NEXIS

model : MA0808

8x8 Digital Audio Processor



User Manual

VER 1.1

Thank you for purchasing this product

For optimum performance and safety, please read these instructions carefully before connecting, operating or adjusting this product. Please keep this manual for future reference.

Surge protection device recommended

This product contains sensitive electrical components that may be damaged by electrical spikes, surges, electric shock, lighting strikes, etc. Use of surge protection systems is highly recommended in order to protect and extend the life of your equipment.

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1. Introduction

This digital audio processor is typically used for video conference, distant learning, and telemedicine. It features 8-ch MIC/linear inputs, and 8-ch linear outputs. The product can process audio signals with algorithms, such as, full-band Adaptive Echo Cancellation (AEC), Adaptive Noise Suppression (ANS), Automatic Gain Control (AGC), and Auto Mixer, to output a clear, clean and resonant sound with high Signal-to-Noise ratio. Concise but intelligent, the processor is designed to be applied in scenarios without additional software assistance for debugging. It is ready to use after installation, perfect for project implementation and testing. The product can be applied in a diverse range of installations and applications across industries, such as, smart system integration in small-medium sized conference room, instruction recording and distance teaching in education, court trial recording and virtual court trial in judiciary, surgery recording and video consultation in healthcare service, and command center establishment in governmental projects.

2. Features

- ☆ Provide 8-ch balanced MIC/linear inputs and 8-ch balanced linear outputs
- ☆ Support adaptive feedback suppression function
- ☆ Support the full-band adaptive acoustic echo cancellation technology
- ☆ Dynamic adaptive noise reduction technology is provided to reduce noise with signal level up to 18dB
- ☆ Auto Mixer function is provided to set the order of priority when multiple microphones are input at one time
- ☆ Inclusive of Digital signal processing modules such as Expander, Equalizer, Compressor, Auto Gain Control, Limiter, High Pass Filter, Low Pass Filter and Delay
- ☆ Capable to switch matrix routings
- ☆ Support volume control, meter, scene control, etc.
- ☆ 48V phantom power supply for 8-ch MIC inputs
- ☆ 48KHz sampling rate, 24-bit for A/D or D/A conversion
- ☆ Support 8-ch programmable GPIO function
- ☆ Compatible to run on Win 7 and Win 10, with standard RJ45 interface control
- ☆ Support RS-232 serial commands control

3. Package Contents

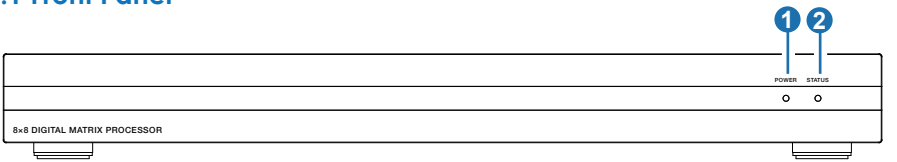
① 1 x Digital Audio Processor
② 1 x AC 100~240V 50/60Hz Power Cord
③ 1 x 3pin-3.81mm Phoenix Connectors (Male)
④ 1 x 10pin-3.81mm Phoenix Connectors (Male)
⑤ 4 x 12pin-3.81mm Phoenix Connectors (Male)
⑥ 2 x Mounting Ears (Wall Mount)
⑦ 8 x Machine Screws (KM3*4)
⑧ 1 x User Manual

4. Specifications

Technical	
Amplitude-frequency (20Hz~20KHZ@+4dBu)	±0.2dB
THD+N (1KHZ@+4dBu)	≤ 0.01%
SNR (linear input)	≥ 90dB
Dynamic Range	≥ 100dB
Channel Level Difference	±0.5dB
Channel Isolation	≥ 80dB
Max Input Level	20dBu
Max MIC Gain	40dB
Input Impedence	20KΩ
Output Impedence	300Ω
Sampling Frequency	48KHZ
A/D and D/A Conversion	24Bit
Phantom Power	+48 VDC
Connection	
Inputs	8 × Balanced MIC/LINE [3-pin phoenix connector] or 4 × Stereo Audio [3-pin phoenix connector]
Outputs	8 × Balanced LINE [3-pin phoenix connector] or 4 × Stereo audio [3-pin phoenix connector]
Controls	1 × LAN [RJ45] 1 × RS-232 [3-pin phoenix connector] 8 × GIPO [10-pin phoenix connector]
Mechanical	
Housing	Metal Enclosure
Color	Black
Dimensions	440mm (W)×250mm (D)×45mm (H)
Weight	3.3kg
Power Supply	AC 100 - 240V 50/60Hz
Power Consumption	9W (Max)
Operating Temperature	0°C ~ 40°C / 32°F ~ 104°F
Storage Temperature	-20°C ~ 60°C / -4°F ~ 140°F
Relative Humidity	20~90% RH (non-condensing)

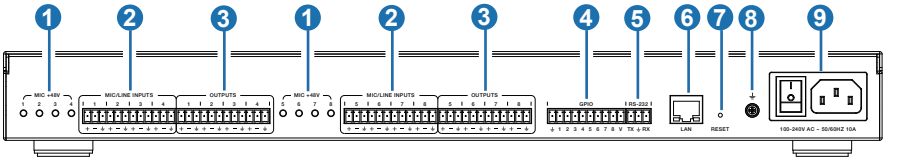
5. Operation Controls and Functions

5.1 Front Panel



No.	Name	Function Description
1	POWER LED	When the device is powered on, the red LED will light on.
2	STATUS LED	When the device runs in normal, the green STATUS LED will flash.

5.2 Rear Panel



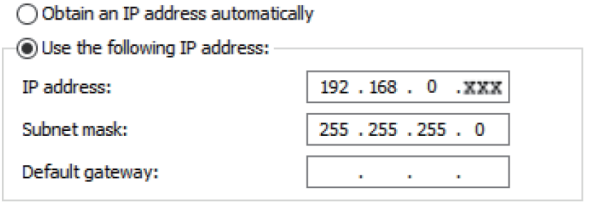
No.	Name	Function Description
1	MIC +48V	8 phantom power supply indicators.
2	MIC/LINE INPUTS	8-ch balanced MIC/linear inputs, used for connecting audio source devices via phoenix connectors.
3	OUTPUT	8-ch balanced linear outputs, used for connecting Amplifier or Speaker via phoenix connectors.
4	GPIO	8-ch programmable GPIO interfaces.
5	RS-232	Connect to PC or control system for transmitting RS-232 control commands.
6	LAN	Standard RJ45 interface for network connection.
7	RESET	Used for reseting the audio processor.
8	GND	Connect outer housing to the ground.
9	Power Switch & Power Port	Power switch and AC 100-240V power input port.

6. Host Computer Control

6.1 Connection

The audio processor admits the control of a host computer in windows system (WIN 7 or WIN 10). Take a cable to connect to the LAN port of the processor with the computer for network connection as following steps instructed.

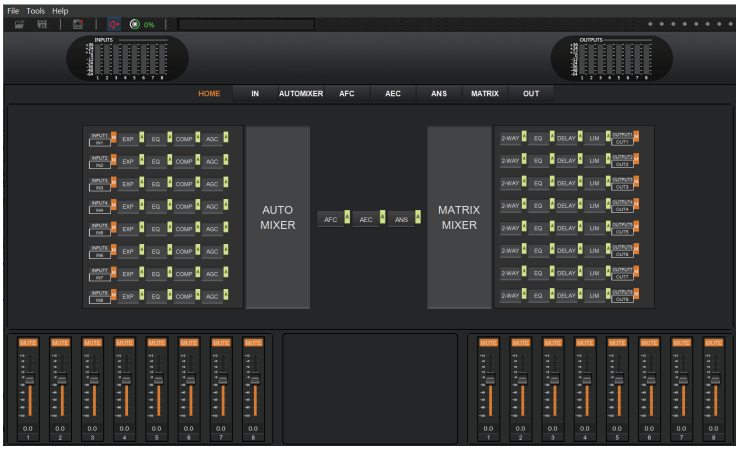
Step 1: Install the “ProAudioProcessor8.exe” application in the host computer and change the computer’s IP address to ensure that IP addresses of the host computer and the processor (Default IP address: 192.168.0.199, Subnet mask: 255.255.255.0) are within the same network segment, as shown in the figure below.



Step 2: Double-click the “ProAudioProcessor8.exe” icon (see the figure below) to run the application in the host computer.



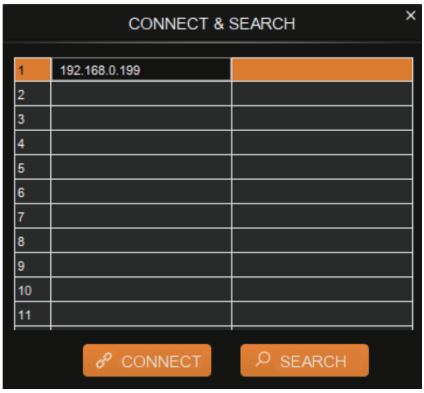
Step 3: When the application is open, the startup page appears and then comes the HOME page, as shown below.



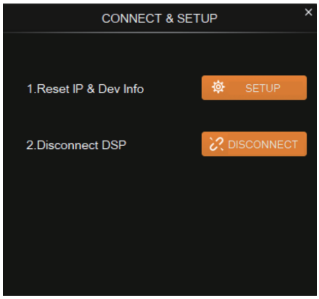
Step 4: Check the Network icon on the tool bar as shown below.



If there is a red X at the top-right corner of this icon as the figure above shown, it means the host computer have not connected with the audio processor yet. In such case, click this icon, then a dialog box with a list of IP addresses of all connected devices will pop out, as the following figure shown.



Step 5: Choose the audio processor's IP address and click CONNECT, then the processor will be connected with the host computer. After connection, the CONNECT & SETUP box (see the figure below) will pop out. You may close it directly if no such operation is needed.



If the audio processor has been connected with the host computer successfully, the Network icon will appear in a way as shown below.



6.2 Volume Meter

The volume meter is typically to denote the signal level of the I/O volume.

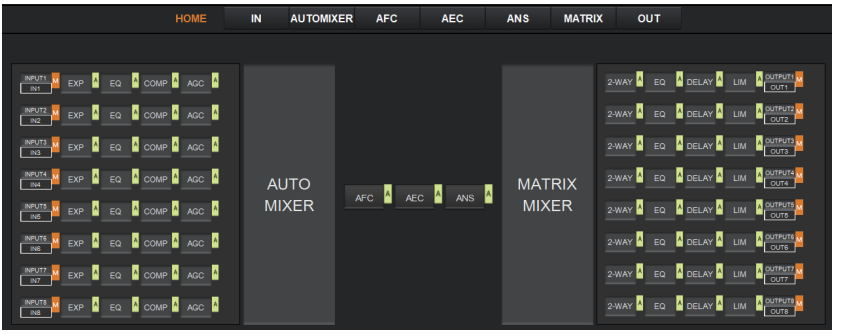


The meter has 3 colors (Red, Yellow and Green) to denote signal levels for the volume of 8-ch inputs and 8-ch outputs, in which the Red denotes that volume may be too high to be limited and the yellow indicates the volume is within a reasonable range.

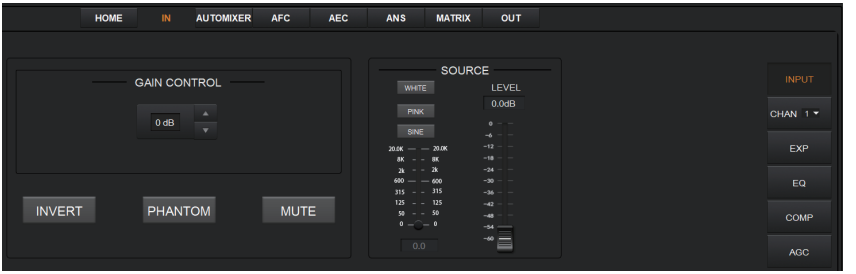
6.3 Input Settings

As shown below, clicking the INPUT module 

on the HOME page or the IN tab on the menu bar, you will enter the INPUT setting page.



The following figure shows the details of the INPUT setting page.



GAIN CONTROL: Indicates the GAIN level of the input signal, which could be adjusted to ensure the gain on the input channel adaptable to the MIC or linear input.
INVERT: When pressed, inverts the polarity of the signal on the input channel.
PHANTOM: When pressed, turns on 48V phantom power for the channel.
MUTE: When pressed, mutes the input channel, equivalent to clicking the letter M at the INPUT module 

on the HOME page.
WHITE: When pressed, generates white noise for signal testing.
PINK: When pressed, generates pink noise for signal testing.
SINE: When pressed, generates sinusoidal signal for testing, the meter below shows the frequency of the sine wave.
FADER: Controls the signal level of the testing signal.

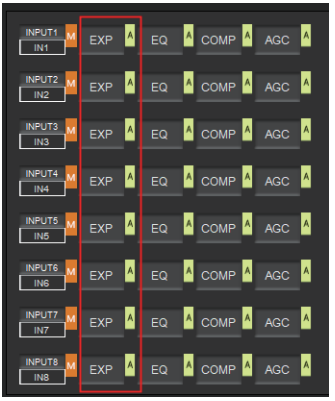
You can set signal parameters of an input channel by entering its CHAN No. on the right panel of the IN page. Audio process modules, such as, EXPANDER, EQUILIZER, COMPRESSOR and AUTO GAIN CONTROL, are available on the right panel for your setting.

6.4 Expander

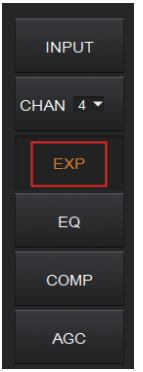
The Expander, which can effectively extend the dynamic range of the input signal, is used to eliminate noise under the threshold level.

As shown below, clicking the EXP module 

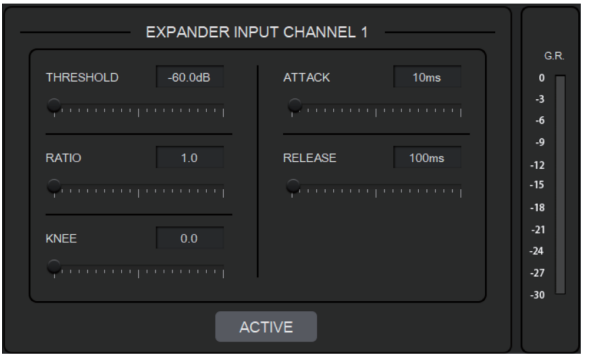
on the HOME page or the EXP button on the right panel of the IN page will give you the option to enter the EXP setting page.



or



The following figure shows complete information to configure the expander module.



THRESHOLD: The level that the signal below it will be identified as noise and attenuated, with a range of [-60.0, 0.0] dB.
RATIO: The compression ratio for signals below threshold, with a scale of [1.0, 20.0]. If the ratio is 2.0, it means the signal below threshold will be reduced to ½ of the original.
KNEE: The curve setting of the inflection point of the Expander, with a scale of [0.0, 20.0]. 0 indicates a hard knee; others indicate a soft knee.
ATTACK: The time required by the Expander to begin the Expander process once a signal drops below threshold. Values are squashed between [1, 500] ms.
RELEASE: The time required by the Expander to stop the Expander process once a signal is over threshold. Values are squashed between [1, 10000] ms.

G.R. (Gain Reduction) : Indicates the amount of gain attenuation of the input signal (in dB) in the Expander process.
ACTIVE: Activates the EXP process. The function is equivalent to clicking the green letter A at the EXP module 

on the HOME page.

Clicking the pair of UP-DOWN arrows of the CHAN No. on the right panel of the IN page will give you an option to select an input channel for the Expander configuration.

6.5 Equalizer

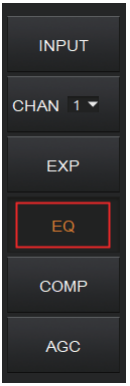
Each I/O channel has an 8-band parametric equalizer for voice processing, with the frequency, gain and bandwidth of each band adjustable.

As shown below, clicking the EQ module 

on the left panel of the HOME page or the EQ button of the **IN** page, you will enter the EQ setting page of the input channel.

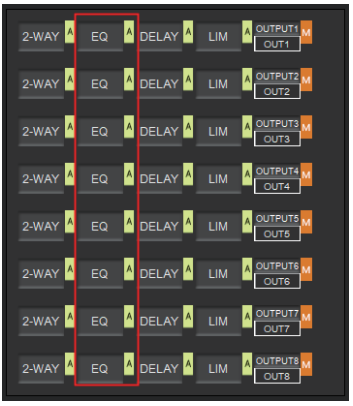


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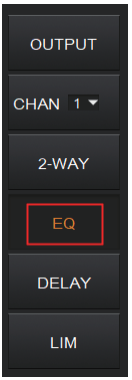


As shown below, click the EQ module 

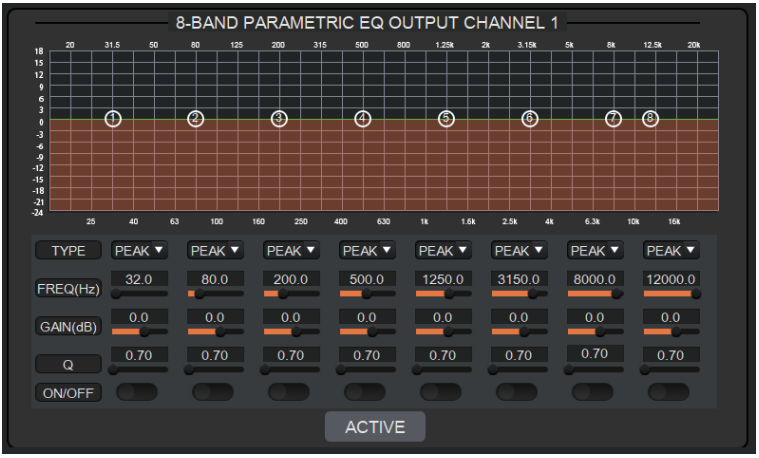
on the right panel of the HOME page or the EQ button of the **OUT** page, you will enter the EQ setting page of the output channel.



or



The figure below shows the setting page of the EQ module.



TYPE: Provides 3 types of band/filter, with High Pass, Low Pass and Peak optional.
FREQUENCY : In High/Low Pass Filter, it means the cut-off frequency of the EQ band; In Peak Filter, it means the center frequency point of the EQ band; with a scale of [20, 20000] Hz.
GAIN: In High/Low Pass Filter, fixed to 0dB; In Peak Filter, squashed between [-24.0, +18.0] dB.
Q FACTOR: value is squashed between [0.02, 50.00].
SWITCH: Each band has a switch. When pressed, the EQ function in this band will be disabled.
ACTIVE: Activates the EQ process. The function is equivalent to clicking the letter A at the EQ module 

on the HOME page.

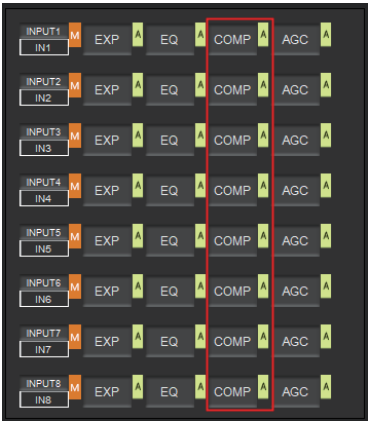
A pair of UP-DOWN arrows on the right panel is provided for you to select an input channel for the Equalizer configuration.

6.6 Compressor

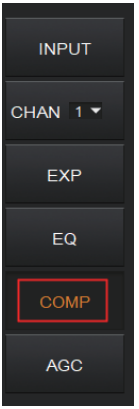
The Compressor is used to reduce the dynamic range of the signal above a user determined threshold so that output sounds could be more solider.

As shown below, click the COMP module 

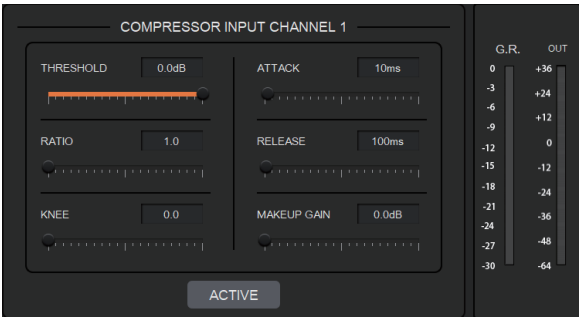
on the HOME page or the COMP button on the right panel of the IN page, you will enter the Compressor setting page.



or



The figure as below shows the setting page of the compressor.



THRESHOLD: The level that the signal above it will be compressed, with a scale of [-60.0, 0.0]dB.
RATIO: The compression ratio for signals with level above threshold, with a scale of [1.0, 20.0]. When the ratio sets to 2.0, it means that the signal with level below threshold will be reduced to ½ of the original.
KNEE: The curve setting of the inflection point of the compressor, with a scale of [0.0, 20.0]. 0 indicates a hard knee; other values for a soft knee.
ATTACK: The time required by the compressor to begin the Compressor process once a signal is over threshold, with a scale of [1, 500] ms.
RELEASE: The time required by the compressor to stop the Compressor process once a signal drops below threshold, with a scale of [1, 10000] ms.
MAKEUP GAIN: Since compressor can reduce the gain level, the Makeup Gain gives you the option to make a compensation for the gain level, with a scale of [-12.0, +18.0] dB.
G.R. (Gain Reduction): Indicates the amount of gain attenuation of the input signal (in dB).
OUT: Meter indicates the output signal level after the Compressor process.
ACTIVE: Activates the COMP process. The function is equivalent to clicking the letter A at the COMP module 

on the HOME page.

A pair of UP-DOWN arrows on the right panel is also provided for you to select any input channel for the Compressor configuration.

6.7 Auto Gain Control

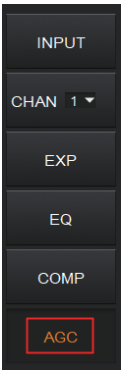
Automatic Gain Control (AGC) intends to control the audio signal within a range so as to extend the distance of pick-up sounds. The AGC process aims to attenuate the signal with excessive high signal level and boost the signal with low signal level up to target level.

As shown below, by clicking the AGC module 

on the HOME page or the AGC button on the right panel of the IN page, you will enter the AGC setting page.



or



The following figure shows the setting page of the AGC module.



TARGET LEVEL: The desired output signal level, with a scale of [-20.0, 0.0] dB.
RATIO: The compression ratio for signals above target level, with a scale of [1.0, 20.0]. If the value sets to 2.0, it means that signals above target level will be amplified in twice of the original. However, not all signals between the threshold and the target level are linearly amplified by this ratio.

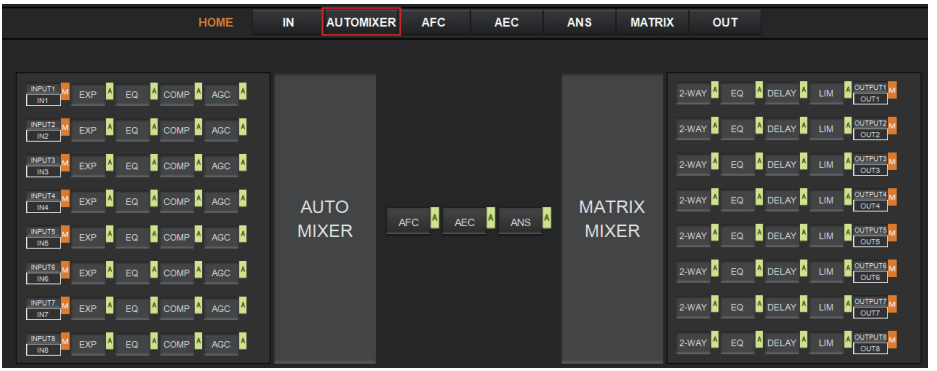
MAX GAIN: It will affect the smooth adjustment of the gain value of the signal between the threshold and the target level, with a scale of [0.0, 12.0] dB.
THRESHOLD: Signals below threshold will remain unchanged while those between threshold and target level will be amplified (at a ratio determined by the TARGET LEVEL, RATIO and MAX GAIN). In order to ensure the work of the AGC process, the Target Level must be set higher than Threshold. The threshold is recommended to set with a range of [-60.0, 0.0]dB.
IN: Meter indicates the detected signal level before the AGC process.
OUT: Meter indicates the output signal level after the AGC process.
ACTIVE: Activates the AGC process. The function is equivalent to clicking the green letter A at the AGC module 

on the HOME page.
A pair of up-down arrows on the right panel is available for you to select an input channel for the AGC configuration.

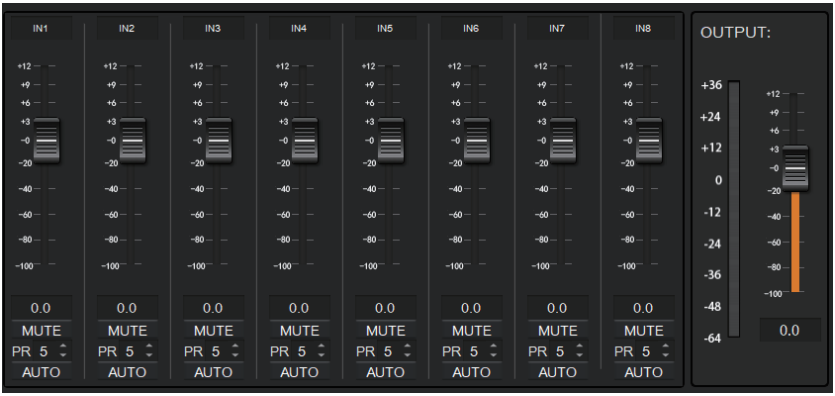
6.8 Auto Mixer

The AUTO MIXER is built to set the order of priority for MIC/linear input when 8 MIC/LINE ports are connected at one time.

As shown below, by clicking the AUTO MIXER section on the HOME page or the AUTOMIXER tab on the menu bar, you will enter the AUTO MIXER setting page.



The figure as below shows the setting page of the Auto Mixer module.



FADER: Each MIC input channel has a fader to adjust its signal level. It controls the mixed volume on the input channel.

MUTE: Each MIC input channel has a MUTE button. Clicking it will enable you to mute the channel.

PR (Priority): Sets the priority of the MIC input, 0 = lowest and 10 = highest priority. Channel of higher-priority will override the one of lower-priority to be prioritized in the auto mixer's algorithm. If two or more channels are at the same priority, the channel for the main speaker will be prioritized in the AUTO MIXER process.

AUTO: When pressed, the input MIC will be added to the auto mixing system, otherwise, it won't be auto-mixed.

METER: Indicates the output volume after the AUTO MIXER process.

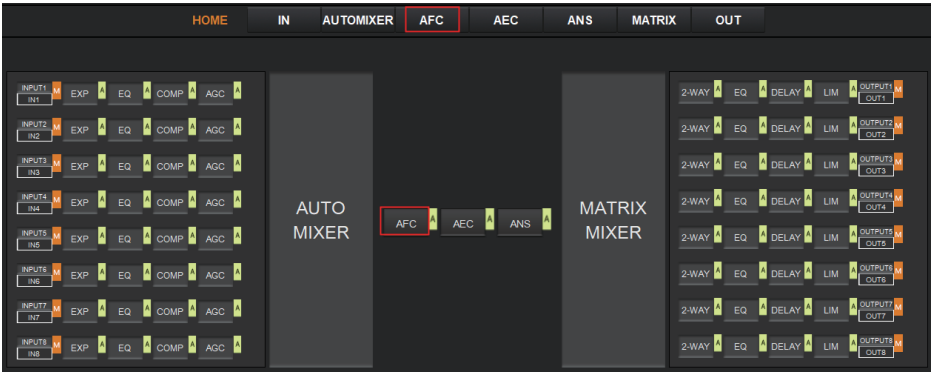
OUT: Fader to control the output volume after the AUTO MIXER process.

6.9 Adaptive Feedback Cancellation

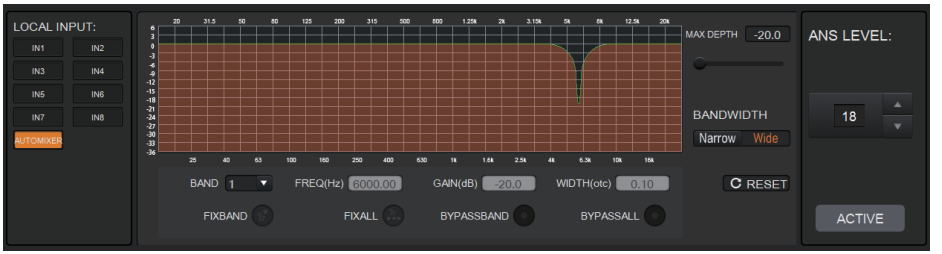
The Adaptive Feedback Cancellation (AFC, also known as the Feed Back Suppression) module can detect and suppress the frequency point of the acoustic feedback in the audio field. The AFC module allows you to detect and suppress the frequency points of the acoustic feedback for 8 bands. Parameters on this module are open to be configured.

As shown below, by clicking the AFC module 

on the HOME page or the AFC tab on the menu bar, you will enter the setting page of the AFC module.



The following figure shows the AFC setting page in detail.



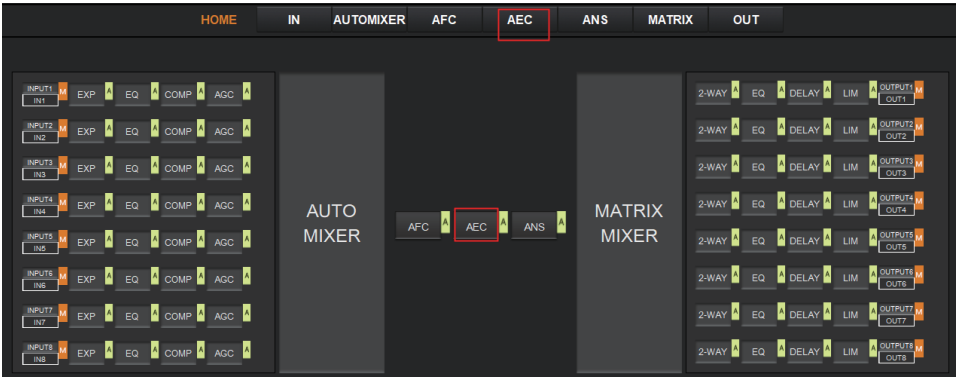
MAX DEPTH: The maximum amount of gain that the module allows to reduce, with a scale of [-20.0, 0.0] dB.
BANDWIDTH: The bandwidth of the filter, with Narrow and Wide optional.
RESET: When pressed, the filter will go back to initial settings and the module will re-detect the frequency point of the acoustic feedback.
FIXBAND: When pressed, fixed values of parameters, such as,Frequency, Gain and Width will be applied to configure the filter in a fixed band. You are allowed to design the filter by setting its parameters.
FIXALL: When pressed, all bands will be processed in a fixed mode or in USER-EDITING mode.
BYPASSBAND: When pressed, the filter of the related band will be disabled.
BYPASSALL: When pressed, the AFC module will be disabled, equivalent to pressing the green letter A at the AFC module on the HOME page.
ANS LEVEL: The Adaptive Noise Suppression (ANS) module is provided to suppress noises with signal level up to 18dB. Clicking the ACTIVE button will give you the option to activate this function.

6.10 Adaptive Echo Cancellation

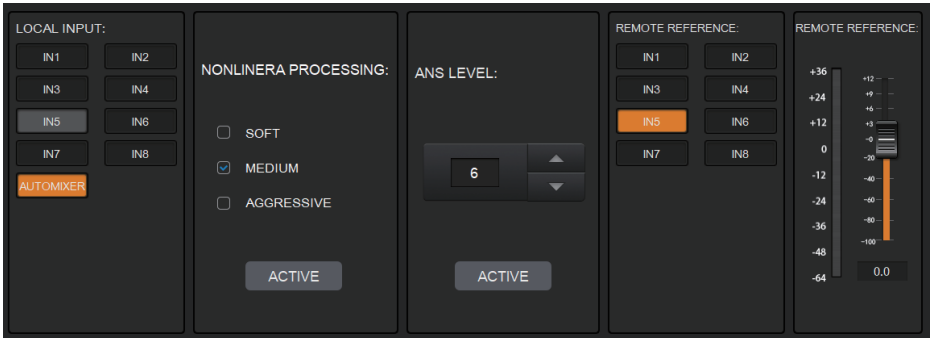
The 48kHz full-band Adaptive Echo Cancellation (AEC) algorithm is set to eliminate echoes coming from the far end in a remote video conference.

As shown below, clicking the AEC module 

on the HOME page or the AEC tab on the menu bar will allow you to enter the AEC configuration page.



The following figure shows the setting page of the AEC module.



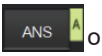
LOCAL INPUT: Shows source signals that are available to have access to the AEC process. AUTOMIXER refers to mixed signals after the AUTOMIXER process.
NONLINEAR PROCESSING: Determines the audio effect for both ends of the video conference, with soft, medium and aggressive optional.
ACTIVE: Activates the AEC process. The function is equivalent to clicking the green letter A at the AEC module on the HOME page.
ANS LEVEL: The Adaptive Noise Suppression (ANS) function is provided to suppress noises with signal level up to 18dB. Pressing the ACTIVE button, the function will be enabled.
REMOTE REFERENCE: Indicates signals that are available for the AEC algorithm to learn how to eliminate analogous echoes in practice.

As shown in the figure above, we choose IN5 as reference signals, in such case, we need to connect the IN5 port of the processor to the terminal of the video conference. As the reference signal after the AEC algorithm will be send to the far end by the terminal, we also need to configure selected input channels in the matrix routing. The figure as below gives an example of configuring the matrix routing.

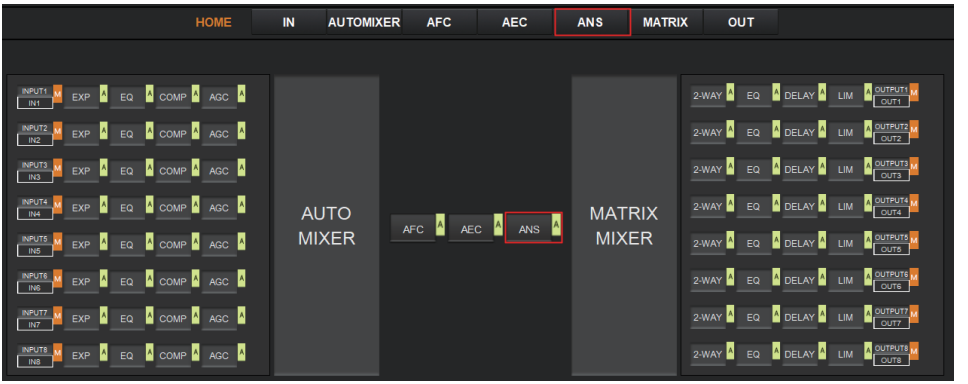
CLEAR	IN1	IN2	IN3	IN4	IN5	IN6	IN7	IN8	AM	AFC	AEC	ANS
OUT1	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT2	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT3	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT4	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT5	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT6	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT7	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT8	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0

6.11 Adaptive Noise Suppression

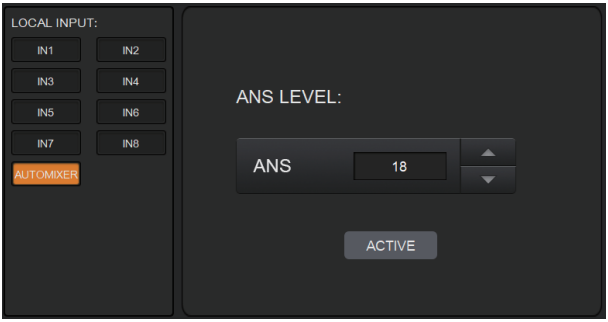
The 48kHZ full-band Adaptive Noise Suppression (ANS) algorithm can effectively increase the signal to noise (S/N) ratio and suppress noises from the blower or air conditioner.

As shown below, by pressing the ANS module 

on the HOME page or the ANS tab on the menu bar, you will enter the ANS configuration page.



The following figure shows the details of the ANS setting page.

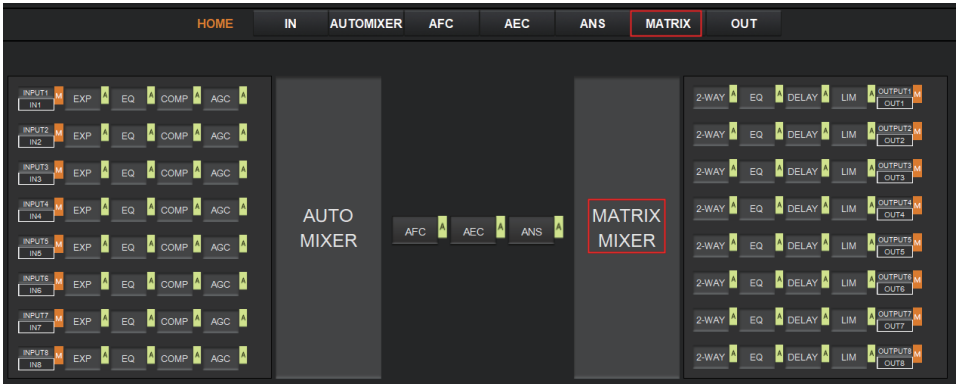


LOCAL INPUT: Shows source signals that are available to have access to the ANS process. AUTOMIXER refers to mixed signals after the AUTOMIXER process.
ANS LEVEL: The ANS module provides 12 noise suppression levels, ranging from 6dB to 18dB.
ACTIVE: Activates the ANS process. The function is equivalent to clicking the green letter A at the ANS module on the HOME page.

6.12 Matrix Mixer

The Matrix Mixer allows you to free choose audio route by matrix mode.

As shown below, clicking the MATRIX MIXER section on the HOME page or the MATRIX tab on the menu bar will enable you to enter the MATRIX MIXER configuration page.



The following figure shows the configuration page in detail.

CLEAR	IN1	IN2	IN3	IN4	IN5	IN6	IN7	IN8	AM	AFC	AEC	ANS
OUT1	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT2	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT3	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT4	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT5	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT6	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT7	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT8	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0

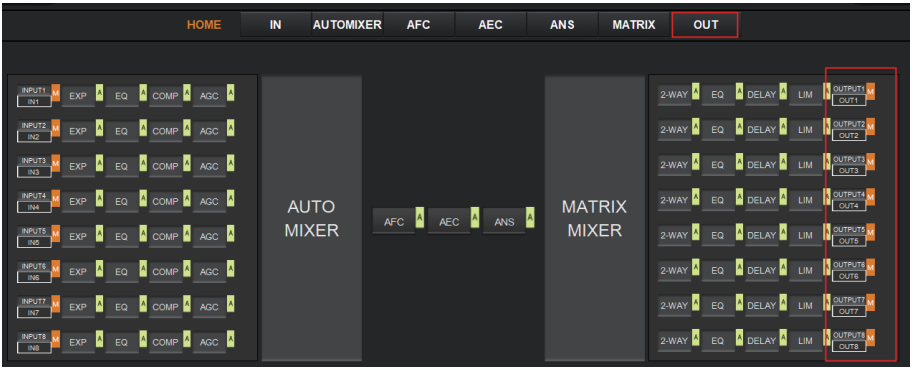
As shown above, the horizontal line stands for output channels and the vertical for input channels (AM, AFC, AEC and ANS exceptable).

AM: Indicates the output signal after the ATUO MIXER process.
AFC: Indicates the output signal after the AFC process.
AEC: Indicates the output signal after the AEC process.
ANS: Indicates the output signal after the ANS process.
Fader: Each channel has a fader to control the output volume.

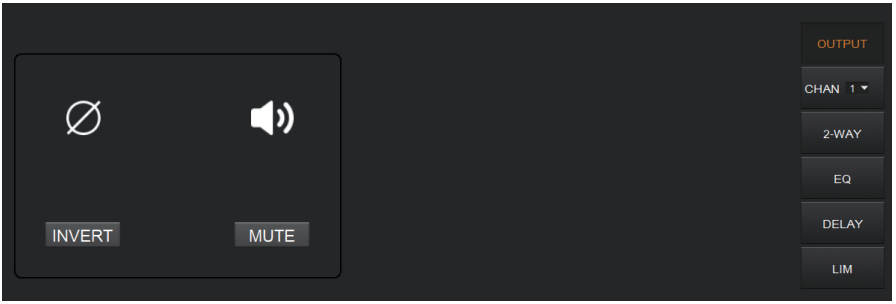
6.13 Output Settings

As shown below, clicking the OUTPUT module 

on the HOME page or the OUT tab on the menu bar will allow you to enter the OUTPUT setting page.



The figure as below shows the details of the setting page.



INVERT: When pressed, inverts the polarity of the signal on the channel.
MUTE: When pressed, mutes the output channel. This function is equivalent to press the letter M at the output module on the HOME page.

On the right panel of the OUT page, modules such as 2-way crossover, equalizer, delay and limiter are also available for you to set.



There is a trumpet icon on the tool bar which enables you to mute all analog output channels (excluding Dante channels) at one time (see the figure above). If it is in red, it indicates that all output channels are mute; otherwise, it will be green.

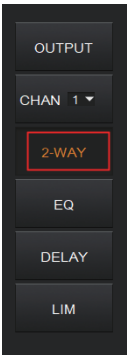
6.14 Two-way Crossover

The two-way crossover consists of a High Pass Filter and a Low Pass Filter, which can divide the input signal up into 2 discreet audio bands (Low and High).

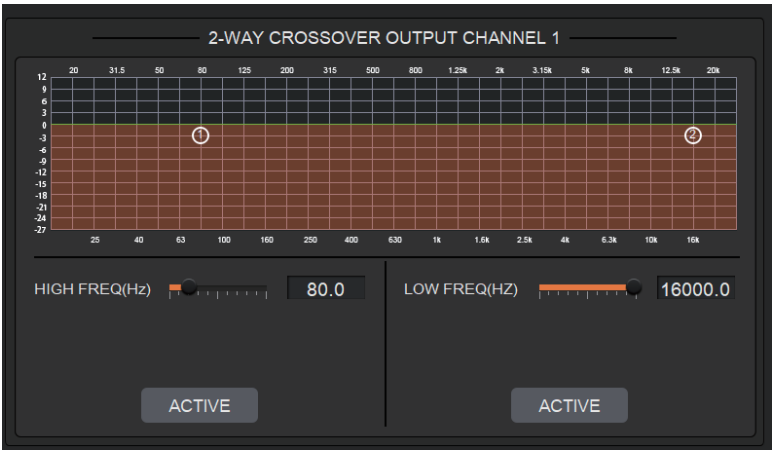
As shown below, pressing the 2-WAY module  on the HOME page or the 2-WAY button on the right panel of the OUT page, you will enter the configuration page.



or



The figure as below shows the setting page.



ACTIVE: When pressed, you will activate/inactivate the frequency filter (High Pass or Low Pass). The activation can be realized by clicking the green letter A at the 2-WAY module 

on the HOME page. The difference is that clicking the letter A will activate/inactivate the High Pass Filter and the Low Pass Filter at the same time while clicking an ACTIVE button on the setting page will only activate/inactivate the corresponding frequency filter.
FREQUENCY: Indicates the cut-off frequency of the filter.
TYPE: Butterworth filter by default.
SLOPE: 6dB/oct by default.

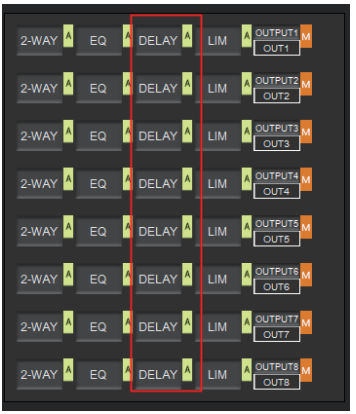
A pair of UP-DOWN arrows on the right panel is available for you to select an output channel for the Two-way Crossover configuration.

6.15 Delay

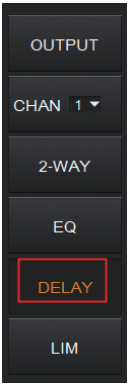
The Delay module is typically used to delay the output speed of audio signals, which can be applied to meet the purpose of Lip Sync in a video conference.

By clicking the DELAY module 

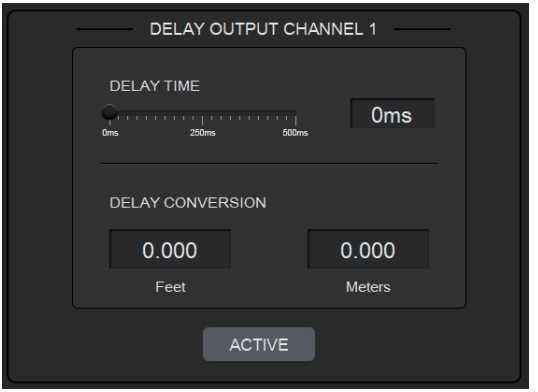
on the HOME page or the DLY button on the OUT page, you will enter the configuration page of the Delay module.

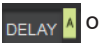


or



The figure as below shows the setting page of this function.



DELAY TIME: Adjusts the delay time of the output audio signal, measured in milliseconds.
DELAY CONVERSION: Displays the amount of delay measured in both Feet and Meters.
ACTIVE: When pressed, you will activate or inactivate the Delay module. This function can also be realized by clicking the green letter A at the DELAY module 

on the HOME page.

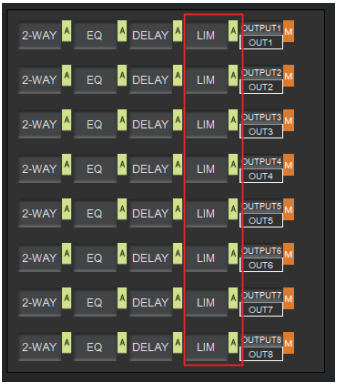
A pair of UP-DOWN arrows on the right panel is also provided for you to select an output channel for the Delay configuration.

6.16 Limiter

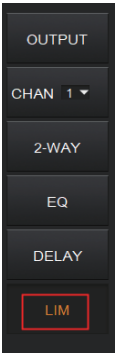
A Limiter can reduce the dynamic range of signals by strictly limiting the maximum output level so as to ensure that the connected amplifier won't be overloaded because of signals with high signal level.

By clicking the LIM module on the HOME page 

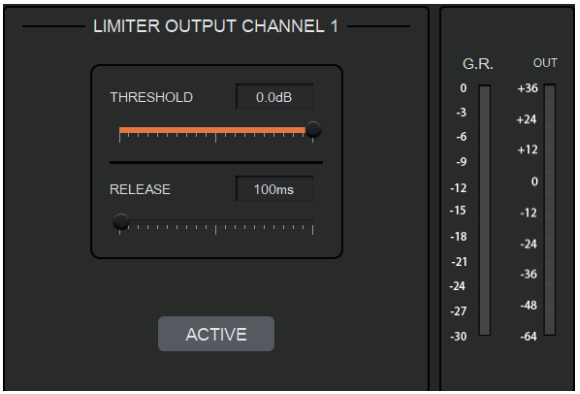
or the LIMITER button on the OUT page, you will enter the LIMITER configuration page.



or



The figure as below shows the setting page of this module.

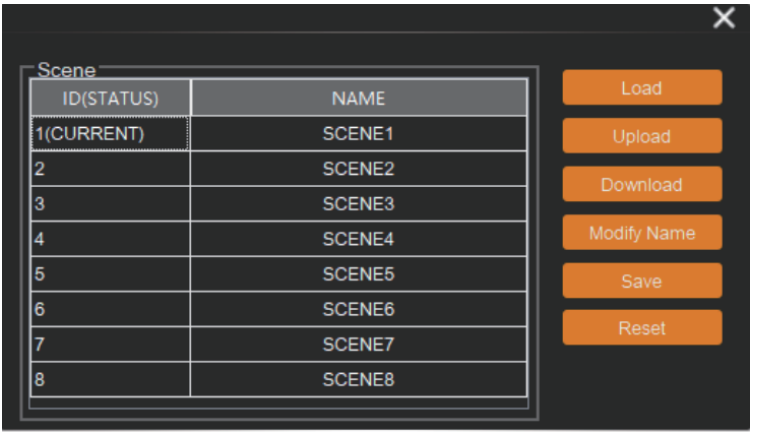


THRESHOLD: Shows that level signals above threshold will be attenuated, with a scale of [-60.0, 0.0] dB.
RELEASE: The time required by the Limiter to stop the LIMITER process once a signal is below threshold, with a scale of [1, 1000] ms.
G.R. (Gain Reduction): Shows the amount of gain attenuation of the input signal, measured in dB.
OUT: Meter indicates output signal level after the LIMITER process.
ACTIVE: When pressed, you will activate/inactivate the function of the Limiter module. The function is equivalent to clicking the green letter A at the LIM module on the HOME page.

You can also click the UP/DOWN arrow on the right panel of the setting page to select an output channel for the LIMITER configuration.

6.17 Scene Management

Up to 8 scenes are available to be managed in this audio processor. By clicking “Tools”> “Scene” on the menu bar, you will enter the setting page as shown below.



Load: When pressed, the scene saved in this processor will be exported.
Upload: When pressed, the configuration file saved in the host computer will be sent to the processor. The UPLOAD function is equivalent to clicking the OPEN icon on the tool bar as shown below.



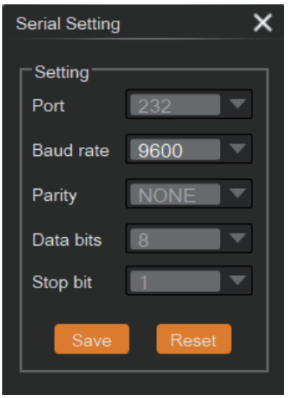
The processor can only read XML configuration files.
Download: When pressed, the present scene will be downloaded to the host computer. The DOWNLOAD function is equivalent to clicking the SAVE icon on the tool bar as shown below.



By default, the file is saved in .xml file format.
Modify Name: When pressed, the scene name displayed on the window will be changed.
Save: When pressed, the scene currently configured will be saved.
Reset: When pressed, the processor will go back to the factory settings.

6.18 Serial Settings

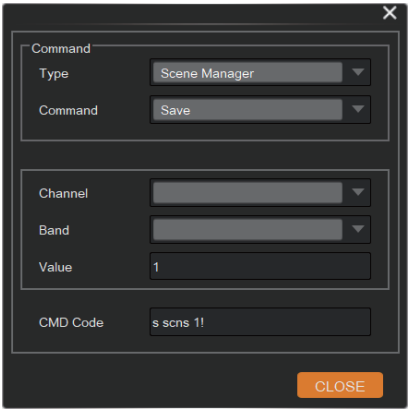
You can configure the RS-232 serial parameters by clicking “Tools” > “Serial” on the tool bar to enter its setting page as shown below.



Baud Rate: 9600 by default, open to change.
Reset: Clicking it to restore the system default settings.

6.19 Control Commands

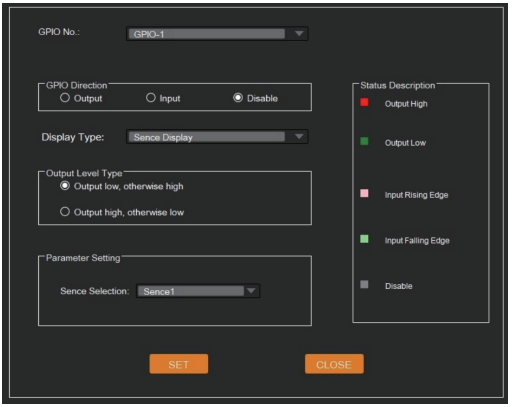
The audio processor is controllable via Control Commands. As shown below, by clicking “Tools” > “Command” on the tool bar, you will enter the setting page.



All RS-232 control commands, such as, Scene Management, I/O Control, Modules’ Configuration, and Device Information, can be found in the page. The command code will be listed at the bottom of this page once you have selected the Type, Command, Channel, Band from the drop-down box and entered its value.

6.20 GPIO Setting

Up to 8 programmable GPIO function are available to be set in this audio processor. By clicking “Tools”> “GPIO” on the menu bar, you will enter the setting page as shown below.



You can set the GPIO function as needed, see details at GPIO Setting page. You can also check GPIO status by GPIO LED at the top-right corner of Host page.

6.21 Reset Setting

Press and hold the RESET button on the back panel for 10 seconds until the STATUS LED on the front panel flashes slowly, then release the button, the audio processor will reboot automatically and restore to default settings (including IP address, configuration information etc.).

7. TCP/IP Control Command

The product supports TCP/IP command control. Connect the LAN port of the product to a PC, then open a TCP/IP Command tool on PC to send TCP/IP commands to control the product. The TCP/IP command list about the product is shown as below.

TCP/IP Commands	
Service Port Number: 3433	
The control command ends with “\n”, and no symbols in the form of "\n" can appear in the middle of the command. Channel number: 0~7, corresponding to the Mic/Line input channels 1~8 of the device. Channel number: 96 indicates AM, which is the audio signal after auto mixer process. Channel number: 97 indicates AFC, which is the audio signal processed by the AFC module. Channel number: 98 indicates AEC, which is the audio signal processed by the AEC module. Channel number: 99 indicates ANS, which is the audio signal processed by the ANS module. Channel number: 100~107, corresponding to the analog output channels 1~8 of the device. Channel number: 196 indicates the reference signal of AEC.	
HChanl (Channel Command)	
<HChanl></HChanl> It is used to control the mute, phantom power, inversion and analog gain adjustment of the channel.	
Field	**no:** Channel number, no=0~7 or 100~107. **mu:** Mute switch, mu="0" (unmute), mu="1" (mute). **phpow:** Phantom power switch， phpow="0" (phantom power on), phpow="1" (phantom power off). **invrt:** Invert function switch, invrt ="0"(non-invert)， invrt ="1" (invert). **><:** Gain setting. The value range is [-12, +40]dB.
Example	<HChanl no="0" mu="0" phpow="1" invrt="1">1</HChanl> Indicates: Channel 0, unmute, phantom power on, invert, analog gain amplified 1dB.
HNg (Signal Generator)	
<HNg></HNg> It is used to generate white noise, pink noise and sinusoidal signal of arbitrary frequency.	
Field	**no:** Channel number, no=0~7. **opt:** Enable/disable switch, opt="0" (disable), opt="1" (enable). **type:** Signal type, type="0" (white noise), type="1" (pink noise), type="2" (sinusoidal signal) **frq:** The frequency of sinusoidal signal, frq =20~20000Hz. **><:** Level value. The value range is [- 60, 0]dB.
Example	<HNg no="3" opt="1" type="2" frq="1000">-24</HNg> Indicates: Enable the signal generator function module of the input channel 3, and generate 1000Hz sinusoidal signal with -24dB.
HLevCtrl (Level Control)	
<HLevCtrl></HLevCtrl> It is used to control the digital level of the corresponding audio channel.	
Field	**no:** Channel number, no=0~7 or 100~107. **><:** Level value. The value range is [-100, 12]dB.
Example	<HLevCtrl no="5">0</HLevCtrl> Indicates: Set the level value of the input channel 5 to be 0dB.

HAmx (Auto Mixer)	
<HAmx><In >…</In></HAmx> It is used to control the auto mixer function.	
Field	**no:** Channel number, no=0~7. **mu:** Mute switch, mu="0" (unmute), mu="1" (mute). **pry:** Priority, pry=0~10， A larger value indicates a higher priority. **><:** Level value of the auto mixer channel. The value range is [-100, 12]dB.
Example	<HAmx> <In no="1" mu="0" pry="5">0</In> <In no="3" mu="0" pry="5">0</In> <In no="6" mu="1" pry="5">0</In> <In no="7" mu="0" pry="5">0</In> </HAmx> Indicates: Add input channels 1, 3, 6, 7 to the auto mixer module for auto mixing algorithm processing.
HAns (Adaptive Noise Suppression)	
<HAns><In>…</In></HAns> It is used to control the ANS (Adaptive Noise Suppression) module.	
Field	**ans:** ANS switch or suppression level, ans=0 or 6~18， ans="0" (disable ANS), ans="6"~"18" (noise suppression level). **no:** Channel number, no=0~7 or 96.
Example	<HAns ans="15"> <In no="1" ></In> <In no="3"></In> <In no="6" ></In> <In no="96"></In> </HAns> Indicates: Select input channels 1, 3, 6 and AM mixing signals to enter the noise suppression module, and suppress 15dB stationary noise.
HAec (Adaptive Echo Cancellation)	
<HAec><In>…</In></HAec> It is used to control the parameter settings for AEC module, such as local inputs, module switch and non-linear processing (NLP).	
Field	**aec:** AEC switch or NLP parameter setting, aec="0" (Disable AEC), aec="1" (Enable AEC, and set the NLP level to mild), aec="2" (Enable AEC, and set the NLP level to moderate), aec="3" (Enable AEC, and set the NLP level to severe). **ans:** ANS switch or noise suppression level for the AEC module, ans=0 or 6~18, ans="0" (Disable ANS), ans="6"~"18" (ANS level). **no:** Channel number, no=0~7 or 96.
Example	<HAec aec="2" ans="15"> <In no="1" ></In> <In no="3"></In> <In no="6" ></In> <In no="96"></In> </HAec> Indicates: Select input channels 1, 3, 6 and AM mixing signals to enter the Echo Cancellation module, set the NLP no-linear parameters of the residual echo to moderate and suppress 15dB noise at the same time.

HAhc (Adaptive Feedback Cancellation)	
<HAhc><In>…</In></HAhc> It is used to control the input source selection and other functions of the AFC module.	
Field	**ahc:** Reserved **ans:** ANS switch or noise suppression level for the AFC module, ans=0 or 6~18, ans="0" (Disable ANS), ans="6"~"18" (noise suppression level). **no:** Channel number, no=0~7 or 96.
Example	<HAhc ahc="1" ans="15"> <In no="1" ></In> <In no="3"></In> <In no="6" ></In> <In no="96"></In> </HAhc> Indicates: Select input channels 1, 3, 6 and AM mixing signal to enter the Feedback Cancellation module, and suppress 15dB noise at the same time.
HMmx (Matrix Mixer)	
<HMmx><In>…</In></HMmx> Matrix mixer can rout any input audio signal to any output.	
Field	**no** (in <HMmx></HMmx>)**:** Channel number, no=100~107. **no** (in <In></In>)**:** Channel number, no=0~7 or 96, 97, 98, 99. **><** (in <In></In>)**:** Level value. The value range is [- 100, 12] dB.
Example	<HMmx no="100"> <In no="0">0</In> <In no="1">0</In> <In no="6">0</In> </HMmx> Indicates: Mixing input channel 0, 1, 6 to output channel 100 (i.e. the analog audio output channel 1) .
HHpf (High Pass Filter)	
<HHpf ></HHpf> The high pass filter is mainly used to remove low frequency noise.	
Field	**no:** Channel number, no=100~107. **opt:** Enable/disable switch, opt="0" (disable), opt="1" (enable). **><:** Cut-off frequency. The value range is [20, 20000]Hz.
Example	<HHpf no="104" opt="1">300</HHpf> Indicates: Enable the high pass filter on output channel 104 (i.e. the analog output channel 5) with a cut-off frequency of 300Hz.
HLpf (Low Pass Filter)	
<HLpf></HLpf> It is used to control the The low pass filter is mainly used to remove high frequency noise.	
Field	**no:** Channel number, no=100~107. **opt:** Enable/disable switch, opt="0" (disable), opt="1" (enable). **><:** Cut-off frequency. The value range is [20, 20000]Hz.
Example	<HLpf no="104" opt="1">1200</HLpf> Indicates: Enable the low pass filter on output channel 104 (i.e. the analog output channel 5) with a cut-off frequency of 12000Hz.

HDly (Delay)	
<HDly></HDly> The delay module is mainly used to delay sound playing.	
Field	**no:** Channel number, no=100~107. **bypass:** bypass or not，bypass="1" (signal bypass)，bypass="0" (not bypass). **><:** Cut-off frequency. The value range is [0, 500]ms.
Example	<HDly no="104" bypass="0">300</HDly> Indicates: Enable the delay module on output channel 104 (i.e. the analog output channel 5) with a delay value of 300ms.

HPeq (Parameter Equalizer)	
<HPeq><Band>…</Band></HPeq> The parameter equalizer is mainly used to adjust the specific frequency so that the overall frequency response of the signal tends to be flat.	
Field	**no** (in <HPeq></HPeq>): Channel number, no=0~7 or 100~107. **opt** (in <HPeq></HPeq>): Equalizer enable/disable switch, opt="0" (Disable equalizer), opt="1" (Enable equalizer). **no** (in <Band></Band>): Subband number, no=0~7, the entire command must deliver 8 subbands simultaneously. **opt** (in <HPeq></HPeq>): Subband enable/disable switch, opt="0" (Disable the filter function of the corresponding subband), opt="1" (Enable the filter function of the corresponding subband). **type:** Filter type, type="1" (Low Pass Filter), type="2" (How Pass Filter), type="5" (Peak value equalizer). **frq:** Frequency, frq=20~20000Hz. **wd:** Q value, wd=2~5000. **><** (in <Band></Band>)**:** Subband gain value. The value range is [-240, 180].
Example	<HPeq no="3" opt="1"> <Band no="0" opt="0" type="5" frq="32" wd="70">0</Band> <Band no="1" opt="0" type="5" frq="80" wd="70">0</Band> <Band no="2" opt="0" type="5" frq="200" wd="70">0</Band> <Band no="3" opt="1" type="5" frq="500" wd="70">-90</Band> <Band no="4" opt="0" type="5" frq="1250" wd="70">0</Band> <Band no="5" opt="0" type="5" frq="3150" wd="70">0</Band> <Band no="6" opt="0" type="5" frq="8000" wd="70">0</Band> <Band no="7" opt="0" type="5" frq="12000" wd="70">0</Band> </HPeq> Indicates: Enable the PEQ module on channel 3, and set the fourth subband as the peak filter with a center frequency of 500Hz, Q value of 0.07, and gain value of -9dB.

HExp (Expander)	
<HExp></HExp> The expander is mainly used to compress the noise signals less than the threshold value so that the noise can be removed.	
Field	**no:** Channel number, no=0~7. **opt:** Enable/disable switch, opt="0" (disable), opt="1" (enable). **thrd:** Threshold value, thrd=-600~0. **rate:** Compression rate, rate=10~200. **wd:** Knee width, wd=0~200, wd="0"(Hard knee). **atktm:** Attack time, atktm=1~500ms. **rlstm:** Release time, rlstm=1~10000ms.

Example	<HExp no="2" opt="1" thrd="-450" rate="20" wd="0" atktm="10" rlstm="100"></HExp> Indicates: Enable the Expander function on input channel 2, and set the threshold value to -45dB, compression rate to 2, hard knee, attack time to 10ms, and release time to 100ms.
HCom (Compressor)	
<HCom></HCom> The Compressor is mainly used to compress signals larger than the threshold value, and control audio signals within a certain threshold range.	
Field	**no:** Channel number, no=0~7. **opt:** Enable/disable switch, opt="0" (disable), opt="1" (enable). **thrd:** Threshold value, thrd=-600~0. **rate:** Compression rate, rate=10~200. **wd:** Knee width, wd=0~200, wd="0"(Hard knee). **atktm:** Attack time, atktm=1~500ms. **rlstm:** Release time, rlstm=1~10000ms. **><:** Makeup gain. The value range is [-120, 180].
Example	<HCom no="2" opt="1" thrd="-90" rate="20" wd="0" atktm="10" rlstm="100">30</HCom> Indicates: Enable the Compressor function on input channel 2, and set the threshold value to -9dB, compression rate to 2, hard knee, attack time to 10ms, release time to 100ms, and makeup gain to 3dB.
HAgc (Auto Gain Control)	
<HAgc></HAgc> The AGC is mainly used to control the signal to the target level.	
Field	**no:** Channel number, no=0~7. **opt:** Enable/disable switch, opt="0" (disable), opt="1" (enable). **thrd:** Threshold, thrd=-60~0. **lvl:** Target level, lvl=-200~0. rate: Compression ratio， rate=10~200. **><:** Maximum gain value. The value range is [0, 120].
Example	<HAgc no="2" opt="1" thrd="-50" lvl="-90" rate="30">60</HAgc> Indicates: Enable the AGC function on input channel 2, and set the threshold value to -50dB, target level to -9dB, compression rate to 3, and maximum gain value to 6dB.
HLmt (Limiter)	
<HLmt></HLmt> The Limiter is a special case of the Compressor.	
Field	**no:** Channel number, no=100~107. **opt:** Enable/disable switch, opt="0" (disable), opt="1" (enable). **thrd:** Threshold, thrd=-600~0. **rlstm:** Release time, rlstm=1~10000ms.
Example	<HLmt no="102" opt="1" thrd="-90" rlstm="100"></HLmt> Indicates: Enable the Limiter function on output channel 102 (i.e. the analog output channel 3), and set the threshold to -9dB, and release time to 100ms.

HUart (Uart Parameter Configuration)	
<HUart></HUart> It is used to configure the parameters of the third-party control serial port, such as the baud rate.	
Field	**><:** Baud rate. The value can be 300, 600, 1200, 4800, 9600, 14400, 19200, 38400, 56000, 57600, or 115200.
Example	<HUart>9600</HUart> Indicates: Set the baud rate of the serial port to 9600.
HDev (Device Information Configuration)	
<HDe></HDev> It is used to configure the IP address and the name of the device.	
Field	**ip:** New IP address **mask:** Subnet mask **gtway:** Gateway **name:** Device name, less than or equal to 20 char characters. (**Note:** Please make sure to input the correct information when configuring the IP address.)
Example	<HDev ip = "192.168.1.123" mask = "255.255.255.0" gtway = "192.168.1.1" name = "XXX"></HDev> Indicates: Set the IP address to 192.168.1.123, Subnet mask to 255.255.255.0, gateway to 192.168.1.1, device name to XXX.
HSvCfg (Save Configuration)	
<HSvCfg></HSvCfg> It is used to deliver the command to enable the DSP to save the current configuration.	
Field	None
Example	<HSvCfg></HSvCfg>
HSvScnce (Save Scene)	
<HSvScnce></HSvScnce> It is used to deliver the command to save the current configuration to the corresponding scene.	
Field	**no:** Scene number, no=1~8.
Example	<HSvScnce no="7"></HSvScnce> Indicates: Save the current configuration to Scene 7.
HRdScnce (Read Scene)	
<HRdScnce></HRdScnce> It is used to read the saved scene configuration file.	
Field	**no:** Scene number, no=1~8.
Example	<HRdScnce no="7"></HRdScnce> Indicates: Read the configuration information of Scene 7.
HRstScnce (Reset Scene)	
<HRstScnce></HRstScnce> It is used to reset the corresponding scene to the factory default status.	
Field	**no:** Scene number, no=1~8.
Example	<HRstScnce no="7"></HRstScnce> Indicates: Reset the Scene 7 to the default configuration status.

8. RS-232 Control Command

The product supports RS-232 command control. Connect the RS-232 port of the product to a PC, then open a Serial Command tool on PC to send ASCII commands to control the product. The ASCII command list about the product is shown as below.

ASCII Commands				
Serial port protocol. Baud rate: 9600, Data bits: 8bit, Stop bits: 1, Check bit: 0				
Command format for setting parameters: s … ! The command starts with the letter “s”, and ends with “!”. Command format for reading device information: r … ! The command starts with the letter “r” and ends with “!”. If the command is incorrect, the corresponding error code is returned. Error codes are as following: E01 -> Invalid command E02 -> Parameters error or out of range E03 -> The CPU usage of the system is too high, and no more functions can be enabled. You need to disable some functions properly before enabling new functions.				
Command Code	**Function Description**	**Example**	**Feedback**	**Default Setting**
Scene				
s scns x!	Save the scene, x=1~8	s scns 1!	scene 1 is saved	Scene 1
r scns x!	Read the scene, x=1~8	r scns 1!	scene 1 has been read	
s scns reset x!	Restore the default scene, x=1~8	s scns reset 1!	scene 1 is restored to factory default	
Input				
s in x mute!	Mute on, x=0~8(0=all)	s in 1 mute!	input 1 volume mute	
s in x mute off!	Mute off, x=0~8(0=all)	s in 1 mute off!	input 1 volume mute off	
s in x vol+ z!	Increase the volume by step, x=0~8(0=all), z=1~25	s in 1 vol+ 3!	input 1 volume up 3dB to 3dB	
s in x vol- z!	Decrease the volume by step, x=0~8(0=all), z=1~25	s in 1 vol- 3!	input 1 volume down 3dB to -3dB	
s in x lvl z!	Set the volume, x=0~8(0=all), z=-100~12	s in 1 lvl 3!	input 1 level is 3dB	
s in x phpow on!	Turn on phantom power, x=0~8(0=all)	s in 1 phpow on!	input 1 phpow on	
s in x phpow off!	Turn off phantom power, x=0~8(0=all)	s in 1 phpow off!	input 1 phpow off	Off
r in x meter!	Get meter info, x=0~8(0=all)	r in 1 meter!	input 1 meter -64dB	
r in x mute!	Get mute status, x=1~8	r in 1 mute!	input 1 volume mute off	
r in x lvl!	Get the level, x=0~8(0=all)	r in 1 lvl!	input 1 level is 3dB	
r in x phpow!	Get phantom power status, x=1~8	r in 1 phpow!	input 1 phpow off	
Expander				
s in x expd on threshold z!	Set the threshold, x=1~8, z=-600~0	s in 1 expd on threshold -500!	input 1 expander is active and set threshold -50.0dB	
s in x expd on rate z!	Set the compression rate, x=1~8, z=10~200	s in 1 expd on rate 20!	input 1 expander is active and set rate 2.0	
s in x expd on knee z!	Set the knee width, x=1~8, z=0~200	s in 1 expd on knee 0!	input 1 expander is active and set knee 0.0	

Command Code	Function Description	Example	Feedback	Default Setting
s in x expd on attacktime z!	Set the attack time, x=1~8, z=1~500	s in 1 expd on attacktime 10!	input 1 expander is active and set attacktime 10ms	
s in x expd on releasetime z!	Set the release time, x=1~8, z=1~10000	s in 1 expd on releasetime 100!	input 1 expander is active and set releasetime 100ms	
s in x expd off!	Disable the Expander, x=1~8	s in 1 expd off!	input 1 expander is unactive	Disable
Equalizer (input)				
s in x peq on band b on type z!	Set the filter type, x=1~8, b=1~8, z=1(LP), 2(HP), 5(Peak)	s in 1 peq on band 1 on type 5!	input 1 peq is active and set band 1 filter to peak	
s in x peq on band b on frq z!	Set the frequency, x=1~8, b=1~8, z=20~20000	s in 1 peq on band 1 on frq 1000!	input 1 peq is active and set band 1 frequency 1000Hz	
s in x peq on band b on wide z!	Set the Q value, x=1~8, b=1~8, z=2~5000	s in 1 peq on band 1 on wide 2!	input 1 peq is active and set band 1 wide 0.02	
s in x peq on band b on gain z!	Set the gain value, x=1~8, b=1~8, z=-240~180	s in 1 peq on band 1 on gain -60!	input 1 peq is active and set band 1 gain -6.0dB	
s in x peq on band b off!	Disable the subband, x=1~8, b=1~8	s in 1 peq on band 1 off!	input 1 peq is active but band 1 is unactive	Disable
s in x peq off!	Disable the Equalizer, x=1~8	s in 1 peq off!	input 1 peq is unactive	Disable
Compressor				
s in x comp on threshold z!	Set the threshold, x=1~8, z=-600~0	s in 1 comp on threshold -100!	input 1 compressor is active and set threshold -10.0dB	
s in x comp on rate z!	Set the compression rate, x=1~8, z=10~200	s in 1 comp on rate 20!	input 1 compressor is active and set rate 2.0	
s in x comp on knee z!	Set the knee width, x=1~8, z=0~200	s in 1 comp on knee 0!	input 1 compressor is active and set knee 0.0	
s in x comp on gain z!	Set the makeup gain, x=1~8, z=-120~180	s in 1 comp on gain 30!	input 1 compressor is active and set makeup gain 3.0dB	
s in x comp on attacktime z!	Set the attack time, x=1~8, z=1~500	s in 1 comp on attacktime 10!	input 1 compressor is active and set attacktime 10ms	
s in x comp on releasetime z!	Set the release time, x=1~8, z=1~10000	s in 1 comp on releasetime 100!	input 1 compressor is active and set releasetime 100ms	
s in x comp off!	Disable the Compressor, x=1~8	s in 1 comp off!	input 1 compressor is unactive	Disable
Auto Gain Control				
s in x agc on target z!	Set the target level, x=1~8, z=-200~0	s in 1 agc on target -90!	input 1 agc is active and set target level -9.0dB	

Command Code	Function Description	Example	Feedback	Default Setting
s in x agc on rate z!	Set the agc rate, x=1~8, z=10~200	s in 1 agc on rate 30!	input 1 agc is active and set rate 3.0	
s in x agc on maxgain z!	Set the maximum gain value, x=1~8, z=0~120	s in 1 agc on maxgain 60!	input 1 agc is active and set maxgain 6.0dB	
s in x agc on threshold z!	Set the threshold, x=1~8, z=-60~0	s in 1 agc on threshold -50!	input 1 agc is active and set threshold -50dB	
s in x agc off!	Disable Auto Gian Control, x=1~8	s in 1 agc off!	input 1 agc is unactive	Disable
Output				
s out y mute!	Mute on, y=0~8(0=all)	s out 1 mute!	output 1 volume mute	
s out y mute off!	Mute off, y=0~8(0=all)	s out 1 mute off!	output 1 volume mute off	
s out y vol+ z!	Increase the volume by step, y=0~8(0=all), z=1~25	s out 1 vol+ 3!	output 1 volume up 3dB to 3dB	
s out y vol- z!	Decrease the volume by step, y=0~8(0=all), z=1~25	s out 1 vol- 3!	output 1 volume down 3dB to -3dB	
s out y lvl z!	Set the volume, y=0~8(0=all), z=-100~12	s out 1 lvl 3!	output 1 level is 3dB	
r out y meter!	Get the meter information, y=0~8(0=all)	r out 1 meter!	output 1 meter -64dB	
r out y mute!	Get the mute status, y=1~8	r out 1 mute!	output 1 volume mute off	
r out y lvl!	Get the level, y=0~8(0=all)	r out 1 lvl!	output 1 level is 0dB	
Delay				
s out y delay on delay z!	Set the delay time, y=1~8, z=0~500	s out 1 delay on delay 10!	output 1 delay is active and delay 10ms	
s out y delay off!	Turn off delay, y=1~8	s out 1 delay off!	output 1 delay is unactive	Off
r out y delay!	Get the delay status, y=1~8	r out 1 delay!	output 1 delay is unactive	
Matrix Routing				
s out y hit in x lvl z!	Select settings, y=1~8, x=1~12 (9-AM,10-AFC, 11-AEC,12-ANS), z=-100~12	s out 1 hit in 1 lvl 3!	output 1 hit input 1 and set volume 3dB	
s out y remove in x!	Remove the input, y=1~8, x=1~12 (9-AM,10-AFC, 11-AEC,12-ANS)	s out 1 remove in 1!	output 1 remove input 1	
r out y matrix!	Get the routing status, y=1~8	r out 1 matrix!	output 1 hit input [1] volume [3]dB	
Auto Mixer				
s amx hit x!	Auto mixer, x=1~8	s amx hit 1!	automixer hit 1	
s amx remove x!	Remove auto mixer, x=1~8	s amx remove 1!	automixer remove 1	

Command Code	Function Description	Example	Feedback	Default Setting
Adaptive Feedback Cancellation				
s afc hit x!	Select the input, x=1~8	s afc hit 1!	afc hit 1	
s afc remove x!	Remove the input, x=1~8	s afc remove 1!	afc remove 1	
s afc on!	Enable AFC	s afc on!	afc is active	
s afc off!	Disable AFC	s afc off!	afc is unactive	Disable
s afc ans on level z!	Enable the noise suppression function, z=6~18	s afc ans on level 12!	ans under afc is active and suppress noise 12dB	
s afc ans off!	Disable the noise suppression function.	s afc ans off!	ans under afc is unactive	Disable
Adaptive Echo Cancellation				
s aec local hit x!	Select the local input, x=1~8	s aec local hit 1!	aec local hit 1	
s aec local remove x!	Remove the local input, x=1~8	s aec local remove 1!	aec local remove 1	
s aec ref hit x!	Select the reference signal, x=1~8	s aec ref hit 3!	aec ref hit 3	
s aec ref remove x!	Remove the reference signal, x=1~8	s aec ref remove 3!	aec ref remove 3	
s aec on nlp z!	Set NLP level, z=1~3	s aec on nlp 2!	aec is active and nlp set 2	
s aec off!	Disable AEC	s aec off!	aec is unactive	Disable
s aec ans on level z!	Enable the noise suppression function, z=6~18	s aec ans on level 12!	ans under aec is active and suppress noise 12dB	
s aec ans off!	Disable the noise suppression function.	s aec ans off!	ans under aec is unactive	Disable
r aec!	Get AEC status	r aec!	aec is unactive	
r aec ans!	Get the noise suppression status.	r aec ans!	ans under aec is unactive	
Adaptive Noise Suppression				
s ans hit x!	Select the input, x=1~8	s ans hit 1!	ans hit 1	
s ans remove x!	Remove the input, x=1~8	s ans remove 1!	ans remove 1	
s ans on level z!	Set the noise suppression level, z=6~18	s ans on level 12!	ans is active and suppress noise 12dB	
s ans off!	Disable ANS.	s ans off!	aec is unactive	Disable
r ans!	Get the noise suppression status.	r ans!	ans is unactive	
Equalizer (Output)				
s out y peq on band b on type z!	Set the filter type, y=1~8, b=1~8, z=1(LP), 2(HP), 5(Peak)	s out 1 peq on band 1 on type 5!	output 1 peq is active and set band 1 filter to peak	
s out y peq on band b on frq z!	Set the frequency, y=1~8, b=1~8, z=20~20000	s out 1 peq on band 1 on frq 1000!	output 1 peq is active and set band 1 frequency 1000Hz	

Command Code	Function Description	Example	Feedback	Default Setting
s out y peq on band b on wide z!	Set the Q value, y=1~8, b=1~8, z=2~5000	s out 1 peq on band 1 on wide 2!	output 1 peq is active and set band 1 wide 0.02	
s out y peq on band b on gain z!	Set the gain value, y=1~8, b=1~8, z=-240~180	s out 1 peq on band 1 on gain -60!	output 1 peq is active and set band 1 gain -6.0dB	
s out y peq on band b off!	Disable the subband, y=1~8, b=1~8	s out 1 peq on band 1 off!	output 1 peq is active but band 1 is unactive	Disable
s out y peq off!	Disable the Equalizer, y=1~8	s out 1 peq off!	output 1 peq is unactive	Disable
Limiter				
s out y lmt on threshold z!	Set the threshold, y=1~8, z=-600~0	s out 1 lmt on threshold -100!	output 1 limiter is active and set threshold -10.0dB	
s out y lmt on releasetime z!	Set the release time, y=1~8, z=1~10000	s out 1 lmt on releasetime 100!	output 1 limiter is active and set releasetime 100ms	
s out y lmt off!	Disable the Limiter, y=1~8	s out 1 lmt off!	output 1 limiter is unactive	Disable
High Pass Filter				
s out y hpf on frq z!	Set the cut-off frequency, y=1~8, z=20~20000	s out 1 hpf on frq 200!	output 1 hpf is active and set cut_off frequency 200Hz	
s out y hpf off!	Disable the High Pass Filter, y=1~8	s out 1 hpf off!	output 1 hpf is unactive	Disable
Low Pass Filter				
s out y lpf on frq z!	Set the cut-off frequency, y=1~8, z=20~20000	s out 1 lpf on frq 12000!	output 1 lpf is active and set cut_off frequency 12000Hz	
s out y lpf off!	Disable the Low Pass Filter, y=1~8	s out 1 lpf off!	output 1 lpf is unactive	Disable
Device Status				
r ipconfig!	Get the IP information.	r ipconfig!	IP:192.168.0.199 Subnet Mask: 255.255.255.0 Gateway: 192.168.0.1	
r cpu rate!	Get the CPU information.	r cpu rate!	cpu occupancy rate is 18%	

9. Troubles and Solutions

If the output audio is abnormal, you may need to check following items at first:

☆ Whether the STATUS LED is normal.
☆ Whether the INPUT/OUTPUT VOLUME is turned down to the lowest level.
☆ Whether cables are in right connection.
☆ Whether the matrix mixer is configured correctly.
☆ Whether the audio modules are applied correctly.

If echo persists when the processor is working, you need to check:

☆ Whether the processor is in normal operation.
☆ Whether the AEC module is activated.
☆ Whether the matrix routing is correctly configured.
☆ Whether the connection of Amplifier or Speaker for outputting reference signals is consistent with the matrix routing.

10. Application Example

The audio processor supports microphone input, computer, DVD, mixer and other audio source input. It can be used in combination with video conference terminal, recording and broadcasting system, and mixing console. The following figure shows the general connection diagram. Users can connect the devices based on actual requirements.

