AudioGridController

User Guide

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I. Overview

AudioGridController is a software product developed by Shenzhen Audiocom Technology Co., Ltd. for operating its A-series and B-series network audio transmission modules. It is compatible with widely used related products and is designed for use within the same network environment.

In addition to basic audio routing subscription, multicast settings, and device parameter configuration, AudioGridController can additionally display the clock status and the volume, that is convenient for the you to observe the operating status of the device.

Through AudioGridController interface, the you can operate the input and output mixers built into the module. Set and save up to 16 scenes according to your needs, and call them up at any time.

In addition to AG multicast, both A-series and B-series modules are compatible with the AES67 protocol and can communicate with widely popular AES67 devices in the market. AudioGridController helps you to create, delete, and interoperate with other devices that support this protocol.

Using AudioGridController, you can back up all the data of the module, including the configuration of the module, audio subscription information, settings for the multicast stream, etc., and restore them when appropriate.

The AudioGridController software needs to be added to the firewall whitelist.
For A series modules, the you can configure the relevant parameters of the module directly through the web interface.
In terms of language, AudioGridController supports both Chinese and English interfaces. By default, it is an automatic mode, and AudioGridController automatically and seamlessly selects the appropriate language display according to the operating system. Users can additionally specify the display language directly.

II. Terminology

Before using the AudioGridController and our products, you need to clarify a few basic concepts:

1. Channel. One channel corresponds to one audio channel. As many channels as the module supports, you can transmit as many channels of audio as you can at the same time. The number of transmit and receive channels of the module can be different.

2. Flow. A flow is a container for some channels, and any transmission of channel data must be included in the flow. A flow can contain multiple channels. The subscription and cancellation of channels between all modules are flow-based operations. Flow has unicast and multicast, each unicast flow can contain 1-4 channels, and a multicast flow can contain up to 8 channels.

3. Unicast flow. According to the set audio route, the audio data is transmitted between two devices on a one-to-one basis, and this data stream is called a unicast flow. It is characterized by the fact that it arises with the creation of a route, and if the route is canceled, the data flow disappears. For switches, unicast streams are forwarded from one port to another, without increasing the network reception pressure on other devices.

4. Multicast flow. If a transmitting device does not have a specific receiver, but instead transmit an audio stream to a specific multicast address and port, it is called a multicast flow. The multicast flow exists after it is created and is deleted on the transmitting device. It doesn't matter if there are or how many receivers there are.

Multicast flows are used when multiple devices receive the same channel from the same transmitting device. Suppose that there are 10 receivers that each receive a number of channels from one transmitter. If unicast streams are used, the sender needs to create 10 unicast streams and send them to each receiving device, and the total traffic on the network will be large. If a multicast stream is used, the sender only needs to create a single multicast stream that includes these channels to meet the requirements. Only one multicast stream exists on the network, which greatly reduces the total network traffic.

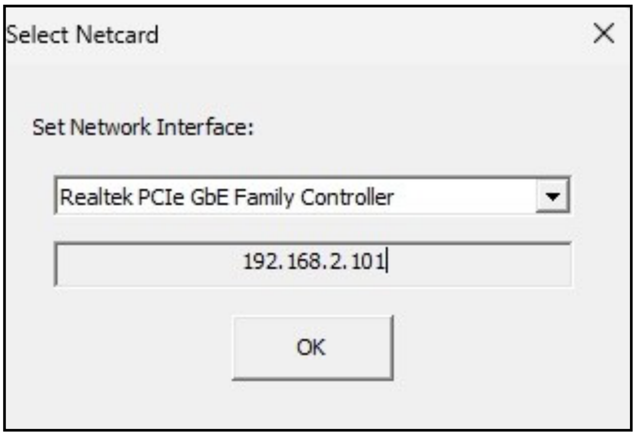
5. Subscription. Subscription refers to the operation of having a channel of a receiving module receive audio data from a channel of the transmitting module. After the operation is complete, the information of the sending channel is stored in the relevant location of the receiving device in the format "channel name@sending device name".

III. Basic operations

Basic operations include connection settings, device discovery, channel subscription (cancellation), using filters, sorting, volume checking, and modifying channel names.

1. Connection settings

Before using AudioGridController, you need to configure the network interface for AudioGridController to ensure that all modules and the computer where the control software is located are on the same network, so that AudioGridController and modules can communicate properly. Extract the AudioGridController software to any directory and double-select it to open AudioGridController. When it is run for the first time, it will automatically and seamlessly pop up the network settings window, and the you only needs to select the network card that is actually connected to the audio network, and after choosing, close AudioGridController. When you turn it on again, you can use the network interface you just set to communicate.



This software supports wireless network, and you can control all online devices through wireless network. When using Wi-Fi, do not use a virtual sound card.

2. Device discovery

After setting up the network interface, open AudioGridController again, and all online devices and the multicasts will be displayed on the subscription screen in turn.

To make it easier to distinguish between different types of devices, AudioGridController uses different colors to display the device name. The A-series devices use a black display, and the B-series devices use a green display. Multicast streams are shown in blue with a flow type prefix such as AES67 or AG.

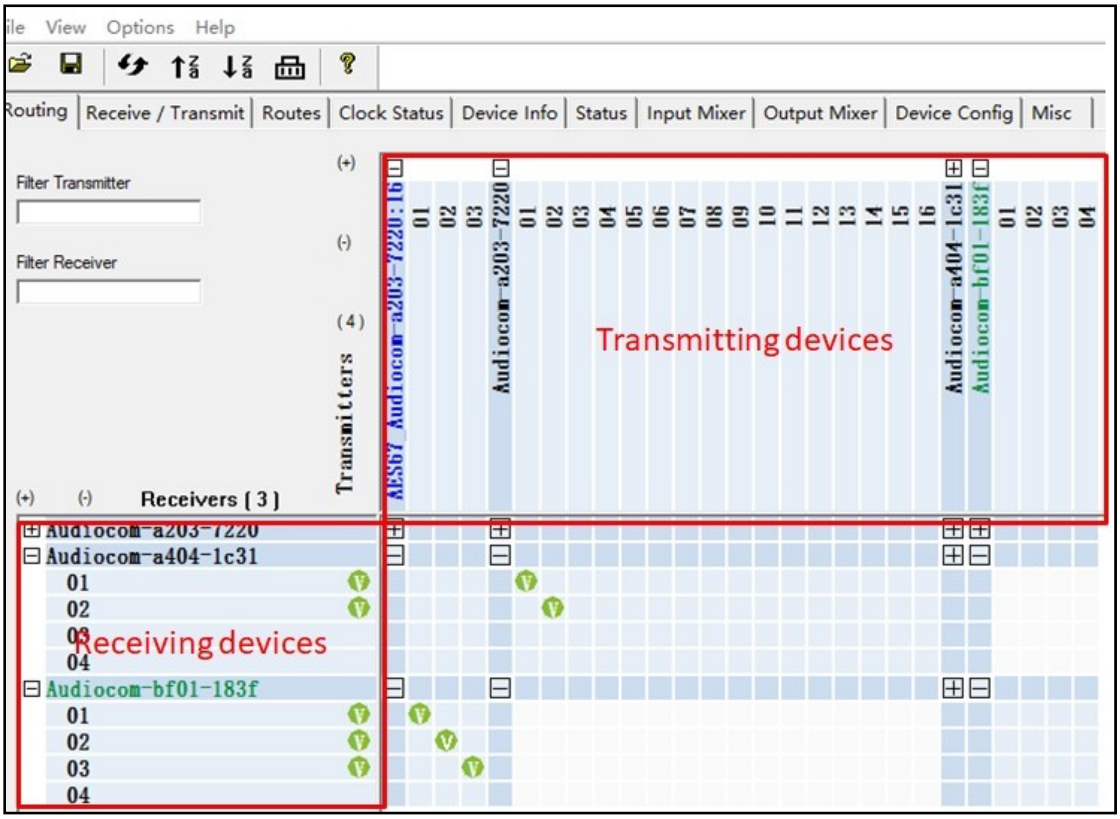
Devices that are powered off or disconnected from the network are automatically and seamlessly deleted.

Modules and multicast streams that contain send channels are displayed in the list of transmiting modules above.

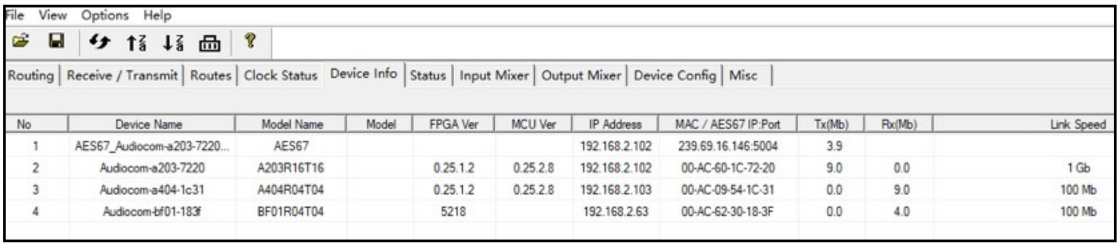
The modules that contain the receive channel are displayed in the list of receive modules on the left.

Modules that include both transmit and receive channels are displayed in the list of transmit and receive modules.

It should be noted that because the multicast information of AES67 is broadcast every 15-30 seconds, the AES67 stream may be delayed for a period of time before it is displayed, that is normal.



If you want to view the name, model, IP address, software version, and other related information of the module, select the Device Information tab. The MAC/AES67 column displays the MAC address of the physical device or the IP address and port of the multicast stream. The you double-clicks the box in this column, and the corresponding content will be copied to the clipboard for the you's convenience.

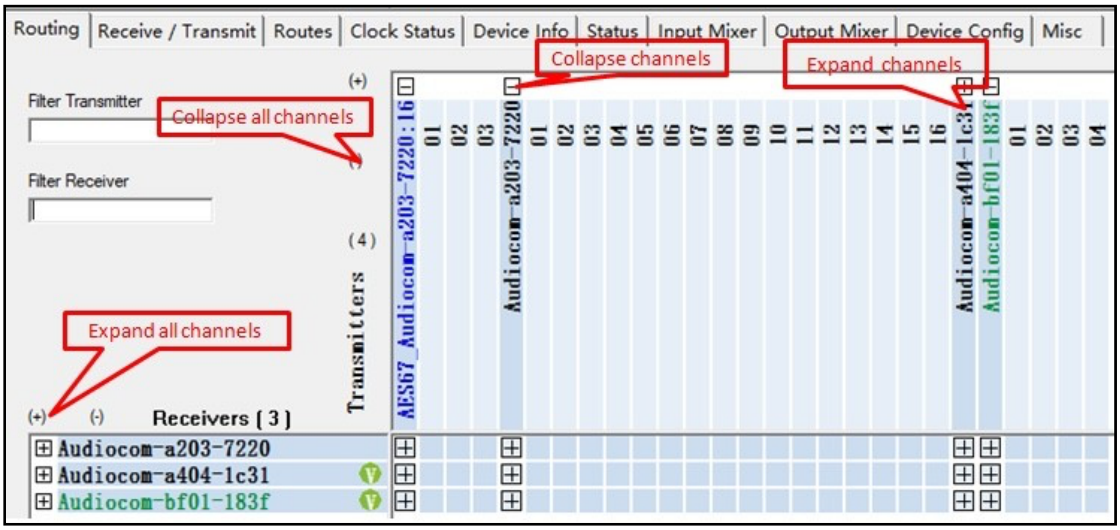


No	Device Name	Model Name	Model	FPGA Ver	MCU Ver	IP Address	MAC / AES67 IP:Port	Tx(Mb)	Rx(Mb)	Link Speed
1	AES67_Audiocom-a203-7220...	AES67				192.168.2.102	239.69.16.146:5004	3.9		
2	Audiocom-a203-7220	A203R16T16		0.25.1.2	0.25.2.8	192.168.2.102	00-AC-60-1C-72-20	9.0	0.0	1 Gb
3	Audiocom-a404-1c31	A404R04T04		0.25.1.2	0.25.2.8	192.168.2.103	00-AC-09-54-1C-31	0.0	9.0	100 Mb
4	Audiocom-bf01-183f	BF01R04T04		5218		192.168.2.63	00-AC-62-30-18-3F	0.0	4.0	100 Mb

3. Channel subscription and cancellation

Above the name of the transmitting module and to the left of the name of the receiving module, there are channel expansion and collapse buttons for turning the channel name display on and off. When the mouse is slid over the receiving channel, the corresponding receiving information is displayed on a floating window.

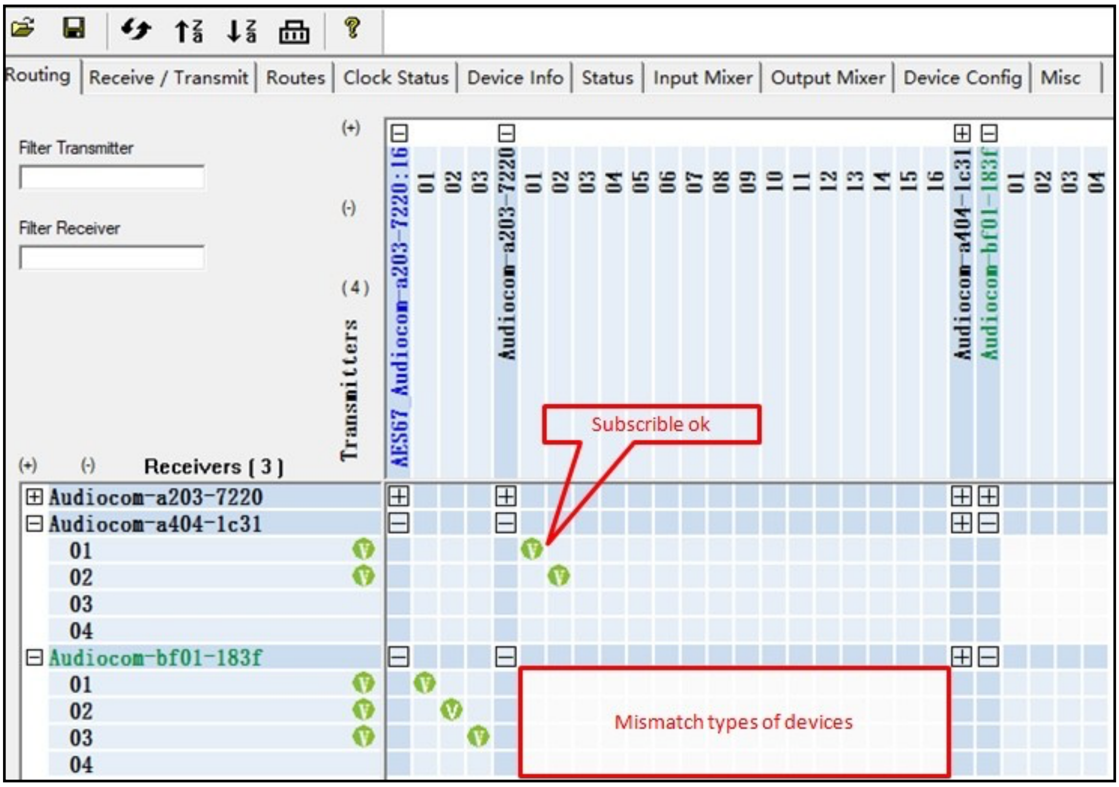
The (+) (-) symbol next to the filter is used to turn the channel display on and off for all modules, and the number after it is the number of transmit (receive) modules currently displayed.



Subscribe to a channel. After opening the channel of the transmitting device and the receiving device, select the grid on the intersection of the transmitting channel and the receiving channel with the mouse to send the sound of the channel of the transmitting device to the corresponding channel of the receiving device.

Cancel subscription. Click in the grid where the green checkmark appears. The green symbol disappears, the subscription is canceled and the sound of the channel is no longer received.

For the receiving channel, if the corresponding transmitting device is disconnected after the subscription is successful, a yellow F will be displayed on the right side of the channel, indicating that the subscription has failed.



If the grid for some channel intersections is gray-white and clicking does not work, the channel between the two modules is not subscribeable. There are several reasons for this:

(1) Mismatch between modules. For example, the A series and the B series cannot communicate directly with each other. Interworking can only be done by creating AES67 multicast.

(2) The module itself does not support self-configuration. Some models do not support receiving their own transmit channels, and their transceiver channel crosspoints cannot be operated.

(3) The module cannot receive AES67 multicast transmitted by itself.

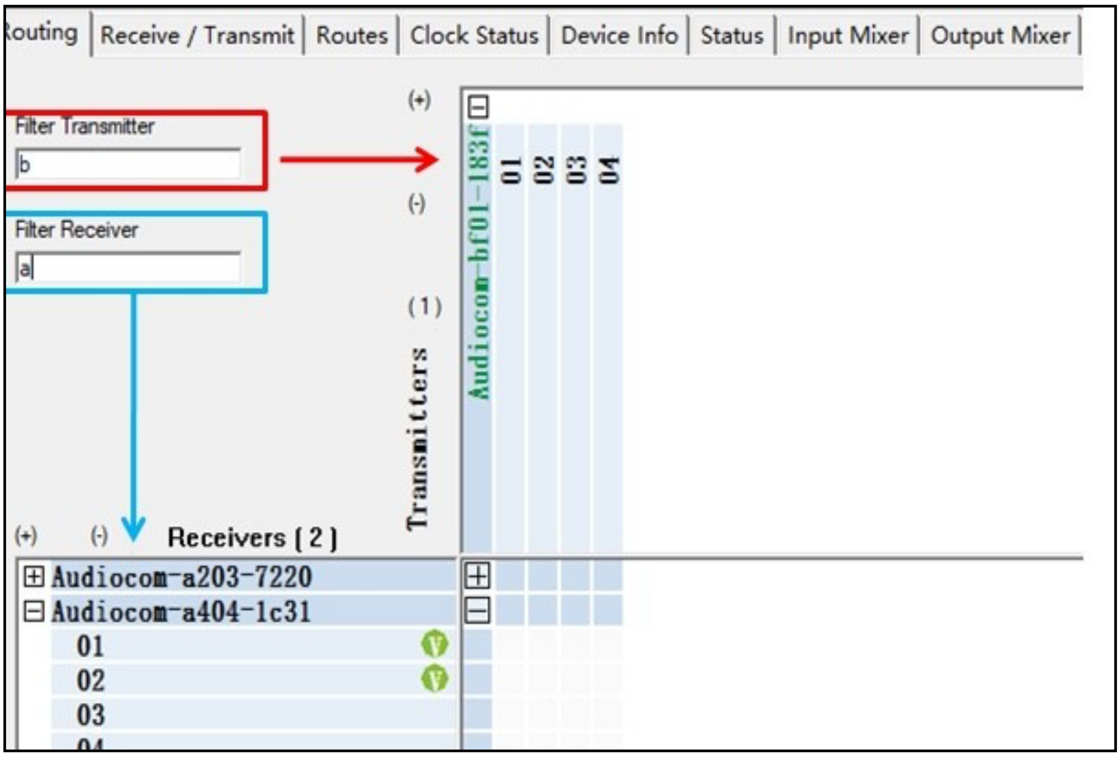
(4) The sampling rate and number of coding bits of the two sides are inconsistent. After the device is online, AudioGridController will query the configuration data of the module, but if the modules of some manufacturers do not respond to the query, AudioGridController will not be able to obtain their configuration information, resulting in the inability to subscribe. Users can disable the sample rate and number of code bits check through the two options under the Options menu: Match Sample Rate and Match Number of Coded Bits. After refreshing, you can subscribe to the operation, and at this time, you need to ensure that the parameters configured by both parties are consistent in order to output normal sound.

Double-select the device name on this page to set the device as the current device and go to the volume display interface of the transmitting and receiving channel. Other pages, including input mixes, output mixes, and device parameter configuration interfaces, are automatically and seamlessly switched to the device.

4. Using filters

In the upper left corner of the subscription interface, there are two filters, transmitter and receiver, which are mainly used to filter out modules that do not contain specified characters in the device name. The transmitter filter is for the transmitting device, and the receive filter is for the receiving device. When there are many devices, you can use filters to display only devices whose device names contain specified character strings, improving operation efficiency.

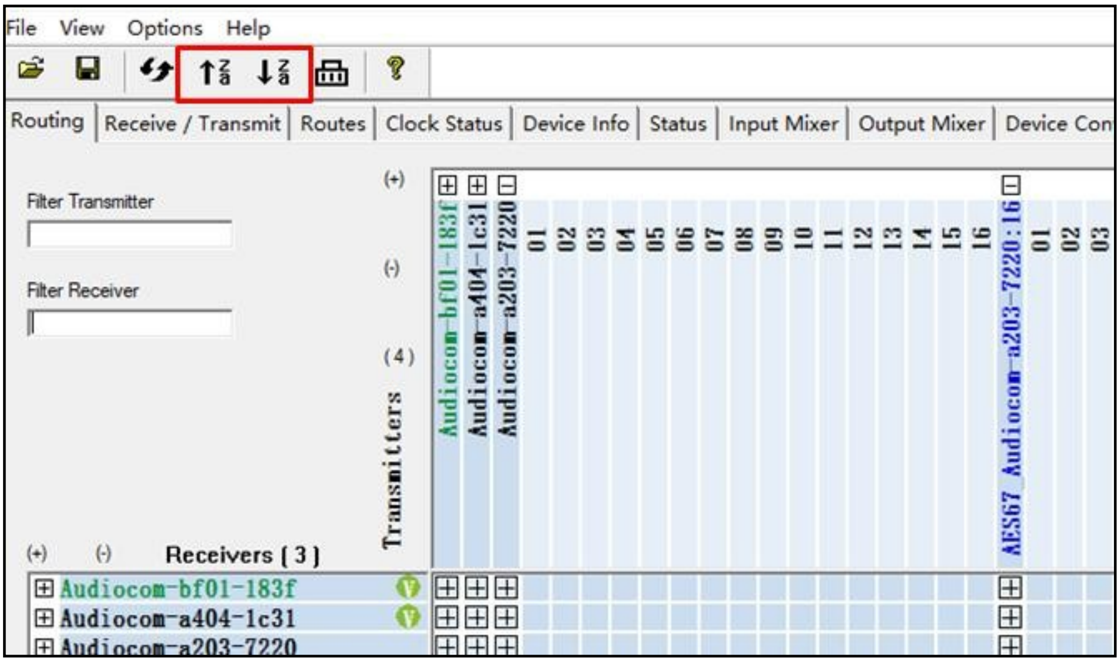
If you enter the character A in the transmitting filter, only the module with A in its name can be displayed in the list of transmitting devices, and all operations of the you can only be performed on these devices. The same is true for receiving filters.



5. Sorting

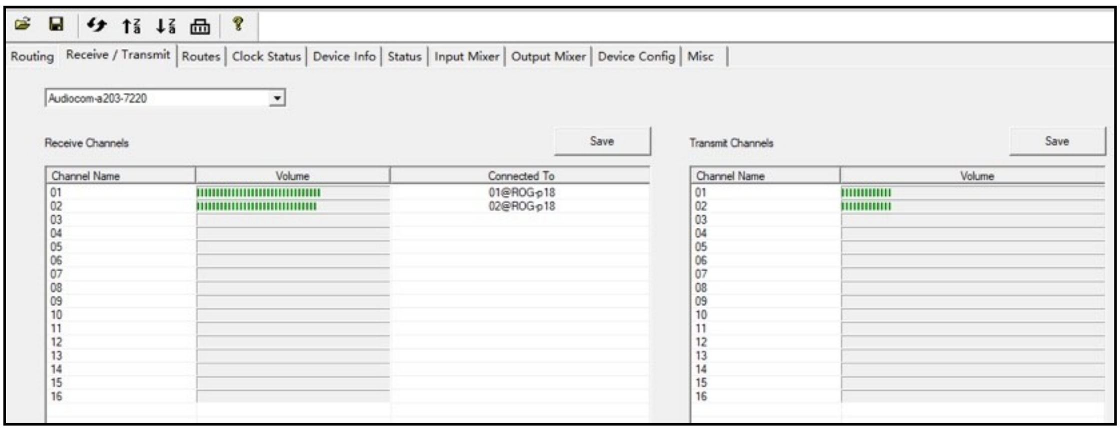
On the toolbar menu, there are 2 sort buttons, which display modules in ascending and descending order according to the module name. Users can use them as needed, and all displayed devices are sorted by name in ascending or descending order to make it easy to find devices.

You can select the header to sort the device by device name, model, FPGA version, MCU version, IP address, and network traffic.



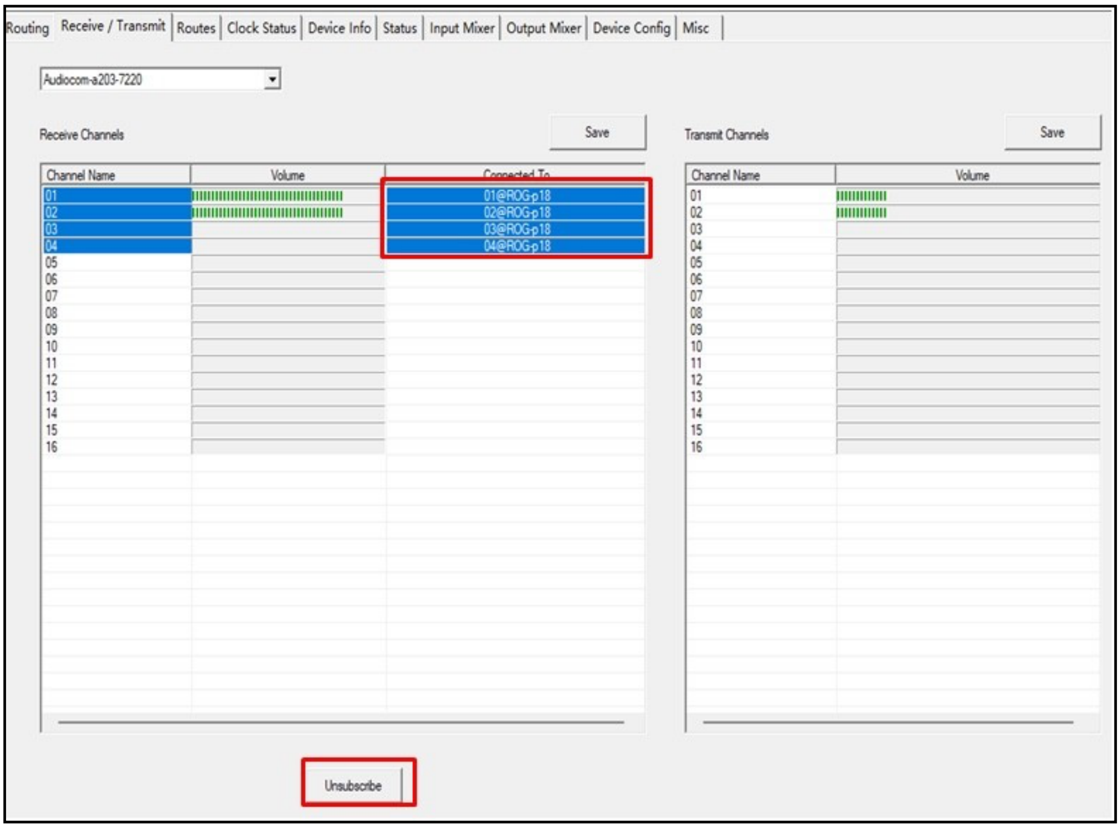
6. Volume check and batch unsubscribe

After the channel subscription is successful, the you can observe the channel volume of the sending and receiving modules and check the effect in real time. On the channel subscription page, double-select the name of the receiving device to go directly to the corresponding sending and receiving channel interface of the device. You can additionally select the Transceiver Channel tab, open the drop-down list, and select the desired device. On the left side is the receiving channel of the current device and the name of the sending channel of the subscription, and on the right side is the sending channel. The green column graph represents the real-time volume of the channel.



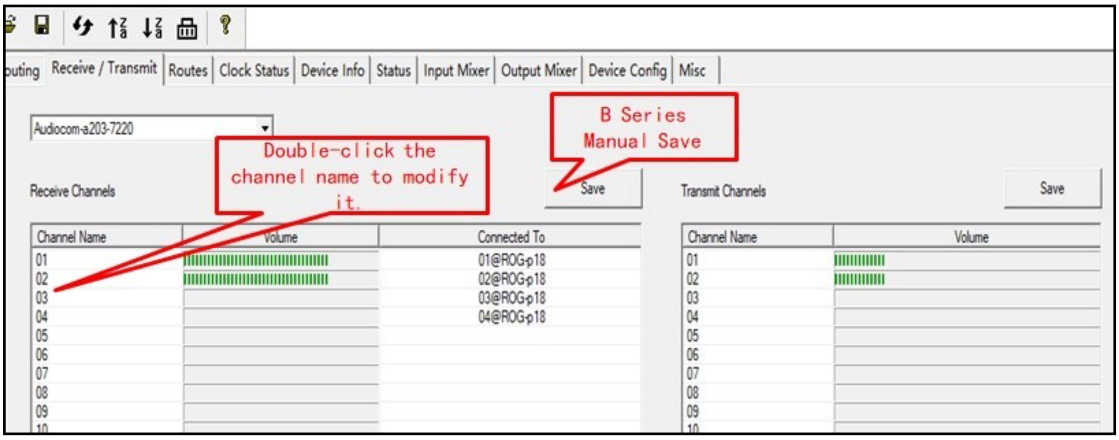
In addition to the volume indicator, users can also unsubscribe from channels in bulk within this page (in the Routing page, you can only unsubscribe one by one).

The method is to hold down the shift key, select the receiving channel you want to cancel in turn, and after the selection is completed, select the unsubscribe button to complete the batch cancellation.



7. Modify the channel name

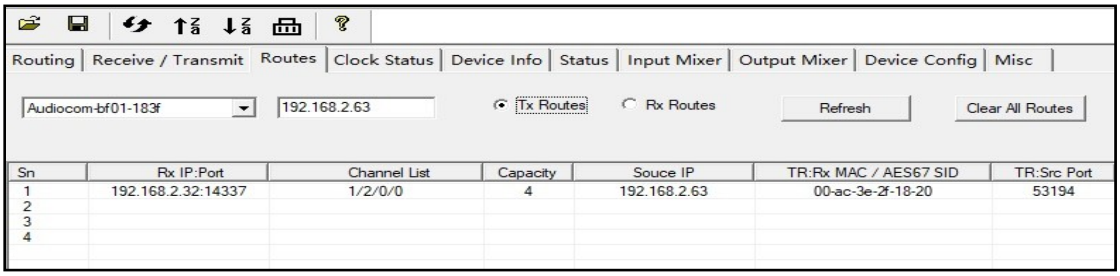
On the transceiver channel page, in addition to displaying the volume of reception and transmission, you can additionally modify the channel name by double-clicking on the channel name, and the ENTER key confirms after the modification is completed. The channel name can only use letters, digits, or hyphens, and other characters cannot be entered. The maximum length of the channel name of the A series is 31 characters, and it is automatically and seamlessly saved after the modification is completed. The maximum length of the B-series channel name is 15 characters, and after the modification is complete, select the Save button at the top to save the modification result.



For B Series, after the channel name is modified, if the sender is online, the subscription route is displayed in the format of channel name @ device name. When the sender is not online, it is still displayed as a numeric @IP address.

IV. Routes

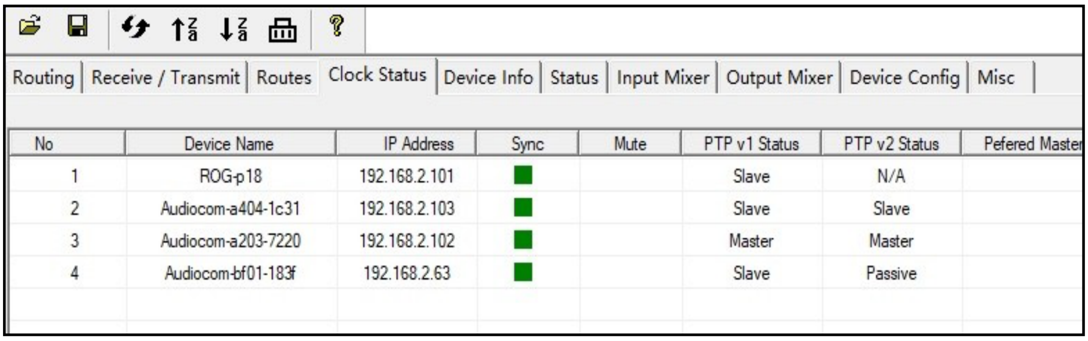
The routes page is used to operate B-series transceiver routes. Only B-series devices will appear in the device list on this page. When a device is selected, the page lists all the send and receive routes for the device. Depending on the need, one or all routes can be cleared. Double-select in the sequence number column to clear the route. Clicking Clear All Routes clears all transmitting / receiving routes for that device.



V. Clock status and setting

1. Status

The Clock Status page lists the clock status of all modules in the network, including Sync, Mute, PTPv1, PTPv2, and whether the Preferred Master or Slave Only options are set.



No	Device Name	IP Address	Sync	Mute	PTP v1 Status	PTP v2 Status	Pefered Master
1	ROG-p18	192.168.2.101			Slave	N/A	
2	Audiocom-a404-1c31	192.168.2.103			Slave	Slave	
3	Audiocom-a203-7220	192.168.2.102			Master	Master	
4	Audiocom-bf01-183f	192.168.2.63			Slave	Passive	

The sync status is displayed as an icon, and the device where the sync is completed is displayed as an icon . To continue, sync the device marked with the icon .

Mute status to icon , it means the device is on mute, usually due to incomplete clock synchronization or failure to recognize master clock synchronization information
The two columns of PTPv1 and PTPv2 respectively display the status of the device under the two clock protocols, and the possible states are as follows:

(1) Master, the device is the master clock of v1 or v2 in the network;
(2) Slave, the device is the slave clock of v1 or v2 in the network;
(3) Passive, the device does not listen to the clock synchronization information;
(4) Listening, PreMaster, Initializing, and Fault are usually set. When the standby is started or during the adjustment of the master clock, it means that the device is in a state change, that is a temporary state and will disappear soon;
(5) Uncalibrated, indicating that the device cannot recognize the master clock synchronization information of the corresponding version;
(6) Disable, this flag usually appears in the PTPv2 column, indicating that the PTPv2 protocol of the device is in a forbidden state;
(7) N/A, This flag usually appears in the PTPv2 column, indicating that the device does not support the PTPv2 protocol.

2. Settings

Master and slave clocks usually do not need to be set by the you. In the case of good network communication, the master clock device is automatically and seamlessly selected among multiple devices. Other devices automatically and seamlessly synchronize with the master clock device based on the synchronization information sent by the master clock device.

If you need to prefer a device as the master clock for some reason, you can double-select in the box corresponding to the device in the Force Master Clock column, and the flag PM will be displayed in the box, indicating that the device is in the priority clock state. If only one device in the network has this flag and is in the Slave state, it will be converted to the master clock immediately. If there are multiple devices in the network with PM flags, they are still in competition and will generate a master clock according to the corresponding rules

In addition to setting the priority master clock flag, the A series module and B series module produced by our company also support slave clock setting only, that is, the module only does slave clock. The setting method is to double-select in the box corresponding to the device under the column of only doing slave clock, and the box displays the SO logo, indicating that the device only does slave clock.

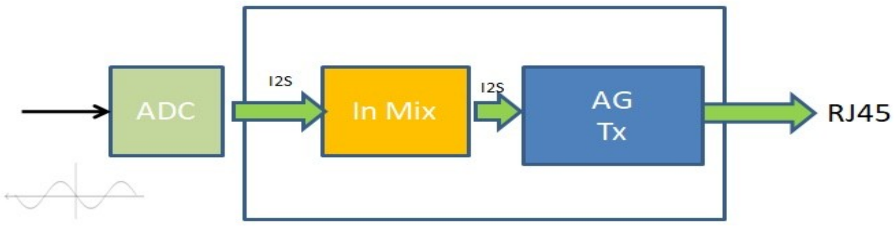
The PM or SO flag is both on/off state. Double-select again in the box where these two flags already exist, the logo disappears and the device returns to its normal state. The PM or SO flag is automatically and seamlessly saved after setting, and the set state is still maintained when the power is turned on again.

It should be noted that the device is delayed, and if the power is off immediately after setting, it may not be saved in time. To avoid this problem, the power should usually be turned off at least 10 seconds after setting.

VI. Input mix (before transmitting)

In addition to the exchange of audio channels, the A/B series module has an input mixing matrix and an output mixing matrix before transmitting and after receiving, which is suitable for use by you according to their needs.

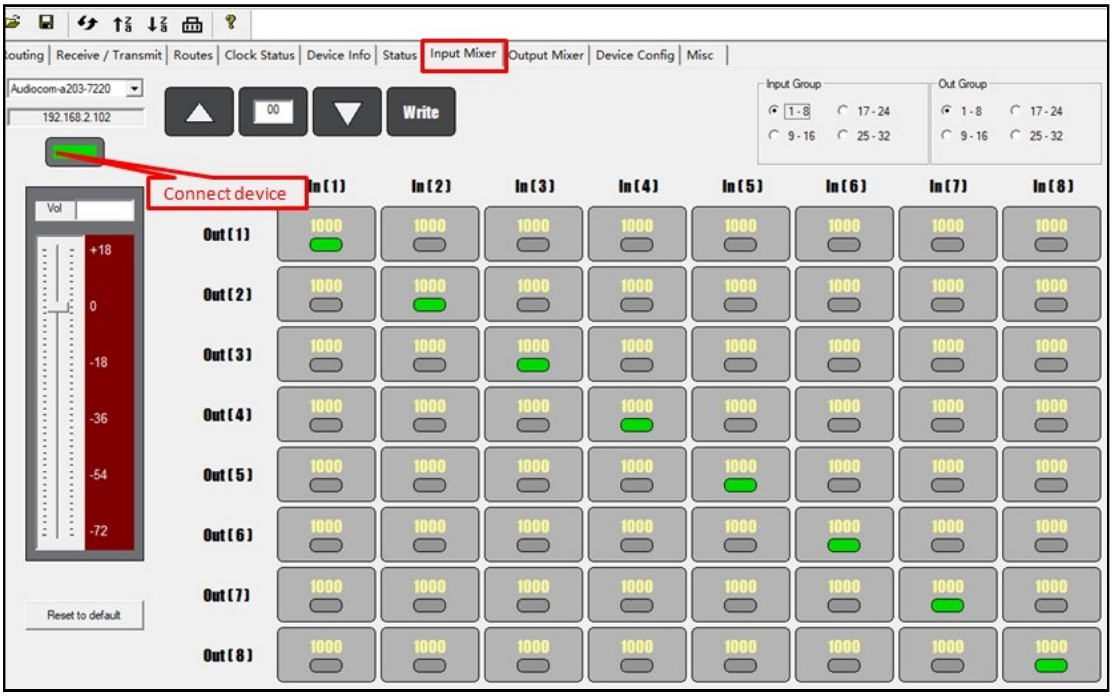
The input mix sits between the ADC (DSP) and the transmit channel.



Click the Input Mixer tab to enter the input mixing interface. Click the drop-down box to find the module you want to set, and then select the gray connection button, the button turns green, that is, the corresponding module is successfully connected.

Below the connect button are the volume fader and noise gate settings for setting the mix volume and noise gate.

The Reset to default button is used to restore the default values of the input mixmette and noise gate.



Scene call and data save buttons

The four buttons are the Previous Scene, Scene Number, Next Scene, and Save button. Data for browsing and recalling pre-saved mix settings and saving the current scene settings. One scene corresponds to a set of settings, and there are a total of 16 scenes for you to use.

These two sets of radio buttons are input and output group selectors. Due to the limited space in the interface, only 8 audio inputs and 8 transmitting channels can be displayed at a time. With different combinations of group selectors, all channels of the module can be displayed and set.

The vertical representation in the diagram shows each audio channel fed into the module by the ADC (or DSP) and ready to be transmitted over the network. The input audio is arranged from left to right, and the number of channels represented is also determined by the input group selector. Routes 1-8 are shown in the figure.

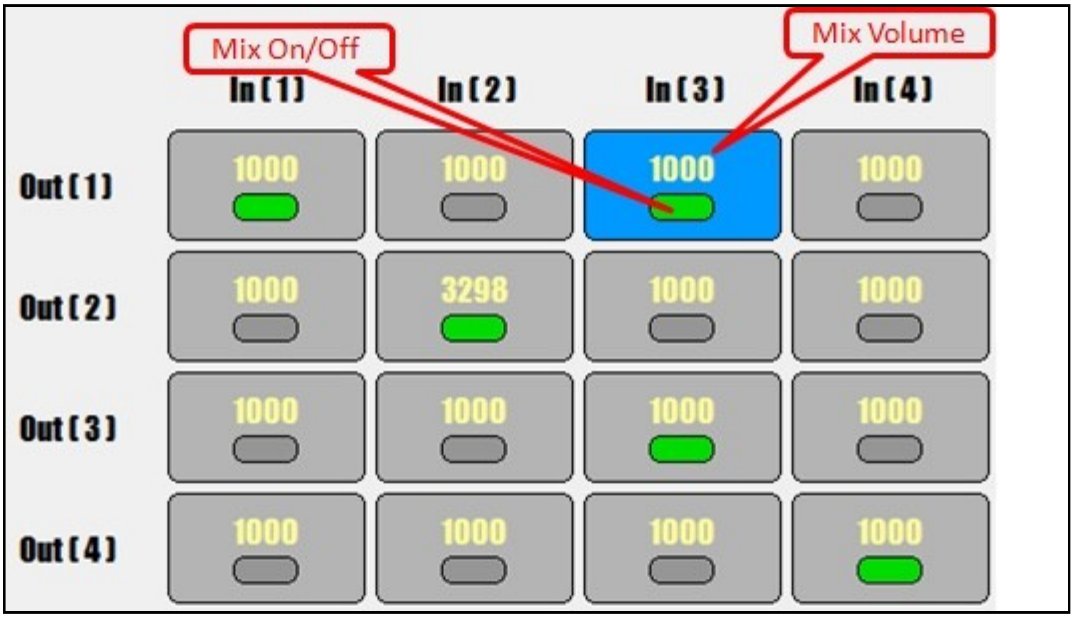
The horizontal represents the transmit channel, from top to bottom, there are 8 in order, which 8 channels are determined by the output group selector in the upper right corner, as shown in the figure 1-8.

The intersection of horizontal and vertical is the operating unit of whether an audio channel is mixed into a transmit channel. The input/output correspondence is shown above and to the left of the cell, with In3/Out1 indicating that the unit is operating on the 3rd audio and transmit channel 1. When the mouse clicks left, the unit turns blue, that is selected. There are two types of operation: the mix switch and the mix volume setting.

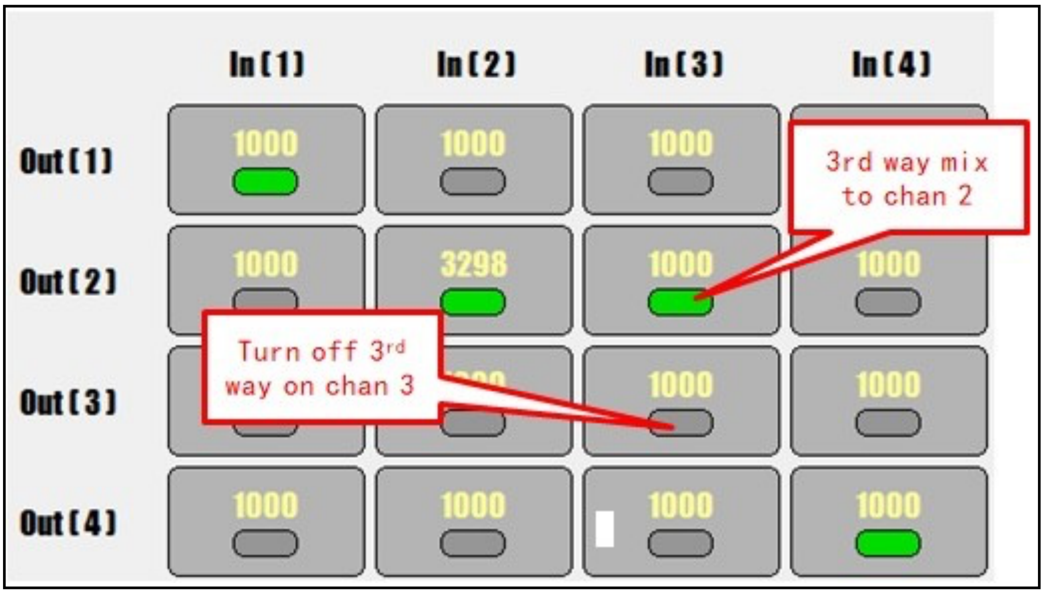
The elliptical bar inside the unit is a mix switch, which can be clicked to switch between green and gray. Green means that the mix switch is turned on, and the audio is mixed into the send channel to be transmitted, and gray means that the mix switch is turned off.

The number above the switch indicates the mixing volume. Expressed in hexadecimal system. Once the cell is selected, you can adjust the volume using the volume fader on the left, or you can enter a number directly in the edit box above the fader (confirm with enter). The dB value displayed on the scale, 0 dB is mixed according to the original volume (corresponding to the digital 0x1000), and the maximum can be mixed by +18dB. 0x1000 is the default setting, and the original volume is mixed into the send channel. The A series module needs to select the Write button to save, and the B series will be automatically and seamlessly saved.

By default, the module uses scenario 0, where there is a one-to-one correspondence between the input mix and the send channel, that is, the first audio is sent to the send channel 1, the second audio is mixed into the send channel 2, and so on.

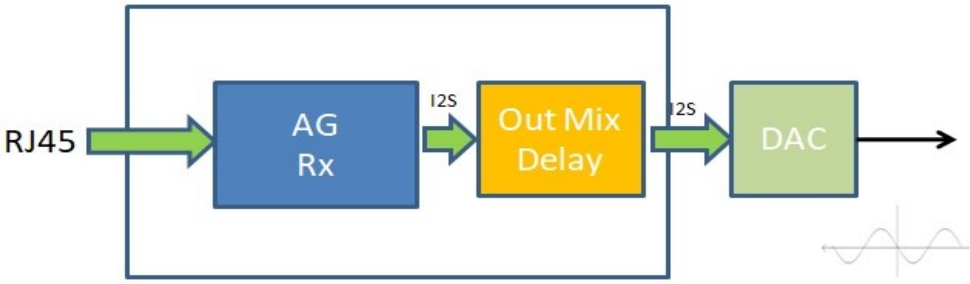


Suppose the you feeds in four channels of audio data, which are called the 1st, 2nd, 3rd, and 4th channels, where the 1st and 4th channels are sent in channels 1 and 4 respectively, and the 2nd and 3rd channels are mixed and sent in the second channel. In addition to turning on the mix switches on 1st, 2nd, and 4th, you also need to turn on the mix switches on the 3rd and transmit channels 2 to turn off the default 3rd mix switch. The following figure shows the settings after completion.



VII. Output mixing (after receiving)

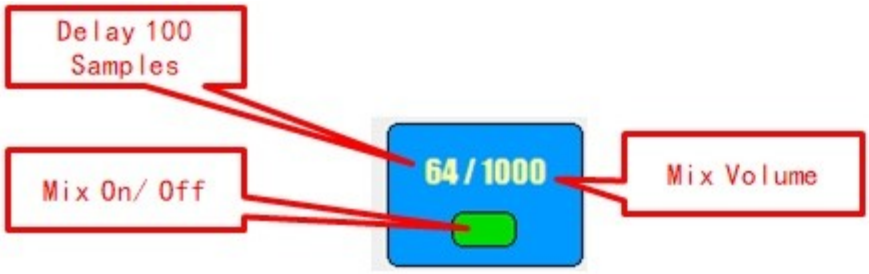
The output mix matrix is after the receiving channel and before the DAC (Output DSP).



The usage method is basically the same as the input mix, and there are 7 action buttons:

The first 4 are the same as the input mix, that is used to recall different scenes and save the data of the current scene.
Open is used to open the scene data saved on the computer.
The Download button downloads the scene data you just opened into the module.
The Save button saves the output mix data of the currently connected module to your computer.

In addition to the mix switch and mix volume, the output mix has a delay setting, and each channel can be delayed for a period of time before being mixed into the output channel and sent to DAC orOutput DSP.



The delay value needs to be entered via the delay text box above the volume fader. The value is hexadecimal, that is the number of samples for this channel delay. 0x64 is the time corresponding to the delay of 100 samples in this channel, and then the sound is mixed to the corresponding output channel. The delay time corresponding to 100 samples is related to the sampling rate. At a 48K sample rate, the latency is 100/48000 seconds, or 2.08 milliseconds. If the sample rate is 96K, the corresponding delay time is 1.04 ms.

The mixing volume can be adjusted in the same way as an input mix, and can be adjusted via the volume fader or directly by entering the numbers.

By default, the receive channel and output mix are one-to-one, as shown in the figure below, with the control units turned on by the mix switch diagonally distributed.

VIII. Multicast

1. Relevant knowledge

As the name suggests, multicast is communication with some devices in the network. Multicast is relative to unicast and broadcast.

Unicast communicates with a specific destination host, and the destination IP address and destination MAC address are both used by the destination host.

The broadcast is to communicate with all devices in the local area network, and the destination IP is 255.255.255.255, and the destination MAC is 6 0xFF.

Multicast is somewhere in between and uses a specific range of addresses as the destination IP, with the first 3 bytes of the destination MAC address being a fixed 0x01, 0x00, 0x5E, and the last 3 bytes being the same as the one usedMulticast IP.

The IP addresses in the range 224.0.0.0-239.255.255.255 are called multicast addresses. 224.0.1.0-224.0.1.255 is specified, and you generally use 224.0.2.0-239.255.255.255 as the multicast address. The MAC address corresponding to the IP address in the above range starts with 0x01, 0x00, and 0x5E, and the last three bytes are related to the specific IP address used.

To ensure compatibility, some devices may have stricter address range restrictions.

Because there is no explicit communication object, multicast must be created by the transmitting device. In terms of audio transmission, we call this multicast. Once a multicast is created, it persists, and the transmitting device does not know how many devices receive the multicast stream.

On a switch that supports and configures multicast, the switch forwards multicast packets only to the hosts that join the group. A device that wants to receive multicast must explicitly join the group represented by the multicast address in order to receive multicast packets.

Switches that do not support multicast will treat multicast packets as broadcast packets and forward them to all non-sending ports.

2. Create use and delete multicast

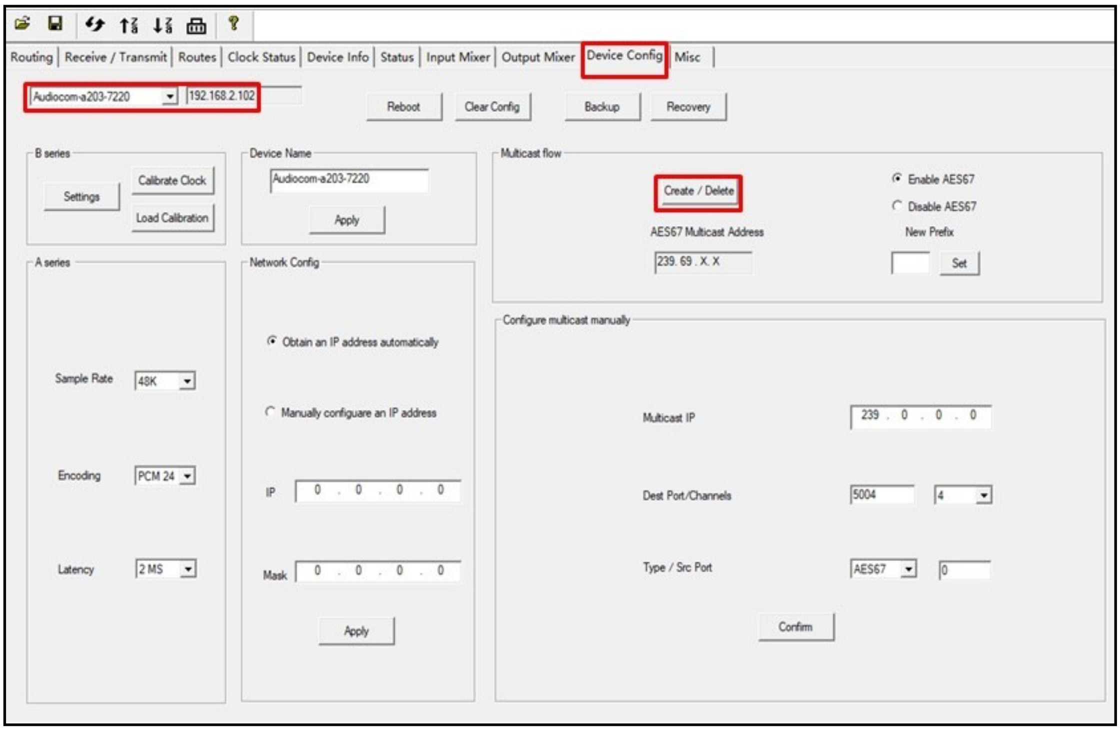
The A-series modules support AG and AES67 multicast standards, and can communicate with devices that support these two standards. A multicast can contain up to 8 channels. When you create a multicast, you need to specify exactly what type of multicast it is.

B-Series modules can only send AES67 multicast, and can only use AES67 communication with A-Series , but can receive AG multicast.

The AG multicast created by the A-series module does not appear on the transmitting device list and is preferentially used for communication between AG devices. The multicast of the AES67 standard is created and appears on the list of transmitting devices for receiving by devices that support this standard.

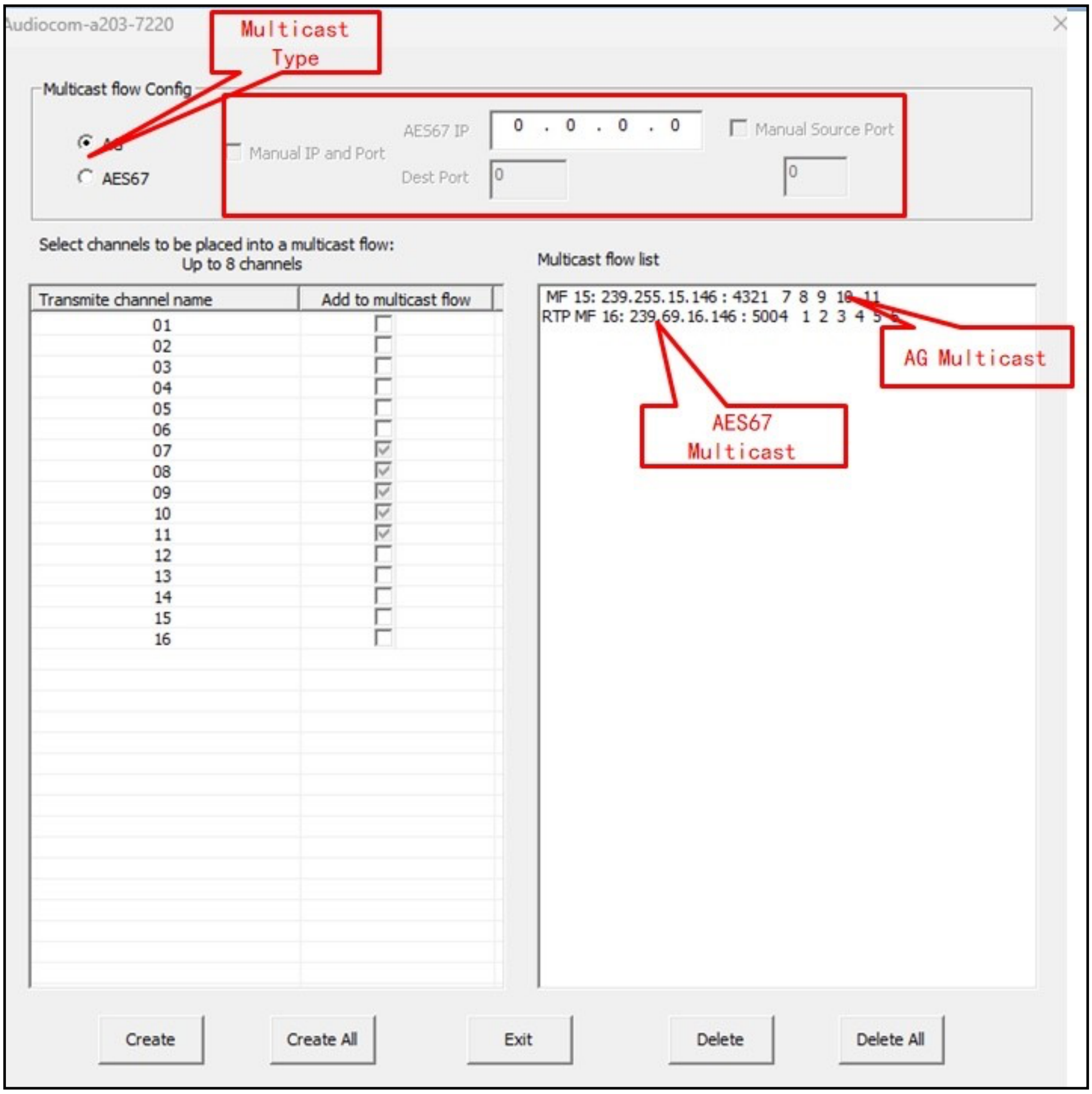
Click the Device Configuration tab, select the device you want to create a multicast in the Device drop-down list, and then select the Create/Delete button in the Multicast Management box to enter the multicast managent interface.

In the multicast management bar, you can enable or disable AES67 or set the multicast address (2nd byte only).



After entering the multicast management dialog box, it will list all the transmitting channels of the module and the created multicast, the multicast will display the type, the multicast IP and port used, and list the included send channels.

As shown in the figure, the module creates two multicast streams, namely an AES67 multicast (containing 6 channels from 1 to 6) and an AG multicast (containing 5 channels 7, 8, 9, 10, and 11) The AES67 multicast is marked with RTP in front of it, and it is displayed on the list of transmitting devices with the name "Module Name: Multicast ID".



(1) Create a multicast

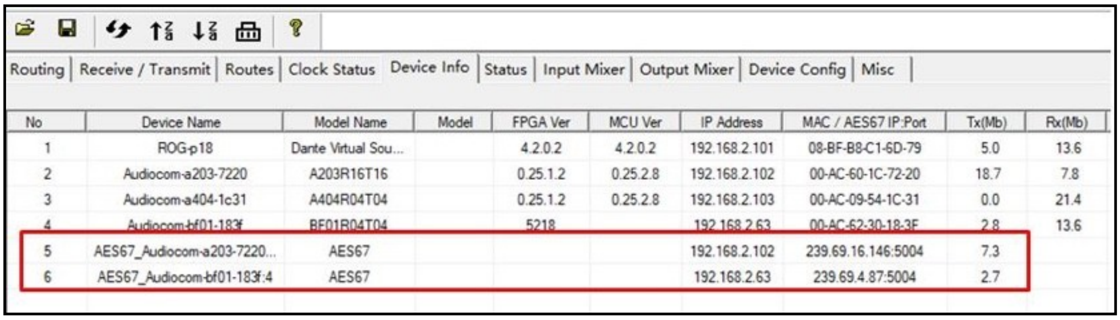
You first select the multicast type, then select the channel you want to transmit (a multicast can contain up to 8 channels), and then select the Create button to create a multicast. The created multicast appears in the list on the right.

The Create All button creates all channels as multicasts.

By default, multicast uses automatically and seamlessly generated multicast IPs and ports. When creating an AES67 multicast, you can specify the multicast IP address and port. When using the AES67 protocol to communicate with some modules that support AES67, the multicast IP address used must be the same as the first two bytes of the module, otherwise the communication cannot be normal.

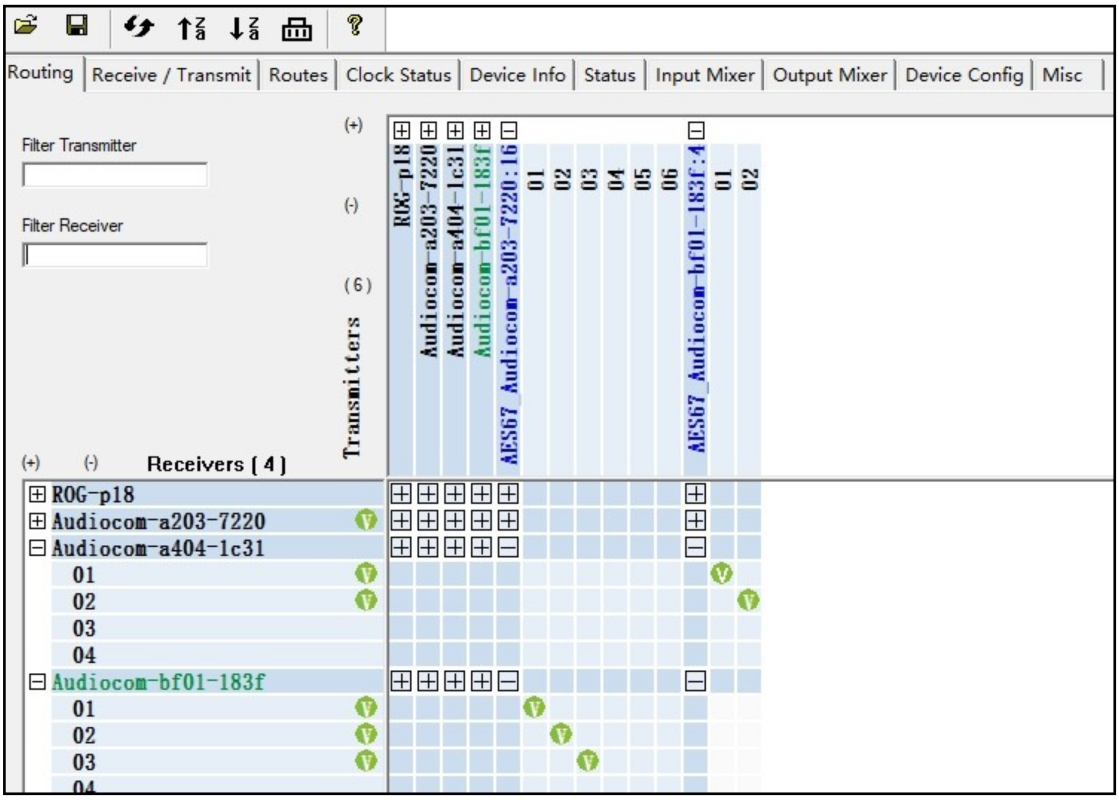
Port 5004 is generally used for AES67 multicast, and there is no need to change it unless there are special circumstances. The AES67 protocol of some vendors requires the use of a specified source port, which can be set by the you.

In the figure below, the red box shows the AES67 flow on the Device Info page, with AES67 at the front of the name. If the sending device is visible in the list, you can double-select the transmitting device to go to the Multicast Creation dialog box on the configuration page to view the specific channels that the multicast stream contains.



(2) Receive multicast

The created multicast is displayed in the transmitting device list on the Routing page. Turn on the intersection point of the receiving device and the multicast, and you can configure the specified channel to the receiving device. The three channels received by the A404 module in the following figure are all channels included in the multicast stream.



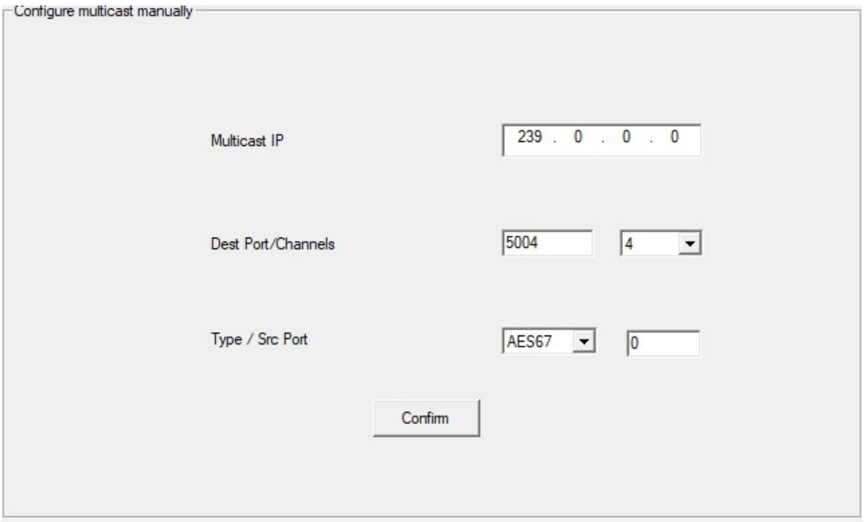
(3) Delete the multicast stream

In the Multicast Management dialog box, select a stream in the list and select the Delete button to delete the multicast. Delete All deletes all multicast from the current device .

3. Manually configure multicast

In general, AES67 multicast packets are automatically and seamlessly identified by AudioGridController when the announcement packets are sent by the device. If the device sends an announcement packet that is not standard or does not send an announcement packet, other devices cannot receive the multicast streamPorts and number of channels), which can be manually added in this software, and then you can configure routes on the Routing interface.

The software automatically and seamlessly saves the multicast data set by the you and loads it when AudioGridController starts. If you add a multicast stream manually, you can double-select the device name in the corresponding row in the device list to delete it.

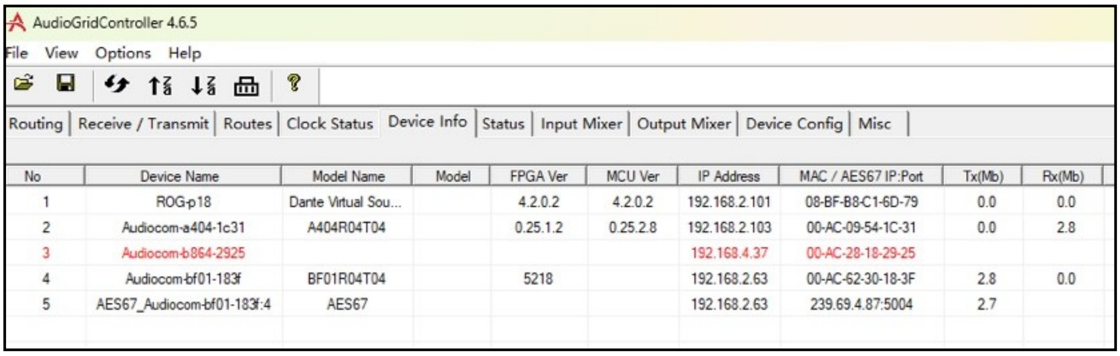


Normally, multicast does not require a source port to be set. For some special types of multicast, such as multicast transmitted by Dolby devices, it uses the source port to distinguish different multicasts, and the source port needs to be added here. For multicast with a source port set, AudioGridController starts with Dolby when displaying the name and identifies it with the source port at the end to distinguish it from other multicasts.

IX. Module configuration

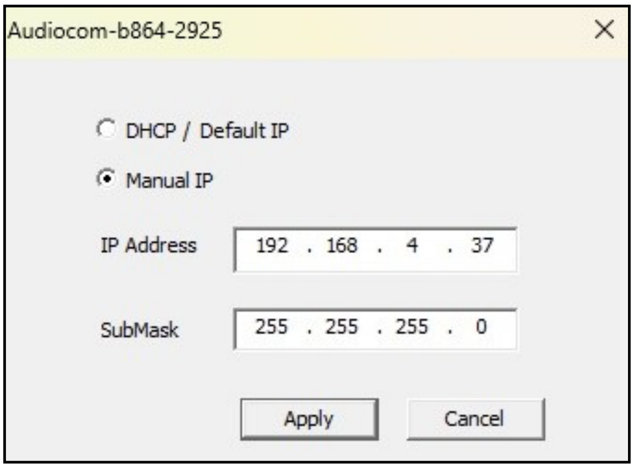
1. Modify device IP

For the modules produced by our company , In addition to modifying the IP address in the device configuration page, you can additionally double-select the serial number on the device information page to modify the device IP address in the pop-up dialog box. This is especially convenient for devices that are not on the same network segment as the computer (highlighted in red on the device information page).



No	Device Name	Model Name	Model	FPGA Ver	MCU Ver	IP Address	MAC / AES67 IP:Port	Tx(Mb)	Rx(Mb)
1	ROG-p18	Dante Virtual Sou...		4.2.0.2	4.2.0.2	192.168.2.101	08-BF-B8-C1-6D-79	0.0	0.0
2	Audiocom-a404-1c31	A404R04T04		0.25.1.2	0.25.2.8	192.168.2.103	00-AC-09-54-1C-31	0.0	2.8
3	Audiocom-b864-2925					192.168.4.37	00-AC-28-18-29-25		
4	Audiocom-bf01-183f	BF01R04T04		5218		192.168.2.63	00-AC-62-30-18-3F	2.8	0.0
5	AES67_Audiocom-bf01-183f:4	AES67				192.168.2.63	239.69.4.87:5004	2.7	

Double-select the serial number of the device, a dialog box will pop up to modify the IP address of the device, modify the IP address according to your needs, select the Apply button, the device will automatically and seamlessly restart, and run according to the new IP.

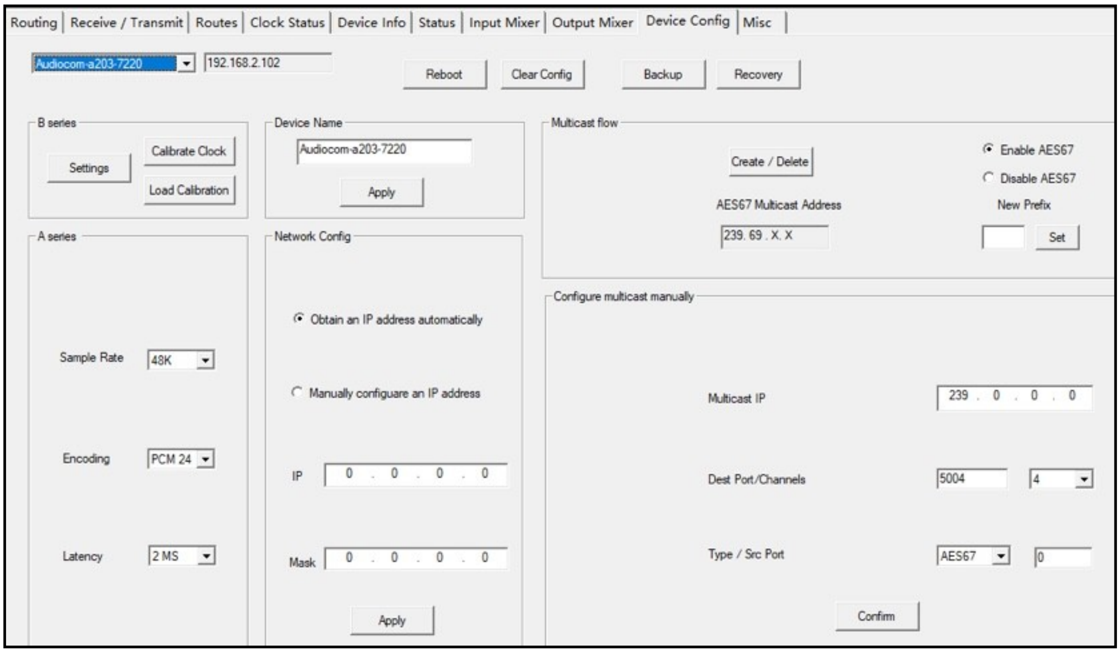


If a device not manufactured by our company is displayed in red on the device information page, you can only modify the IP address of the computer to make it in the same network segment as the device, modify it, restart the device, and finally change the IP address of the computer back to the original value.

2. Other settings

The module configuration function provides the function of changing the module name, IP address, sampling rate, number of coding bits and delay settings or modification functions. The software interface is also clearly marked, and you can operate it directly.

Before modifying the module parameters, you must first select the module you want to modify in the device list, and after selecting the device, the IP address will be changed accordingly.

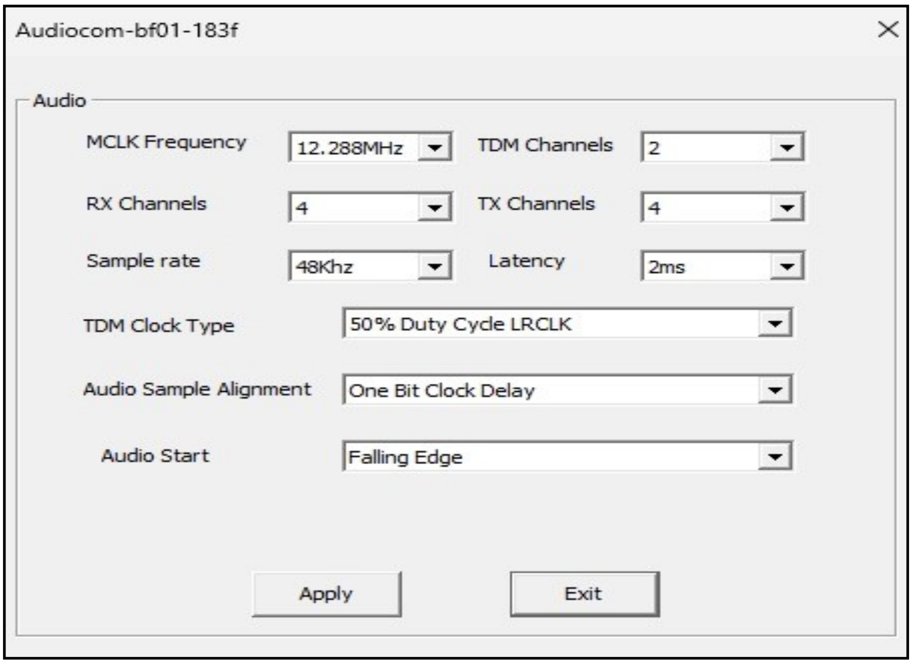


After the device name and IP address are modified, you need to select the Apply button to submit the device for execution, and the device will be restarted. The device name can contain up to 31 characters.

B-series modules can only use the specified IP address and cannot use DHCP to obtain the IP address. Users can use the 169.254.x.x address that it automatically and seamlessly generates or specify the address themselves.

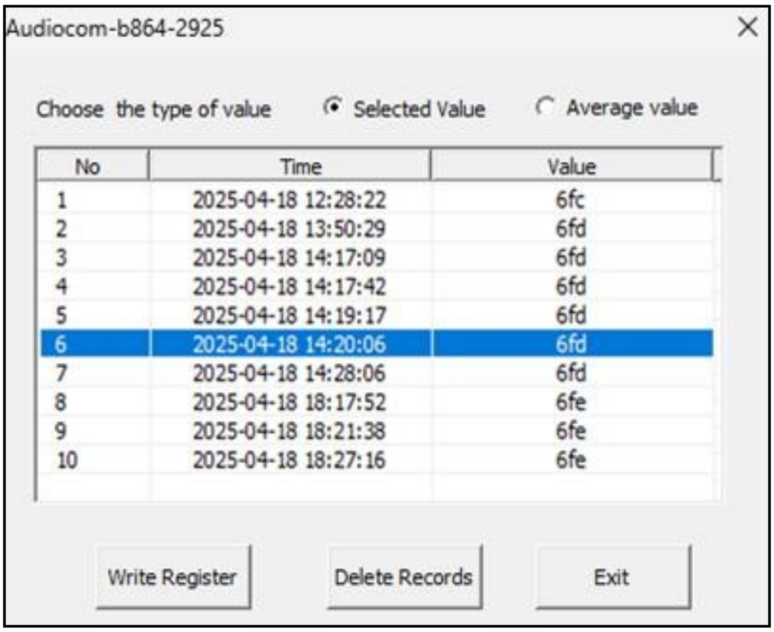
The sampling rate, number of coding bits and delay time are set in the same way, open the drop-down box, select the required parameters, and then select the Reboot button to restart the module.

The Clear Configuration button is to clear all parameters set by the you and restore the factory defaults. If the module is not behaving properly, you can use this button to restore the default values.
The hardware composition of the B-series module differs from that of the A-series, requiring specific handling for parameter configuration. To begin, select the target device from the device drop-down menu. Then, click the corresponding button in the B-series group box to open the settings dialog. After configuring the parameters, click ‘Apply’ and allow approximately 10 seconds for the device to save the settings properly.



3. Using GPS clock synchronization mode

(1) Calibrate module clock with GPS clock. Since v4.5.1, AudioGridController provides the GPS clock calibration function of the B series module, which is suitable for use to synchronize and calibrate the clock with the second signal of the GPS under the condition of GPS signal, record the calibration value in the module, and reload it when needed.



Up to 10 up-to-date calibration records can be stored in the memory area, including calibration time and calibration values.

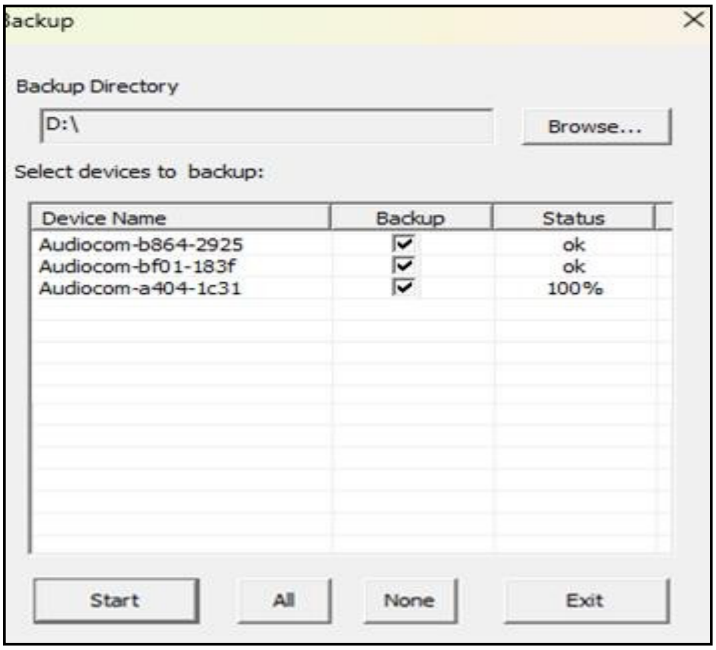
The you clicks on the calibration clock, and the module starts calibration. Depending on the strength of the GPS signal, the calibration time may take up to a few minutes, and a dialog box will pop up to inform the you when the calibration is complete.
When loaded, AudioGridController reads out all calibration records and displays them as a list. The you can select any of the write registers or load the average. If these calibration records are not needed, they can additionally be cleared.
(2) Turn on GPS clock synchronization mode. After the clock is calibrated, you need to set the 0x18 register to enable the GPS clock synchronization mode. The highest bit of the 0x18 register, b15, controls whether to use the GPS clock synchronization mode, defaults to 0, does not use. Read the value of this register on the Miscellaneous page, change the highest bit to 1, and then write. The highest bit of the 0x19 register is the GPS clock synchronization mode status bit, you can read its value after setting, if the highest bit is 1, it will be turned on successfully.

X. Data backup and recovery

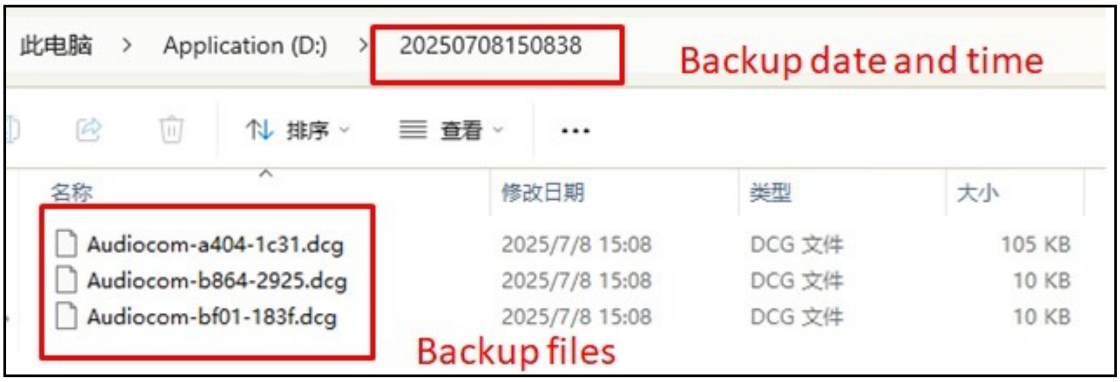
1. Data backup

The data backup function is to save all the configuration, subscription information, input mix, output mix and other data of the module in the specified directory of the computer. The software creates a secondary directory with the name of the date and time in this directory, uploads the selected module information to this directory one by one, and the backup file is named after the module.

Select the Device Configuration tab and select the Backup button to display the Backup dialog box, which will list the names of all modules. First, select the browse button to select the backup directory, then check the modules you want to save in the backup column, and select the start button to save the selected module information in the selected directory.



The following figure shows the results of backing up 3 modules, and the backup time is 15:08:38 on 8 July, 2025.



2. Data Recovery

The restore function is to re-import the backed-up module data into the module.

(1) Types of data recovery. There are two types of data recovery: full data recovery and configuration data recovery.

a. Full data recovery. The you is required to specify a backup directory. This recovery method is one-to-one, that is, the data can only be restored to the original module. The software will automatically and seamlessly find the data files that match the specified module during recovery, and restore the data to the module, and the module will automatically and seamlessly restart after the recovery is completed.

b. Configure data recovery. Configuration data refers to the module-related parameters configured by the you on the web interface, such as the master clock frequency, the number of transmitting and receiving channels, the TDM format, the audio sample alignment, the data packaging time, etc., that is called Web config data. In this way, the you needs to select the backup file of a module, and AudioGridController will automatically and seamlessly extract the configuration data contained in it when restoring, and restore it to all selected modules.

This method is suitable for you to customize the configuration parameters of a module, save them, and then restore them to other modules to improve configuration efficiency.

(2) Operation process:

a. Determine the type of recovery. Click the device tab and the recovery button to enter the recovery dialog interface. Once you are on this page, first select the recovery type, recover all data recovery or only the configuration data.

b. Select the backup directory (file). After selecting the recovery type, select the Browse button, and if it is an all-data recovery mode, select the directory named after the backup time. If you are configuring data recovery, select a backup file for a module.

c. Select the module that needs to be restored. Select the recovery type and backup directory (file), and then select the module to be restored. If there are many modules that need to be restored, you can use the All or Deselect All buttons to improve operational efficiency.

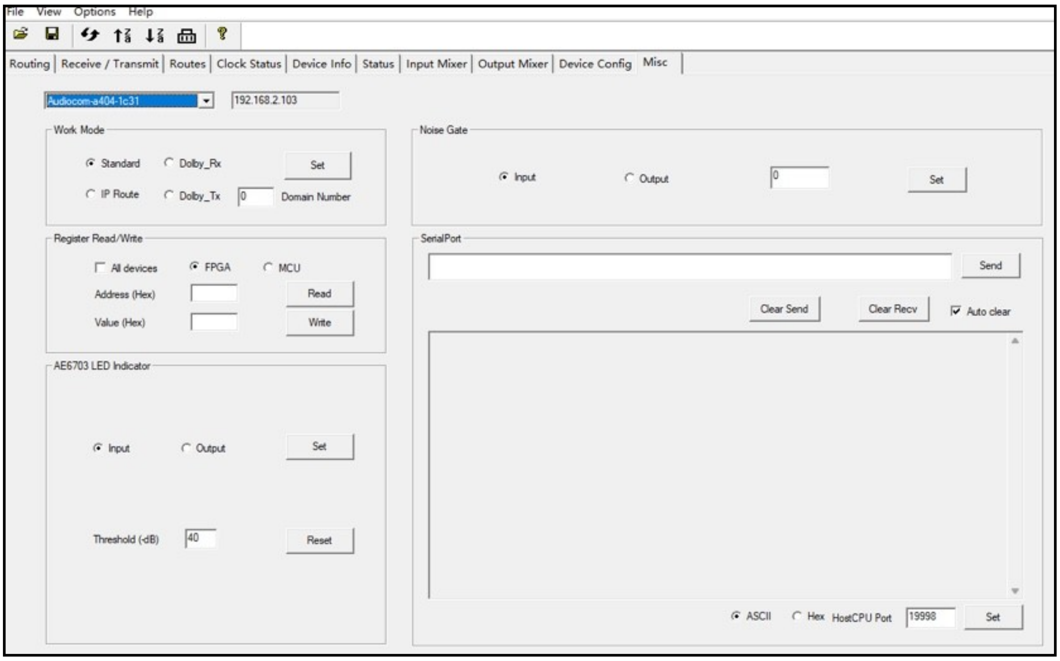
d. Perform recovery. Click on the Start button to start the recovery process. For all data recovery, AudioGridController will automatically and seamlessly find the backup file corresponding to each module (even if the backup file name is modified) and import the data into the module. If the configuration parameters are restored, AudioGridController will compare the module model and restore the data only for the module that matches the model of the backup file.

XI. Other operations

Depending on the customer's needs, we have set up the Re moteControl page for certain models of equipment or functions.

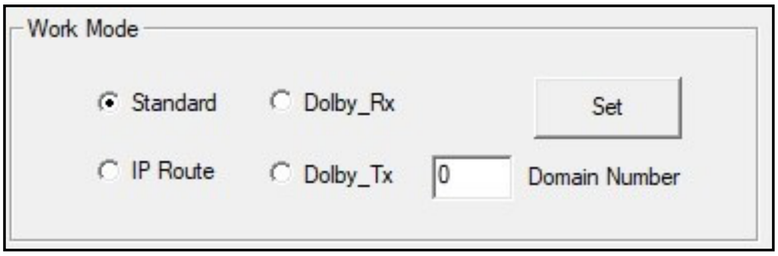
This page is set up for the special operation or function of the customized device, and can additionally be used by some you to debug or verify whether the data sent and received is correct.

Users generally do not need to operate on this page to avoid operating errors that cause the device to be unable to be used normally because they do not understand its meaning.



This page mainly includes the following features:

1. Set the working mode. For A-series module, it is possible to switch between working modes here. After the setting is completed, after 10 seconds, the power off and restart will run in the new mode. If you need to set the PTP domain number in Dolby mode, you can set it here at the same time.



2. Register reading and writing. Users can read and write registers according to their needs. Both the register address and the value are hexadecimal digits. For example, register 0 can read the model of the module, register 1 can read the firmware version, etc.

If you don't understand the meaning of the registers, you should not set them, otherwise the device may not work properly.

3. AE6703 LED input/output switching and lighting threshold setting. Users can set the LED on the AE6703 panel to display the input signal or output signal status according to their own needs.

The LED lighting threshold is -40dB by default, that is, the LED will only light up if it is higher than this value. The effective value range is 0~-120dB.

4. Serial port sending and receiving. It is mainly for you to debug the equipment to provide convenience. Users can input content in the Send edit box and send it directly to the serial output of the module. The content output from the serial port is COBS encoded, and the sent content can be restored only after decoding. Similarly, the content input from the serial port also needs to be COBS encoded before it can be forwarded to the PC through the module, and the received content can be displayed in both character or hexadecimal number formats (select the ASCII and Hex buttons to switch the display format).

The HostCPU port in the lower right corner refers to the function port used by the target device, and the factory default value is 20000. If you cannot send and receive the 0x12 from the web page, you can read the register of the MCU, and the resulting number is the hexadecimal value of the port. Fill in the port number read in the HostCPU Port field, select Set, and the communication can be carried out normally.

For the data format of the serial port and the coding rules and functions of COBS, please refer to the "Instructions for Network and Serial Port Communication".