

# BG-A88M

# 8x8 Balanced Digital Audio Processor/Matrix

# **User Manual**





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#### Statement

Please read these instructions carefully before connecting, operating, or configuring this product. Please save this manual for future reference.

#### **Safety Precaution**

- To prevent damaging this product, avoid heavy pressure, strong vibration, or immersion during transportation, storage, and installation.
- The housing of this product is made of organic materials. Do not expose to any liquid, gas, or solids which may corrode the shell.
- Do not expose the product to rain or moisture.
- To prevent the risk of electric shock, do not open the case. Installation and maintenance should only be carried out by qualified technicians.
- Do not use the product beyond the specified temperature, humidity, or power supply specifications.
- This product does not contain parts that can be maintained or repaired by users.
   Damage caused by dismantling the product without authorization from BZBGEAR is not covered under the warranty policy.
- Installation and use of this product must strictly comply with local electrical safety standards.

#### Introduction

The BG-A88M is a digital audio processor which offers 8-ch balanced MIC/linear inputs and 8-ch balanced linear outputs. It features adaptive feedback suppression, full-band adaptive acoustic echo cancellation, and dynamic adaptive noise reduction technology. The Auto Mixer function allows users to set a priority when multiple microphones are used.

The BG-A88M includes various digital signal processing features such as an Expander, Equalizer, Compressor, Auto Gain Control, Limiter, High Pass Filter, Low Pass Filter, and Delay. It also supports matrix routing, volume control, metering, scene control, and more. It can provide a 48V phantom power supply for the 8 MIC inputs, has a 48KHz sampling rate, and 24-bit A/D or D/A conversion.

The BG-A88M also supports 8-ch programmable GPIO function. The included control software is compatible with Windows 7/10/11 operating systems. The unit can be connected and controlled via the network using a standard RJ45 interface or via RS-232 control using serial commands.

#### **Features**

- 8-ch balanced MIC/linear inputs and 8-ch balanced linear outputs
- Adaptive feedback suppression
- Full-band adaptive acoustic echo cancellation technology
- Dynamic adaptive noise reduction technology can reduce signal to noise level by up to 18dB
- Auto Mixer function can set the order of priority when multiple microphones are input at one time
- Digital signal processing: Expander, Equalizer, Compressor, Auto Gain Control, Limiter, High Pass Filter, Low Pass Filter, and Delay.
- Matrix routing
- 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
- 8-ch programmable GPIO function
- Software control via Windows 7/10/11 with standard RJ45 interface control
- RS-232 serial command and control supported

# **Package Contents**

- ① 1 x BG-A88M
- (2) 1 x AC 100~240V 50/60Hz Power Cord
- ③ 1 x 3-pin 3.81mm Phoenix Connector (Male)
- ④ 1 x 10-pin 3.81mm Phoenix Connector (Male)
- (5) 4 x 12-pin 3.81mm Phoenix Connectors (Male)
- 6 2 x Mounting Ears (Wall Mount)
- ⑦ 8 x Machine Screws (KM3\*4)
- (8) 1 x User Manual

# **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 Impedance	20ΚΩ
Output Impedance	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	17" x 9.8" x 1.8" [440mm (W) x 250mm (D) x 45mm (H)]
Weight	7.3lbs [3.3kg]
Power Supply	AC 100 - 240V 50/60Hz
Power Consumption	9W (Max)
Operating Temperature	32°F ~ 104°F / 0°C ~ 40°C
Storage Temperature	-4°F ~ 140°F / -20°C ~ 60°C
Relative Humidity	20~90% RH (non-condensing)

# **Device Interface Descriptions**

#### **Front Panel**



No.	Name	Description
1	POWER LED	When the device is powered on, the red LED will be illuminated.
2	STATUS LED	When the device is running normally, the green STATUS LED will flash.

#### **Rear Panel**



No.	Name	Description
1	MIC +48V	8-ch 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 to an amplifier or speaker via phoenix connectors.
4	GPIO	8-ch programmable GPIO interfaces.
5	RS-232	Connect to a PC or control system to transmit RS-232 control commands.
6	LAN	Standard RJ45 interface for network connection.
7	RESET	Used for resetting 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.

# **Computer Control**

#### Connection

The BG-A88M audio processor can be controlled via the provided software on a Windows 7/10/11 machine. Use a category cable to directly connect to the LAN port of the processor to your computer or via your local network using the following steps.

**Step 1**: Install the "BZBGEARAudioMatrix8.exe" application on 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.

Obtain an IP address aut	omatically
• Use the following IP addr	ess:
IP address:	192.168.0.199
Subnet mask:	255.255.255.0
Default gateway:	

**Step 2:** Double-click the "BZBGEARAudioMatrix8.exe" icon to run the application on the host computer.

File Tools Help		
HOME MOTOL BOY BOOM BACC B MOTOL BOY BOOM BACC B MOTOL BOY BOOM BACC B MOTOL BOY	IN AUTOMIXER AFC AEC ANS MA	TRUX         OUT           24447         5         6
	200	

Step 3: When the application is opened the HOME page will appear as shown below.

Step 4: Click the Network icon in the toolbar at the top



If there is a red X at the top-right corner of the icon as shown in the figure above, it means the host computer has not connected with the audio processor.

If there is a red X, click the icon and a dialog box of all IP addresses from connected devices will appear as seen in the following image.



**Step 5**: Choose the audio processor's IP address and click CONNECT. The processor will now be connected with the host computer. After connection is established, the CONNECT & SETUP box will appear as shown below. You may close it if no changes are needed.



If the audio processor has been connected with the host computer successfully, the network icon will have a green check in the top right corner as shown.



#### **Volume Meter**

The volume meters will display the signal level of the various input & output volumes.



The meter has 3 colors (Red, Yellow, and Green) to denote signal levels for the volume of the 8 inputs and 8 outputs. Red means that volume may be too high to be limited, yellow indicates the volume is ideal, while green is within a reasonable lower range.

# Input Settings

INPUT1

As shown below, clicking the INPUT module on the HOME page or the "IN" tab on the menu bar will open the INPUT setting page.

AUTOMIXER AFC AEC ANS MATRIX OUT 2-WAY A EQ A DELAY A LIM OUTPUT 2-WAY A EQ A DELAY A LIM OUTPUT2 2-WAY 🖞 EQ 🧴 DELAY 👗 LIM 🔒 OUTPUT3 AUTO AFC 🐴 AEC Å ANS Å MIXER MIXER 2-WAY EQ DELAY LIM 2-WAY A EQ A DELAY A LIM OUTPUTS 

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

HOME IN AUTOMIXER AFC AEC	ANS MATRIX OUT	
	SOURCE LEVEL	
0 dB	PINK 0.0dB	CHAN 1 -
	SINE -6 - 200K	EXP
	600	EQ
INVERT PHANTOM MUTE	90 - 50	COMP
	0.0	AGC

**GAIN CONTROL:** Indicates the GAIN level of the input signal which can be adjusted. **INVERT:** Inverts the polarity of the signal on the input channel.

**DHANTOM:** Turns on/off 48\/ phantom power for the channel

**PHANTOM:** Turns on/off 48V phantom power for the channel.

MUTE: Mutes the input channel, equivalent to clicking the letter M on the INPUT module

on the HOME page.

WHITE: Generates white noise for signal testing.

**PINK:** Generates pink noise for signal testing.

**SINE:** Generates a sinusoidal signal for testing and the meter below shows the frequency of the sine wave.

FADER: Controls the output level of the test signal.

You can set the signal parameters of an input channel by entering its "CHAN" number in the menu on the right side of the "IN" page. Audio process modules such as EXPANDER, EQUALIZER, COMPRESSOR and AUTO GAIN CONTROL, are also available in the right panel.



The Expander can extend the dynamic range of the input signal and is used to eliminate noise under a certain threshold.

As shown below, clicking the EXP module on the HOME page or the EXP button in the right panel of the IN page will open the EXP setting page.

	EXP 4	EQ	AGC <sup>A</sup>		
	EXP 4	EQ	AGC <sup>A</sup>		INFOI
	EXP 4	EQ	AGC <sup>A</sup>		CHAN 4
INPUT4 M	EXP 4	EQ	AGC <sup>A</sup>		EXP
INPUT5 M	EXP 4	EQ	AGC <sup>A</sup>	or	EQ
	EXP 4	EQ	AGC <sup>A</sup>		COMP
	EXP 4	EQ	AGC <sup>A</sup>		
	EXP 4	EQ	AGC <sup>A</sup>		AGC

The following figure shows the expander module:

EXPANDER INP	PUT CHANNEL 1	
THRESHOLD         -60.0dB           Q         -00.0dB           RATIO         1.0           Q         -00.0dB           KNEE         0.0           Q         -00.0dB	ATTACK 10ms	GR 0 -3 -6 -9 -12 -15 -18 -21 -24 -27 -27 -30
AC	CTIVE	

**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 this 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. It 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.

#### Equalizer

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

As shown below, clicking the EQ module on the left panel of the HOME page or the EQ button of the IN page to open the EQ setting page of the input channel.

eq 🐴 comp 🐴 agc 🔺	
eq 🐴 comp 🐴 agc 🐴	
eq 🐴 comp 🐴 agc 🐴	
eq 🐴 comp 🐴 agc 🐴	
eq 🐴 comp 🐴 agc 🐴	
eq 🐴 comp 🐴 agc 🐴	



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

	8-BA	ND PARAME	TRIC EQ OL		ANNEL 1	——————————————————————————————————————	
18 20 31.5	50 80	125 200 3	15 500 800	1.25k 2k	3.15k 5	Sk 8k	12.5k 20k
15							
\$							
.3	<b>O</b>	9	<b>O</b>	0	<b>O</b>	U	0
-6 -9 -12							
-15							
-21 -24 25	40 63 1	00 160 250	400 630	1k 1.6k	2.5k 4k	6.3k 10	ik 16k
TYPE PE							
			500.0	1250.0	2150.0	2000.0	12000.0
FREQ(Hz)	2.0 OU		500.0	1250.0	3150.0	8000.0	12000.0
GAIN(dB)	0.0 0	.0 0.0	0.0	0.0	0.0	0.0	0.0
	70 0	70 0 70	0.70	0.70	0.70	0.70	0.70
ON/OFF							
			ACTIVE				

**TYPE:** Provides 3 types of band/filters: High Pass, Low Pass, and Peak.

**FREQUENCY:** In High/Low Pass Filter, it refers to the cut-off frequency of the EQ band; In Peak Filter, it refers to the center frequency point of the EQ band; with a scale of [20, 20000] Hz.

**GAIN:** In High/Low Pass Filter it is fixed to 0dB; In Peak Filter it is between [-24.0 to +18.0] dB.

**Q FACTOR:** The value is limited between [0.02, 50.00].

2

**ON/OFF SWITCH:** Each band has a switch. When pressed, the EQ function in this band can be enabled or disabled.

**ACTIVE:** Activates the EQ process. The function is equivalent to clicking the letter A on the EQ module on the HOME page.

A pair of UP/DOWN arrows on the right panel can be used to select an input channel for the Equalizer configuration.

#### Compressor

The Compressor is used to reduce the dynamic range of the signal above a user-determined threshold.

As shown below, click the COMP module on the HOME page or the COMP button on the right panel of the IN page to open the Compressor settings.



The figure 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 in a scale of [1.0, 20.0]. When the ratio sets to 2.0, it means that the signal with a level below that threshold will be reduced to  $\frac{1}{2}$  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; others indicate a soft knee.

**ATTACK:** The time required by the compressor to begin the Compressor process once a signal is over the 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 a threshold, with a scale of [1, 10000] ms.

**MAKEUP GAIN:** Since the compressor can reduce the gain level, the Makeup Gain allows for a compensation of the gain level, with a scale of [-12.0, +18.0] dB.

**G.R. (Gain Reduction):** Indicates the amount of gain attenuation for the input signal (in dB). **OUT:** The 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 to select an input channel for the Compressor configuration.

# Auto Gain Control

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

or

As shown below, by clicking the AGC module on the HOME page or the AGC button on the right panel of the IN page will open the AGC setting.

	EXP <sup>A</sup>	EQ	AGC <sup>A</sup>	
INPUT2 M IN2	EXP	EQ	AGC <sup>A</sup>	
	EXP <sup>A</sup>	EQ	AGC <sup>^</sup>	
INPUT4 M IN4	EXP <sup>^</sup>	EQ	AGC <sup>^</sup>	
INPUT5 M	EXP <sup>A</sup>		AGC <sup>^</sup>	
	EXP <sup>A</sup>	EQ	AGC <sup>A</sup>	
INPUT7 M	EXP <sup>A</sup>	EQ	AGC <sup>^</sup>	
	EXP <sup>A</sup>		AGC <sup>^</sup>	

INPUT	
CHAN 1 🔻	
EXP	
EQ	
COMP	
AGC	



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

**TARGET LEVEL:** The desired output signal level, with a scale of [-20.0, 0.0] dB.

**RATIO:** The compression ratio of signals above the target level, with a scale of [1.0, 20.0]. If the value sets to 2.0, it means that signals above that target level will be amplified twice of the original. Note, not all signals between the threshold and the target level are linearly amplified by this ratio.

**MAX GAIN:** This 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 the threshold will remain unchanged while those between threshold and the target level will be amplified (at a ratio determined by the TARGET LEVEL, RATIO and MAX GAIN). In order to ensure the proper function of the AGC process, the Target Level must be set higher than Threshold. The threshold is recommended to be 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 are used to select an input channel for the AGC configuration.



The AUTO MIXER is used to set the priority for MIC/linear inputs when 2 or more MIC/LINE ports are connected at once.

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



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

IN1	IN2	IN3	IN4	IN5	IN6		IN8	OUTP	UT:
+12 +9 +3 -0 -20 -40 - -60 - -80 - -100	+12 +9 +3 -0 -20 -40 -60 -60 -60 -100	+12 +9 +3 -0 -20 -40 -60 -60 -60 -60	+12 +9 +3 -0 -20 -40 -60 -60 -60	+12 +9 +6 +3 -0 -20 -20 -40 -60 -60 -60 -60	+12 +9 +3 -3 -20 -40 -60 -60 -60	+12 +9 +3 -30 -40 -60 -80 -100	+12 +9 +3 -33 -9 -39 +99 -100	+36 +24 +12 0 -12 -24 -36	+12 +9 +3 -0 -40 -60
0.0 MUTE PR 5 \$ AUTO	0.0 MUTE PR 5 0 AUTO	0.0 MUTE PR 5 \$ AUTO	0.0 MUTE PR 5 \$ AUTO	0.0 MUTE PR 5 \$ AUTO	0.0 MUTE PR 5 \$ AUTO	0.0 MUTE PR 5 \$ AUTO	0.0 MUTE PR 5 \$ AUTO	-48 -64	0.0

**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 users to mute the channel.

**PR (Priority):** Sets the priority of the MIC input, 0 = lowest and 10 = highest priority.

Channels with a higher-priority will override the ones 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..

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

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



#### **Adaptive Feedback Cancellation**

The Adaptive Feedback Cancellation (AFC, also known as Feedback Suppression) module can detect and suppress a frequency point of acoustic feedback. The AFC module allows you to detect and suppress frequency points of acoustic feedback for 8 bands. Parameters of this module are configurable.

As shown below, clicking the AFC module on the HOME page or the AFC tab on the menu bar, will open 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 can reduce, with a scale of [-20.0, 0.0] dB.

**BANDWIDTH:** The bandwidth of the filter, with Narrow and Wide options.

**RESET:** When pressed, the filter will go back to its initial settings and the module will re-detect the frequency points of the acoustic feedback.

**FIX BAND:** When pressed, fixed values of parameters, such as, Frequency, Gain, and Width will be applied to configure the filter in a fixed band. Parameters are configurable.

**FIX ALL:** When pressed, all bands will be processed in a fixed mode or in USER-EDITING mode.

BYPASS BAND: When pressed, the filter of the related band will be disabled.

**BYPASS ALL:** 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 used to suppress noises with signal level up to 18dB. Clicking the ACTIVE button will give you the option to activate this function.

## **Adaptive Echo Cancellation**

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

As shown below, clicking the AEC module on the HOME page or the AEC tab on the menu bar will open the AEC configuration page.



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



**LOCAL INPUT:** Shows source signals that are available 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 options.

**ACTIVE:** Activates the AEC process. The function is equivalent to clicking the green letter A on the AEC module on the HOME page.

**ANS LEVEL:** The Adaptive Noise Suppression (ANS) function is used to suppress noises with signal levels up to 18dB. Pressing the ACTIVE button enables the function.

**REMOTE REFERENCE:** Designates signals that are available for the AEC algorithm to learn how to analyze and eliminate echoes.

As shown in the figure above, IN5 was chosen as a reference signal. In such a case, we would need to connect the IN5 port of the processor to the video conference device. The reference signal after the AEC algorithm will be sent to the far end via the connected device. Users will also need to configure the selected input channels in the matrix routing.



The figure 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

## **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 appliances such as a fan or air conditioner.

As shown below, pressing the ANS module on the HOME page or the ANS tab on the menu bar, will open the ANS configuration page.

HOME	IN AUTOMI	KER AFC	AEC	ANS	MATRIX	OUT		
				-way A EQ	A DELAY			
						-way A EQ	A DELAY	
		AUTO			-way A EQ	A DELAY		
	AUTO			MATR	IX 2-W	-way A EQ	A DELAY	
	MIXER	AFC AEC	C C ANS	MIXE	R	-way A EQ	A DELAY	
						-way A EQ	A DELAY	
						-way A EQ	A DELAY	
					2	-way 👌 EQ	A DELAY	

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

LOCAL INPU	т:		ĺ		
	IN2				
IN3	IN4				
IN5	IN6	ANO			
IN7	IN8				
AUTOMIXER		AN	NS	18	
				ACTIVE	

**LOCAL INPUT:** Shows source signals that are available 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.

#### **Matrix Mixer**

The Matrix Mixer allows you to freely choose the audio route via matrixing.

As shown below, clicking the MATRIX MIXER section on the HOME page or the MATRIX tab on the menu bar will open the MATRIX MIXER configuration page.

HOME	IN AUTOMI	KER AFC	AEC	ANS	MATRIX	ουτ		
		1UTO			2-WAY A EQ	A DELAY		
					2-WAY A EQ			
					2-WAY A EQ	A DELAY		
	AUTO		MAT	RIX	2-WAY A EQ	A DELAY A LIM		
	MIXER	AFC AE	AFC AEC ANS	MIX	ER	2-WAY A EQ	A DELAY	
						2-WAY A EQ	A DELAY	
						2-WAY A EQ	A DELAY	
					1	2-WAY A EQ	A DELAY	

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
0UT4	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
0UT8	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 lines are output channels and the vertical are input channels (except AM, AFC, AEC and ANS).

AM: Indicates the output signal after the AUTO 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.

## **Output Settings**

As shown below, clicking the OUTPUT module on the HOME page or the OUT tab on the menu bar will open the OUTPUT setting page.

OUTPUT

ном	I IN	AUTOMI	KER AFC	AEC	ANS MATE			
	<u>^</u>	Αυτο	AUTO			2-WAY A EQ A DE		
							2-WAY A EQ A DE	
						MATRIX	2-WAY A EQ A DE	
	Δ ,						2-WAY A EQ A DE	
	<u> </u>	<b>IIXER</b>	KER AFC AEC	EC M ANS M	MIXER	2-WAY A EQ A DE		
	<u>.</u>					2-WAY A EQ A DE		
	•						2-WAY 🛕 EQ Å DE	
	<u>A</u>					2-WAY 🙆 EQ Å DE		

The figure below shows the details of the setting page.

a	-10	CHAN 1 -
Ø	- "	2-WAY
		EQ
INVERT	MUTE	DELAY
		LIM

**INVERT:** When pressed, the polarity of the signal of the selected channel will be inverted. **MUTE:** When pressed, the output channel will be muted. This function is equivalent to pressing the letter M on 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 configurable.



There is a sound in the toolbar which can mute all analog output channels at one time (see the figure above). If it is in red, it indicates that all output channels are muted; otherwise, it will be green.

#### **Two-way Crossover**

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

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

2-WAY A EQ A DELAY A		
2-WAY A EQ A DELAY A		OUTPUT
2-WAY A EQ A DELAY A		CHAN 1 -
2-WAY A EQ A DELAY A	or	
2-WAY A EQ A DELAY A		EQ
2-WAY A EQ A DELAY		DELAY
2-WAY A EQ A DELAY A		LIM
2-WAY A EQ A DELAY A		

The figure below shows the Two-way Crossover setting page.



**ACTIVE:** When pressed it will activate/deactivate the frequency filter (High Pass or Low Pass). The process can also be activated by clicking the green letter A in the 2-WAY

module and the HOME page. The difference is that clicking the letter A will activate/deactivate 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/deactivate the corresponding frequency filter.

**FREQUENCY:** Indicates the cut-off frequency of the filter.

**TYPE:** Butterworth filter is default.

SLOPE: 6dB/oct by default.

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



#### Delay

The Delay module is typically used to delay the output speed of audio signals which can be used to sync audio with a video.

Clicking the DELAY module OPELAY on the HOME page or the DLY button on the OUT page will open the configuration page for the Delay module.



The figure below shows the setting page of this function.

DELAY OL	JTPUT CH	ANNEL 1	
DELAY TIME	 500ms	0ms	
DELAY CONVER	SION		
0.000		0.000	
Feet		Meters	
	ACTIVE		

**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, it will activate or deactivate the Delay module. This function can

also be done by clicking the green letter A at the DELAY module DELAY on the HOME page.

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

#### Limiter

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

Clicking the LIM module on the HOME page or the LIMITER button on the OUT page, will open the LIMITER configuration page.



The figure below shows the setting page of this module.

LIMITER OUTPUT CHANNEL 1		
	G.R.	OUT
		+36
		+24
P	-6	+12
	-9	
RELEASE 100ms	-12	0
	-15	-12
	-18	-24
	-21	-36
	-24	40
	-27	-48
ACTIVE	-30	-64

**THRESHOLD:** Shows signal levels above the threshold that 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.

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

#### **Scene Management**

Up to 8 scenes can be configured in the audio processor. By clicking "Tools"> "Scene" on the menu bar, you will enter the setting page shown below.

ID(STATUS)	NAME	Load
1(CURRENT)	SCENE1	Upload
2	SCENE2	Download
3	SCENE3	Download
4	SCENE4	Modify Name
5	SCENE5	Save
6	SCENE6	Dent
7	SCENE7	Reset
8	SCENE8	

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 toolbar 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 toolbar as shown below.



By default, the file is saved in the .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.

## **Serial Settings**

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

Serial Setting		×
Setting		
Port	232	
Baud rate	9600	
Parity	NONE	
Data bits	8	
Stop bit	1	
Save	Reset	

**Baud Rate:** 9600 by default, open to change.

**Reset:** Clicking it to restore the system default settings.

# **Control Commands**

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

		×
Command		
Туре	Scene Manager	<b>-</b>
Command	Save	•
Channel		•
Band		
Value		
CMD Code	s scns 1!	
		CLOSE

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 the page once you have selected the Type, Command, Channel, Band from the drop-down box and entered its value.



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

GPIO Direction	O Input	Disable	] Stat	us Description
Display Type:	Sence Display			Output Low
Output Level Typ Output low	e , otherwise high h, otherwise low		-	Input Rising Edge
L			, ,	Input Falling Edge
Sence Selecti	on: Sence1			Disable

Users can configure the GPIO function as needed and see details on the GPIO Setting page. Users can also check the GPIO status via the GPIO LED at the top-right corner of the main page.

#### **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.).

# **TCP/IP Control Command**

The BG-A88M supports TCP/IP command control. Connect the LAN port of the unit to a PC, then open a TCP/IP Command tool on the PC to send TCP/IP commands to control the device. The TCP/IP command list is shown below.

	TCP/IP Commands			
Service Po	rt Number: 3433			
The contro the comma Channel nu Channel nu Channel nu Channel nu Channel nu Channel nu	I command ends with "\n", and no symbols in the form of "\n" can appear in the middle of and. umber: 0~7, corresponding to the Mic/Line input channels 1~8 of the device. umber: 96 indicates AM, which is the audio signal after the auto mixer process. umber: 97 indicates AFC, which is the audio signal processed by the AFC module. umber: 98 indicates AEC, which is the audio signal processed by the AEC module. umber: 99 indicates ANS, which is the audio signal processed by the ANS module. umber: 100~107, corresponds to the analog output channels 1~8 of the device. umber: 196 indicates the reference signal of AEC.			
HChanl (C	hannel Command)			
<hchanl>«</hchanl>				
Field	<ul> <li>no: Channel number, no=0~7 or 100~107.</li> <li>mu: Mute switch, mu="0" (unmute), mu="1" (mute).</li> <li>phpow: Phantom power switch, phpow="0" (phantom power on), phpow="1" (phantom power off).</li> <li>invrt: Invert function switch, invrt ="0"(non-invert), invrt ="1" (invert).</li> <li>&lt;: Gain setting. The value range is [-12, +40]dB.</li> </ul>			
Example	<pre><hchanl invrt="1" mu="0" no="0" phpow="1">1</hchanl> Indicates: Channel 0, unmute, phantom power on, invert, analog gain amplified 1dB.</pre>			
HNg (Signa	al Generator)			
<hng><td>Ng&gt;</td></hng>	Ng>			
It is used to	o generate white noise, pink noise and sinusoidal signal of arbitrary frequency.			
Field	<ul> <li>no: Channel number, no=0~7.</li> <li>opt: Enable/disable switch, opt="0" (disable), opt="1" (enable).</li> <li>type: Signal type, type="0" (white noise), type="1" (pink noise), type="2" (sinusoidal signal)</li> <li>frq: The frequency of sinusoidal signal, frq =20~20000Hz.</li> <li>&gt;: Level value. The value range is [- 60, 0]dB.</li> </ul>			
Example	<hng frq="1000" no="3" opt="1" type="2">-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	o 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 (Aut	o Mixer)						
<hamx><i< th=""><th>n &gt;</th></i<></hamx>	n >						
It is used to	It is used to control the auto mixer function.						
Field	<ul> <li>no: Channel number, no=0~7.</li> <li>mu: Mute switch, mu="0" (unmute), mu="1" (mute).</li> <li>pry: Priority, pry=0~10, A larger value indicates a higher priority.</li> <li>&gt;&lt;: Level value of the auto mixer channel. The value range is [-100, 12]dB.</li> </ul>						
Example	<hamx> <in mu="0" no="1" pry="5">0</in> <in mu="0" no="3" pry="5">0</in> <in mu="1" no="6" pry="5">0</in> <in mu="0" no="7" pry="5">0</in> </hamx> Indicates: Add input channels 1, 3, 6, 7 to the auto mixer module for auto mixing algorithm						
	processing.						
HAns (Ada	ptive Noise Suppression)						
<hans><ir< th=""><th>&gt;</th></ir<></hans> o control the ANS (Adaptive Noise Suppression) module.	>						
Field	<ul> <li>ans: ANS switch or suppression level, ans=0 or 6~18, ans="0" (disable ANS), ans="6"~"18" (noise suppression level).</li> <li>no: Channel number, no=0~7 or 96.</li> </ul>						
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 (Ada	ptive Echo Cancellation)						
<haec><in It is used to and non-lin</in </haec>	>						
Field	<ul> <li>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).</li> <li>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).</li> <li>no: Channel number, no=0~7 or 96.</li> </ul>						
Example	<haec aec="2" ans="15"> <li><ln no="1"></ln> <ln no="3"></ln> <ln no="6"></ln> <ln no="96"></ln> </li></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 (Ada	aptive Feedback Cancellation)
<hahc><li< td=""><td>n&gt;</td></li<></hahc>	n>
It is used t	o control the input source selection and other functions of the AFC module.
Field	<ul> <li>ahc: Reserved</li> <li>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).</li> <li>no: Channel number, no=0~7 or 96.</li> </ul>
Example	<hahc ahc="1" ans="15"> <ln no="1"></ln> <ln no="3"></ln> <ln no="6"></ln> <ln no="96"></ln> </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 (Ma	atrix Mixer)
<hmmx>&lt; Matrix mix</hmmx>	In> er can rout any input audio signal to any output.
Field	<b>no</b> (in <hmmx></hmmx> ): Channel number, no=100~107. <b>no</b> (in <ln></ln> ): Channel number, no=0~7 or 96, 97, 98, 99. >< (in <ln></ln> ): 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>

iviatrix mixe	er can rout any input audio signal to any output.			
Field	<b>no</b> (in <hmmx></hmmx> ): Channel number, no=100~107. <b>no</b> (in <ln></ln> ): Channel number, no=0~7 or 96, 97, 98, 99. <b>&gt;&lt;</b> (in <ln></ln> ): 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	n Pass Filter)			
<hhpf><!--ł<br-->The high pa</hhpf>	HHpf> ass 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 <hr/> <pre></pre> <pre< th=""></pre<>				
HLpf (Low	Pass Filter)			
It is used to control the The low pass filter is mainly used to remove high frequency noise.				
Field	<ul> <li>no: Channel number, no=100~107.</li> <li>opt: Enable/disable switch, opt="0" (disable), opt="1" (enable).</li> <li>&gt;: Cut-off frequency. The value range is [20, 20000]Hz.</li> </ul>			
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.			



HDlv (Dela						
<hdlv><th>1DIv&gt;</th></hdlv>	1DIv>					
The delay	module is mainly used to delay sound playing.					
	no: Channel number, no=100~107.					
Field	<b>bypass:</b> 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 Indicates: Enable the delay module on output channel 104 (i.e. the analog output channel 5) with a delay value of 300ms.					
HPeq (Para	ameter Equalizer)					
<hpeq><e< th=""><th>Band&gt;</th></e<></hpeq>	Band>					
The param	eter equalizer is mainly used to adjust the specific frequency so that the overall frequency					
response o	of the signal tends to be flat.					
	no (in <hpeq></hpeq> ): Channel number, no=0~7 or 100~107.					
	<b>opt</b> (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					
Field	function of the corresponding subband), opt="1" (Enable the filter function of the corresponding subband).					
	<b>type:</b> Filter type, type="1" (Low Pass Filter), type="2" (How Pass Filter),					
	type="5" (Peak value equalizer).					
	frq: Frequency, frq=20~2000Hz.					
	wd: Q value, wd=2~5000.					
	>< (in <band></band> ): Subband gain value. The value range is [-240, 180].					
	<#Peq no="3" opt="1">					
	<band frq="32" no="0" opt="0" type="5" wd="/0">0</band>					
	<band no="1" opt="0" trq="80" type="5" wd="/0">U</band>					
	<band frq="200" no="2" opt="0" type="5" wd="/0">0</band>					
	<band frq="500" no="3" opt="1" type="5" wd="70">-90</band>					
Example	<band no="4" opt="0" trq="1250" type="5" wd="/0">U</band>					
	<band frq="3150" no="5" opt="0" type="5" wd="/0">0</band>					
	<band frq="8000" no="6" opt="0" type="5" wd="70">0</band>					
	<band no="/" opt="0" trq="12000" type="5" wd="70">0</band>					
	Check of the Company of the Company of the second section of the fourth subband as the need.					
	filter with a context frequency of 500Hz. O value of 0.07, and gain value of .0dR					
	Inter with a center frequency of 500Hz, & value of 0.07, and gain value of -90B.					
<hexp><th>1EXp&gt;</th></hexp>	1EXp>					
The expand	Jer is mainly used to compress the noise signals less than the threshold value so that the					
noise can t						
	opt: Enable/disable switch, opt="0" (disable), opt="1" (enable).					
	thrd: Threshold value, thrd=-600~0.					
Field	rate: Compression rate, rate=10~200.					
Field	wd: Knee width, wd=0~200, wd="0"(Hard knee).					
	atktm: Attack time, atktm=1~500ms.					
	atktm: Attack time, atktm=1~500ms.					
	<b>rlstm:</b> Release time, rlstm=1~10000ms.					



	<pre><hexp atktm="10" no="2" opt="1" rate="20" rlstm="100" thrd="-450" wd="0"></hexp></pre>					
Example	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 Compr	ressor is mainly used to compress signals larger than the threshold value, and control audio					
signals with	nin a certain threshold range.					
	no: Channel number, no=0~7.					
	<b>opt:</b> Enable/disable switch, $opt=0^{\circ}$ (disable), $opt=1^{\circ}$ (enable).					
	<b>rate:</b> Compression rate, $rate=10, 200$					
Field	wd: Knee width wd=0~200 wd="0"(Hard knee)					
	<b>atktm:</b> Attack time, atktm= $1 \sim 500$ ms					
	<b>ristm</b> : Belease time, ristm=1~10000ms					
	><: Makeup gain. The value range is [-120, 180].					
	HCom no="2" opt="1" thrd="-90" rate="20" wd="0" atktm="10"					
	rlstm="100">30 Indicates: Enable the Compressor function on input channel 2.					
Example	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.					
HAge (Auto Gain Control)						
<haac><!--ŀ</th--><th>, HAgc&gt;</th></haac>	, HAgc>					
The AGC is mainly used to control the signal to the target level.						
	no: Channel number, no=0~7.					
	<b>opt:</b> Enable/disable switch, opt="0" (disable), opt="1" (enable).					
	thra: Inreshold, thra=-60~0.					
Field	rate: Compression ratio, rate=10~200.					
	><: Maximum gain value. The value range is [0, 120].					
	<hagc lvl="-90" no="2" opt="1" rate="30" thrd="-50">60</hagc>					
Example	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)						
<pre><hlmt></hlmt></pre>						
The Limiter is a special case of the Compressor.						
	no: Channel number, no=100~107.					
	<b>opt:</b> Enable/disable switch, opt="0" (disable), opt="1" (enable).					
Field	thrd: I hreshold, thrd=-600~0.					
	ristm: Kelease time, ristm=1~10000ms.					
Evensele	<pre><hlmt no="102" opt="1" rlstm="100" thrd="-90"></hlmt></pre>					
⊏xample	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.					



AN DG-A00						
HUart (Uar	rt Parameter Configuration)					
<huart><!--</td--><td>HUart&gt;</td></huart>	HUart>					
It is used to	configure the parameters of the third-party control serial port, such as the baud rate.					
Field	<ul><li>&gt;&lt;: Baud rate. The value can be 300, 600, 1200, 4800, 9600, 14400, 19200, 38400, 56000, 57600, or 115200.</li></ul>					
Example	<pre><huart>9600</huart> Indicates: Set the baud rate of the serial port to 9600.</pre>					
HDev (Dev	ice Information Configuration)					
<hde><td>Dev&gt;</td></hde>	Dev>					
It is used to	configure the IP address and the name of the device.					
Field	<ul> <li>ip: New IP address mask:</li> <li>Subnet mask gtway:</li> <li>Gateway</li> <li>name: Device name, less than or equal to 20 char characters.</li> <li>(Note: Please make sure to input the correct information when configuring the IP address.)</li> </ul>					
Example	<pre><hdev gtway="192.168.1.1" ip="192.168.1.123" mask="255.255.255.0" 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.</pre>					
HSvCfg (S	ave Configuration)					
<hsvcfg>&lt;</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< td=""><td>e&gt;</td></hsvscnce<>	e>					
It is used to	deliver the command to save the current configuration to the corresponding scene.					
Field	no: Scene number, no=1~8.					
Example	<pre><hsvscnce no="7"></hsvscnce> Indicates: Save the current configuration to Scene 7.</pre>					
HRdScnce	e (Read Scene)					
<hrdscnc< td=""><td>e&gt;</td></hrdscnc<>	e>					
It is used to	p 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	e (Reset Scene)					
<hrstscnc< td=""><td>e&gt;</td></hrstscnc<>	e>					
It is used to	o reset the corresponding scene to the factory default status.					
Field	no: Scene number, no=1~8.					

<HRstScnce no="7"></HRstScnce>

Indicates: Reset the Scene 7 to the default configuration status.

Example



The BG-A88M supports RS-232 command control. Connect the RS-232 port of the product to a PC, then open a Serial Command tool on a PC to send ASCII commands to control the product. The ASCII command list is shown 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 F01 -> Invalid cor	s incorrect, the corresponding	error code is retur	ned. Error codes are	as following:			
E02 -> Parameter	rs error or out of range						
disable son	usage of the system is too higr ne functions properly before er	n, and no more fun Nabling new function	ictions can be enable ons.	ed. You need to			
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!	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	Expander						
s in x expd on threshold z!	Set the threshold, $x=1\sim 8$ , 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				

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Command Code	Function Description	Example	Feedback	Default Setting
s in x expd on	Set the attack time,	s in 1 expd on	input 1 expander	
attacktime z!	x=1~8,	attacktime 10!	is active and set	
	z=1~500		attacktime 10ms	
s in x expd on	$x=1\sim8$ ,	s in 1 expd on	active and set	
releasetime z!	z=1~10000	releasetime 100!	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)				
	Set the filter type,	s in 1 peg on	input 1 pea is	
s in x peq on	x=1~8, b=1~8, z=1(LP), 2(HP), 5(Peak)	band 1 on type 5!	active and set	
band b on type z			band 1 filter to	
			peak	
s in x peq on	Set the frequency, x=1~8, b=1~8, z=20~20000	s in 1 peq on band 1 on frq	input 1 peq is	
band b on frq z!			active and set band	
		1000!	1 frequency	
	Set the Q value.			
s in x peq on	x=1~8,	s in 1 peq on	Input 1 peq is	
z!	b=1~8,	21	band 1 wide 0.02	
	z=2~5000	<i>L</i> .		
s in x peq on	x=1~8,	s in 1 peq on	input 1 peq is	
band b on gain	b=1~8,	band I on gain	active and set	
2:	2=-240~100	-00:	-6.0dB	
s in x neg on	Disable the subband.	s in 1 peg on	input 1 peq is	
band b off!	x=1~8,	band 1 off!	active but band 1	Disable
	D=1~8 Disable the Equalizer		is unactive	
s in x peq off!	$x=1\sim8$	s in 1 peq off!	unactive	Disable
Compressor	Cat the threehold		input 1 comproses	
s in x comp on	x=1~8,	s in 1 comp on	is active and set	
threshold z!	z=-600~0	threshold -100!	threshold -10.0dB	
s in x comp on	Set the compression rate, x=1~8, z=10~200	s in 1 comp on rate 20!	input 1	
rate z!			compressor is	
			active and set rate 2.0	
s in x comp on	Set the knee width,	s in 1 comp on	input 1 compressor	
knee z!	z=0~200	knee 0!	is active and set	
	Set the makeup gain			
s in x comp on	x=1~8.	s in 1 comp on	is active and set	
gain z!	z=-120~180	gain 30!	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	input 1	
		attacktime 10!	compressor is	
			active and set	
	Sat the release time		input 1 compressor	
s in x comp on	$x=1\sim8$ .	s in 1 comp on	is active and set	
releasetime z!	z=1~10000	releasetime 100!	releasetime 100ms	
s in x comp off!	x=1~8	s in 1 comp off!	is unactive	Disable
Auto Gain Control				
s in x agc on	Set the target level,	s in 1 agc on	Input 1 agc is	
target z!	x=1~8,  z=-200~0	target -90!	target level -9.0dB	



Command Code	Function Description	Example	Feedback	<b>Default Setting</b>
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 Ivl 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 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 Ivl 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 Ivl 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!	[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	

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	Enable the noise	s afc ans on	ans under afc is	
level z!	suppression function,	level 12!	active and	
	z=6~18		suppress noise 12dB	
s afc ans off!	Disable the noise suppression function.	s afc ans off!	ans under afc is unactive Disable	
Adaptive Echo Ca	ancellation			
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,	s aec ref hit 3!	aec ref hit 3	
s aec ret remove	X=1~8 Remove the reterence signal	s aec ret remove		
x!	x=1~8	3!	aec ref remove 3	
s aec on nln zl	Set NI P level z=1~3	s aec on nin 2l	aec is active and	
			nlp set 2	
s aec off!	Disable AEC	s aec off!	aec is unactive Disable	
s aec ans on	Enable the noise	s aec ans on	active and suppress	
level z!	suppression function,	level 12!	noise 12dB	
	z=6~18			
s aec ans off!	suppression function.	s aec ans off!	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 zl	Set the noise suppression	s ans on level	ans is active and	
	level, z=6~18	12!	suppress noise	
	Dischle ANO	s ans off!	aec is unactive	Diachla
s ans on	Disable ANS.			Disable
r ans!	status.	r ans!	ans is unactive	
Equalizer (Output)				
s out v peg on	Set the filter type,	s out 1 peg on	output 1 peg is	
band b on type	y=1~8,	band 1 on type	active and set	
z!	b=1~8,	5!	band 1 filter to	
	z=1(LP), 2(HP), 5(Peak)		peak	
	Set the frequency,	sout 1 peg on	output 1 peg is	
s out y peq on	y=1~8,	band 1 on fro	active and set hand	
band b on frq z!	b=1~8,	1000!	1 frequency	
	z=20~20000		1000Hz	

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Command Code	Function Description	Example	Feedback	<b>Default Setting</b>
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!	ut 1 peq on d 1 off! d 1 off! d 1 off! d 1 off! d 1 off!	
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! uctive and set releasetime 100ms		
s out y Imt off!	Disable the Limiter, y=1~8	s out 1 Imt 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 lpt 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!	is 18%	

2



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

- Is the STATUS LED normal?
- Is the INPUT/OUTPUT VOLUME turned down to the lowest level.
- Are the cables in the right ports?
- Is the matrix mixer configured correctly?
- Are the audio modules applied correctly?

If echo persists even when the processor is working check the following:

- Is the processor functioning normally?
- Is the AEC module activated?
- Is the matrix routing correctly configured?
- Is the connection to the Amplifier or Speaker for the output reference signals consistent with the matrix routing?

# **Application Example**

The audio processor supports microphone inputs, computers, BluRay players, mixers, and other audio source inputs. It can be used in combination with a video conference terminal, recording and broadcasting system, or a mixing console. The following figure shows a general connection diagram. Users can connect the devices based on actual requirements.





# **Tech Support**

Have technical questions? We may have answered them already!

Please visit BZBGEAR's support page (<u>bzbgear.com/support</u>) for helpful information and tips regarding our products. Here you will find our Knowledge Base (<u>bzbgear.com/knowledge-base</u>) with detailed tutorials, quick start guides, and step-by-step troubleshooting instructions. Or explore our YouTube channel, BZB TV (<u>voutube.com/c/BZBTVchannel</u>), for help setting up, configuring, and other helpful how-to videos about our gear.

Need more in-depth support? Connect with one of our technical specialists directly:

<b>Phone</b>	Email	Live Chat
1.888.499.9906	support@bzbgear.com	bzbgear.com

# **Limited Product Warranty Terms**

Pro Line: 5-year warranty from the date of purchase for AV/Broadcasting products bought on or after August 1, 2024.

Essential Line: 3-year warranty from the date of purchase for AV/Broadcasting products bought on or after August 1, 2024.

Cables: Lifetime Limited Product Warranty.

For complete warranty information, please visit bzbgear.com/warranty.

For questions, please call 1.888.499.9906 or email support@bzbgear.com.

# **Mission Statement**

BZBGEAR is a breakthrough manufacturer of high-quality, innovative audiovisual equipment ranging from AVoIP, professional broadcasting, conferencing, home theater, to live streaming solutions. We pride ourselves on unparalleled customer support and services. Our team offers system design consultation, and highly reviewed technical support for all the products in our catalog. BZBGEAR delivers quality products designed with users in mind.

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