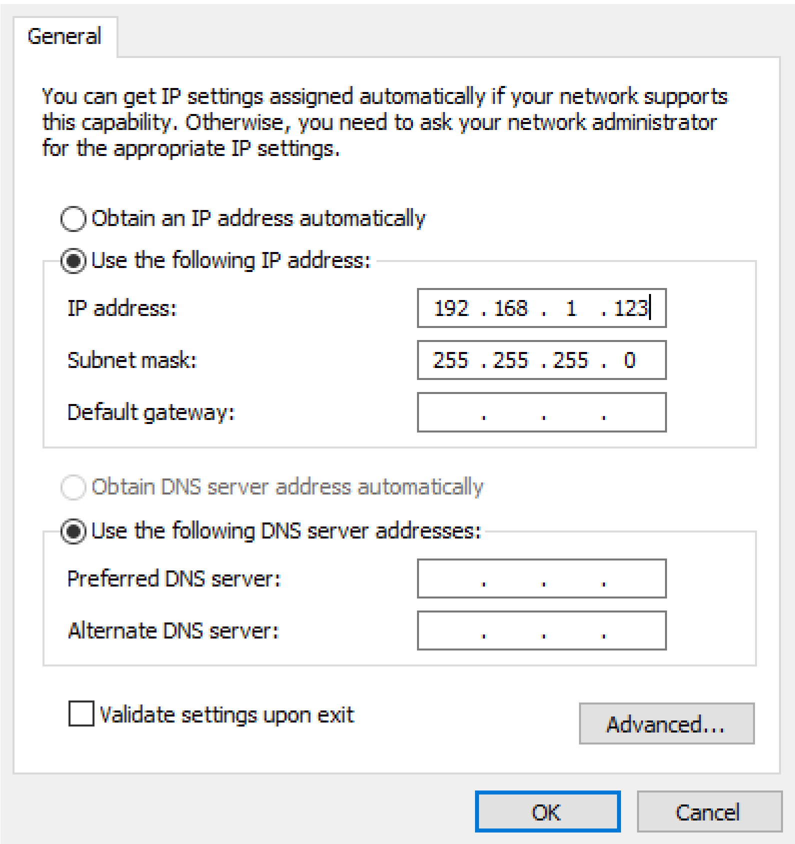
4.1 Connection of PC Client-end and Processor

In the debugging and testing, the parameters of Audio Processor may be reconfigured through PC in an Ethernet cable, the steps are as follows.

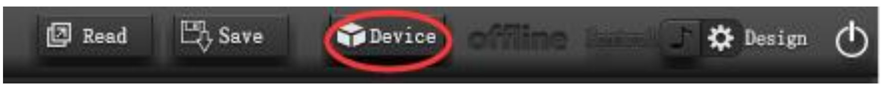
1. Use the Ethernet cable to connect PC and audio processor;

2. Set IP address of PC: 192.168.1.xxx, subnet mask: 255.255.255.0;

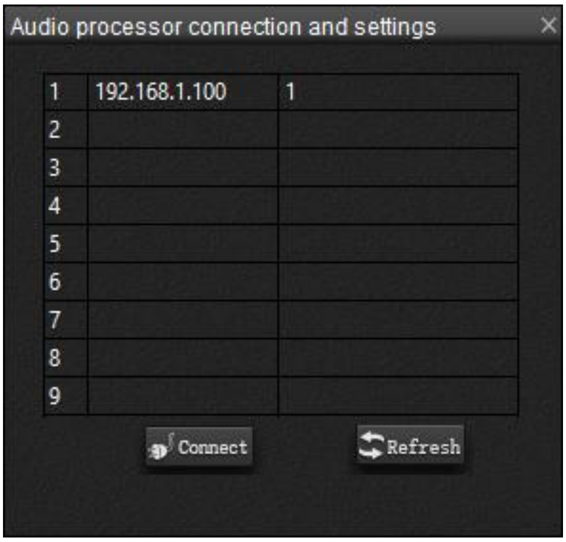
Below for reference.



3. Run configuration software, after it starts, click the "Device" icon on the upper right, and see the red circle in the picture below.



4. Click the processor icon, popup dialog box below, the default IP address of processor: 192.168.1.100



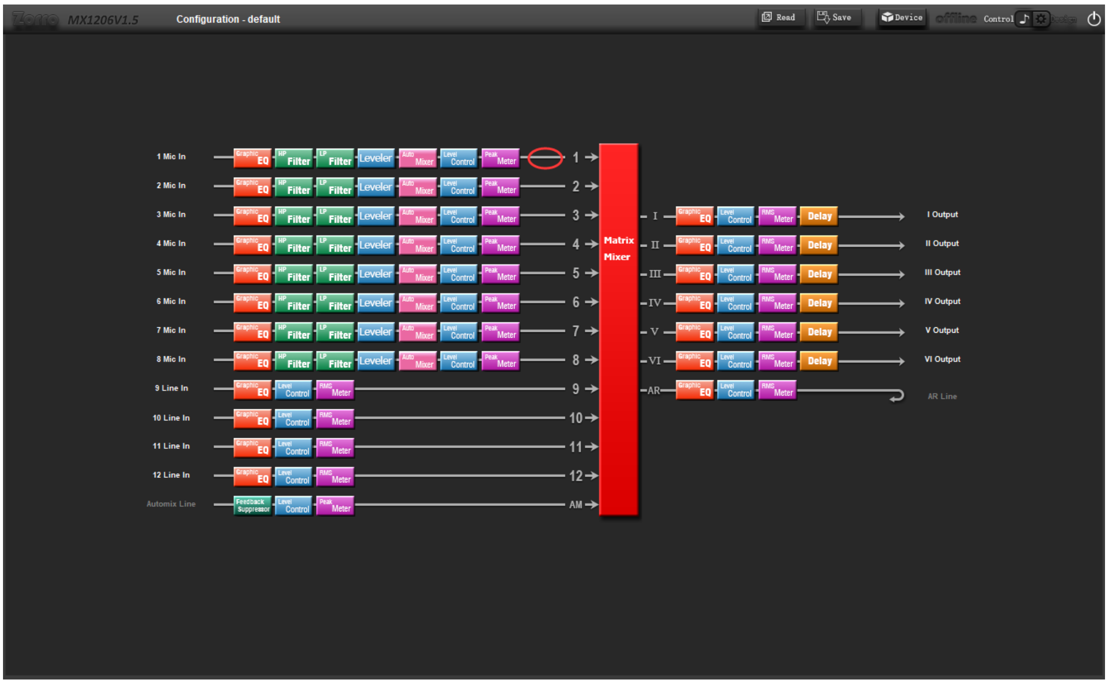
5. Double click the line of IP address, the connection of computer and processor is finished.

4.2 Noise Reduction Setting

The noise suppressor works on 8 microphone input channels respectively. The default value is 12. If you need to change the settings, you can enter into the software to set it up.

1. Click the "Design" icon on the upper right, and see the red circle in the picture below.

2. Then you will open the Design interface as below.



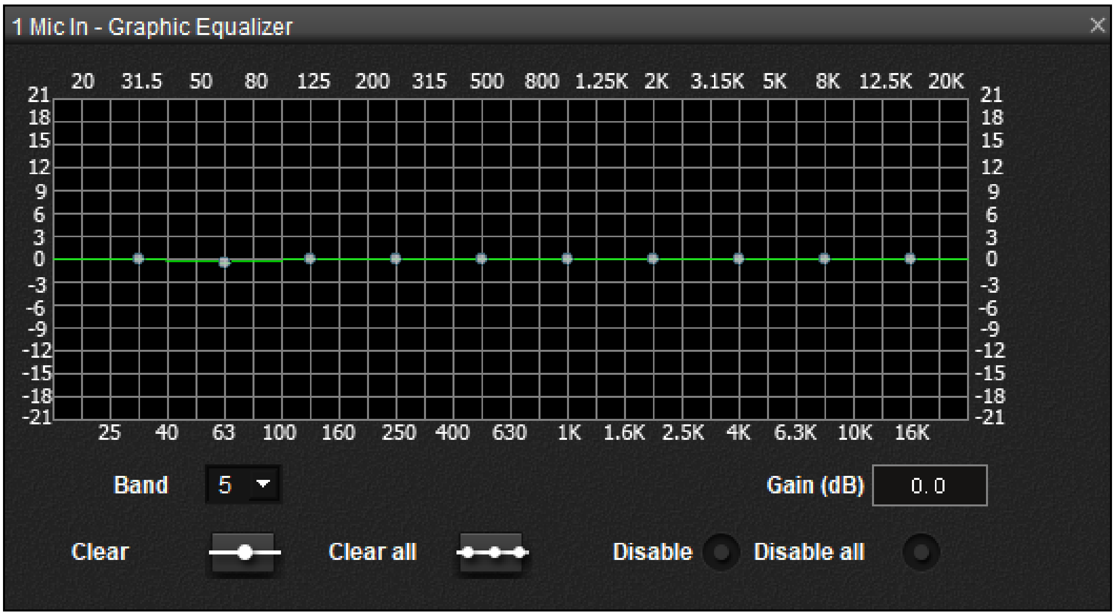
3. Double click the line of the microphone input channel, as shown in the red circle above, it will enter the following sub-interface.

4. The parameters of Noise Reduction have 0, 6~18, total 14 choices. 0 represents NR algorithm closed; 6 represents NR ratio 6dB; 18 represents NR ratio 18dB; 12dB for recommendation, see the red circle above.

4.3 Ten-band Graphic Equalizer Setting

1. Double click the "Graphic EQ" icon.

2. Finish the step 1, enter the next sub-panel below.

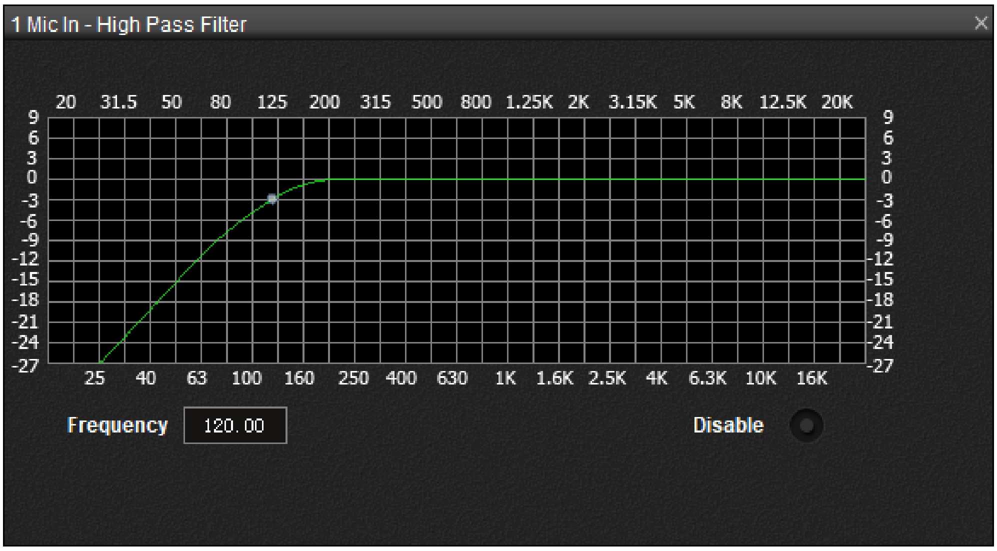


3. Users can choose gain of each frequency band, 0dB by default.

4.4 High-Pass Filter Setting

1. Double click the green icon "HP Filter".

2. Finish the step 1, enter the next sub-panel below.

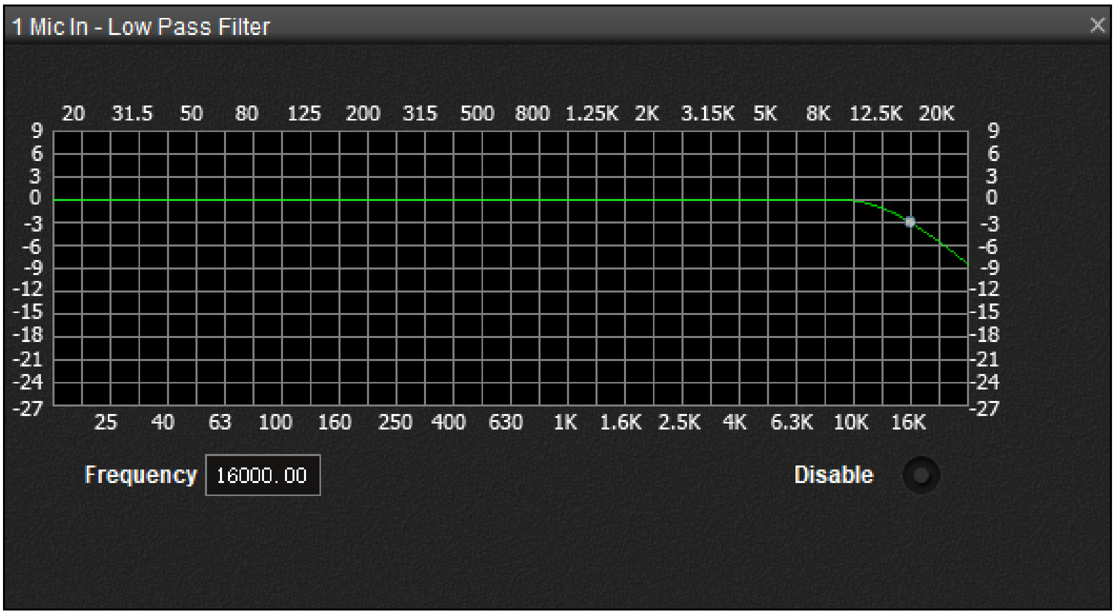


3. Users can choose cut-off frequency of High-Pass Filter, 120Hz for recommendation.

4.5 Low-Pass Filter Setting

1. Double click the green icon "LP Filter".

2. Finish the step 1, enter the next sub-panel below.



3. Users can choose cut-off frequency of Low-Pass Filter, 16000Hz for recommendation.

4.6 Leveler Setting

Leveler is a compressor, of which the compress ratio is 10:1. Leveler is open by default, usually we suggest end user not to change these settings.

4.7 Auto Mixer Setting

The Auto Mixer turns 8-channel Mic Inputs to auto-mixing outputs. This module is strongly suggested to use default parameters, instead of changing settings.

4.8 Level Control Setting

Level control module exists system input-end and output-end, adjusting the volume of system input and output, setting is as below.

1. Double click the "Level Control" icon.

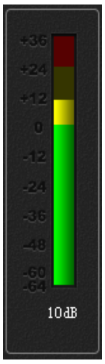
2. The display is below, users can use the slider to vary level, setting ranges, [-100dB，12dB].

Attention: Varying the volume of Microphone input will not change the volume of the Auto-Mixer Line input.

4.9 Meter Display

There are 2 volume displays of input and output level, Peak value and Root-Mean-Square value, RMS. Usually Mic Input uses Peak value meter, Line Input/ Output uses RMS value meter.

Usually, it is reasonable that the value of output volume is in the yellow area, see the picture below.



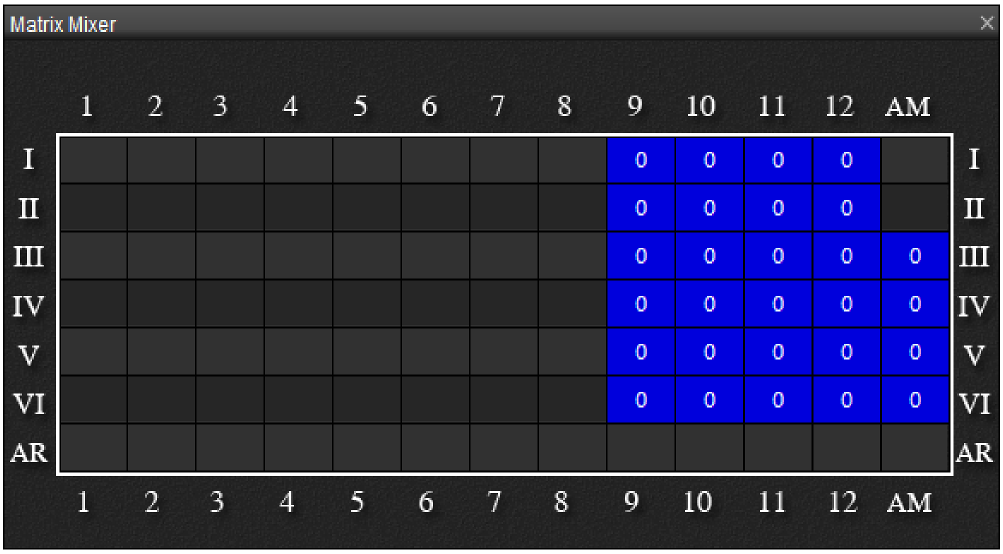
4.10 Matrix Mixer Setting

In the configuration software, the input interface is named by Arabic numbers: 1, 2, 3, 4, …, the output interface is named by Roman numbers: I, II, III, IV,…

Besides, there are two special interfaces in the device, AM and AR. AR is invalid for this model MX1206.

- AM is an input interface of the Auto Mixer, all microphone signals after auto mixing are outputting from this one;
- AR is a reference output interface of Echo Cancellation, receiving the signal for echo cancellation reference.

Usually we use default configuration below.



The detailed instruction is as below,

- Output I and Output II are access to amplifier, playing local background music(Input 9, 10, 11 & 12) .
- Output III and Output IV are access to recording system, meanwhile recording local background music(Input 9, 10, 11 & 12) and local microphone mixing output(AM input).

- Output V and Output VI, the same connection as Output III and Output IV, if needed.

4.11 Delay Setting

There is a delay module in the Line Input channel, users can set the delay, 0ms by default, it is banned usually.

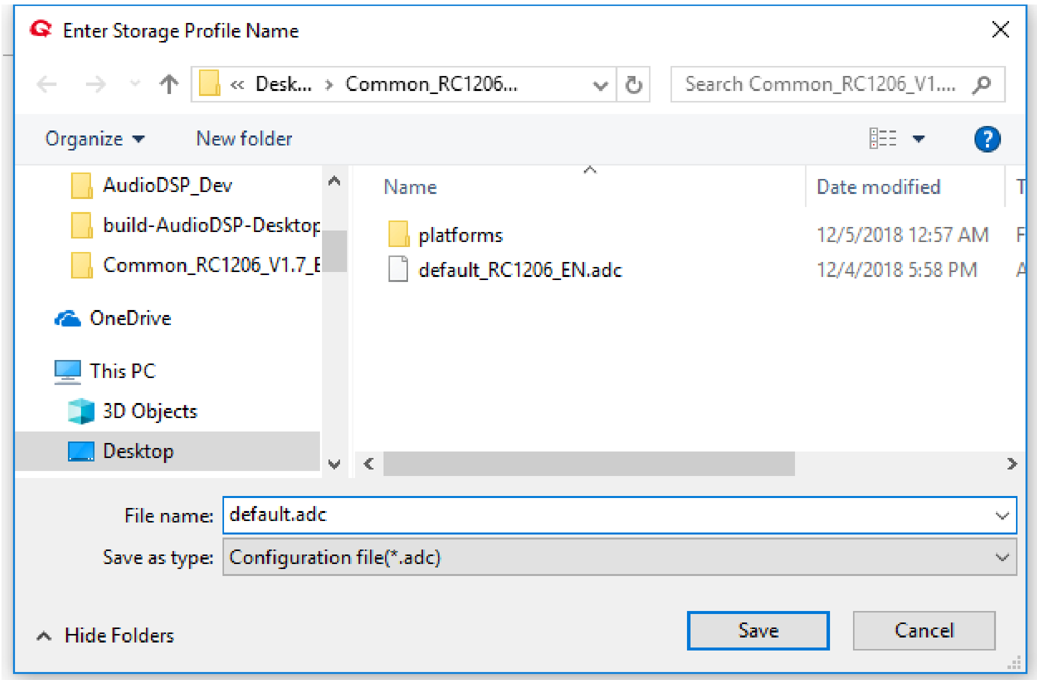
4.12 Save the Scenario Configuration File

Configuration of the client-end and processor keep synchronization when the processor is connected to configuration software client-end. Break the connection, the processor will save the current configuration automatically. If dump and restart, the processor runs in this configuration that is very convenient for debugging and testing.

During the installation, user finishes the debugging of the audio processor, can save configuration file. When this processor works at the same scenario, it can read the configuration file directly, instead of re-debugging every time. The steps are as follows.

1. Click the icon of control panel, see the red circle below,

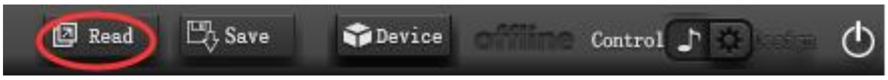
2. Save as xxx.adc configuration file, xxx is user's naming. This handbook names it default.adc, see the picture below.



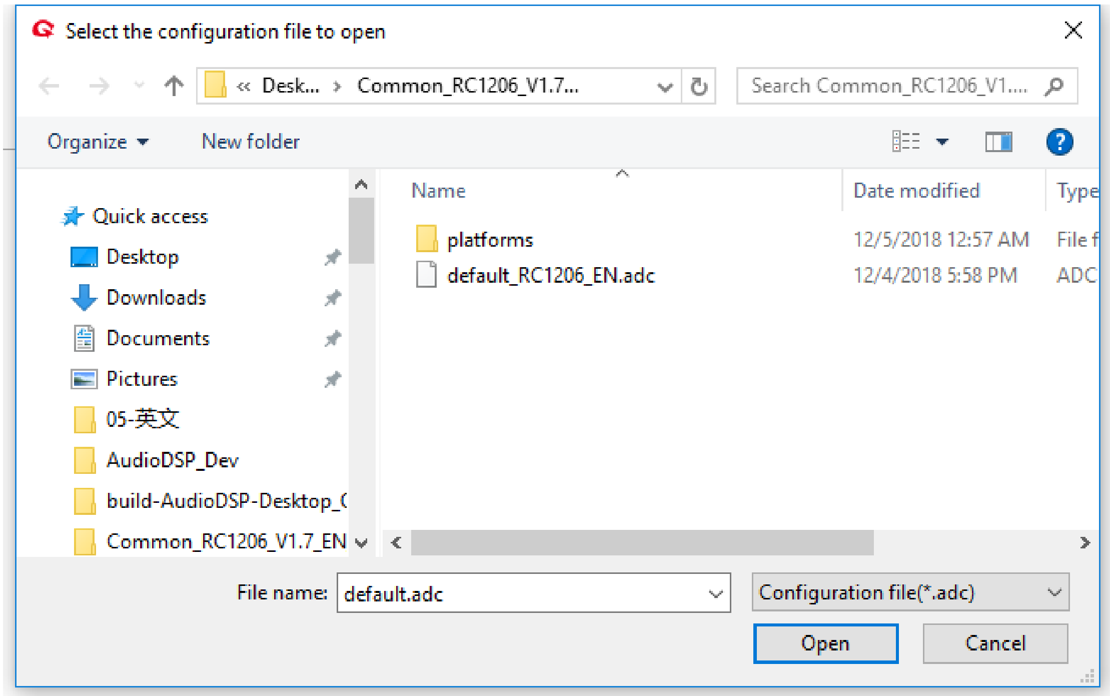
4.13 Read Scenario Configuration File

Installing the audio processor at the same scenario, it can read the configured parameters, speeding up the installation and debugging, the steps are as follows.

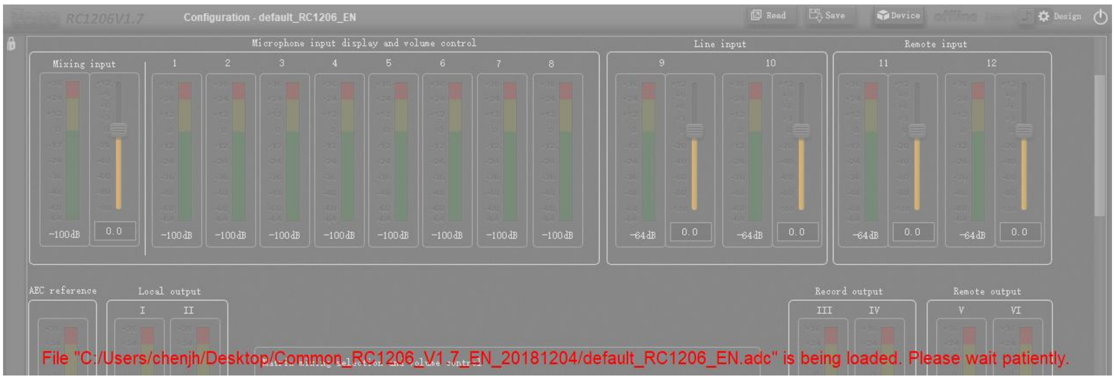
1. Click the "Read" icon on the control panel, see the picture below.



2. Read the xxx.adc configuration file saved before, see the picture below,



3. When the audio processor is reading the configuration file, the panel will display the picture below, waiting for around 10 seconds, and it will complete the parameters configuration.



5 FAQ

5.1 Output without Voice

1. Check the power indicator of processor on or not.

2. Check the RUN indicator slow flashing or not.

3. Check the input/ output interface connection right or not.

4. Check the signal of Mic Input source and Line Input source normal or not.

5. Check Matrix configuration right or not(See section 4.10).

6. Start software to check the meter running or not.

5.2 Current Noise in Output Voice

1. Does audio patch cord make right?

2. Does the audio wire need the shield wire to connect?

3. Does the input signal level oversize?